Wireless Communications

T L Singal

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About the Author

T L Singal is currently Professor in the Department of Electronics and Communication Engineering, Chitkara Institute of Engineering and Technology, Rajpura, Punjab. He graduated in Electronics and Communication Engineering from National Institute of Technology (NIT), Kurukshetra, and post-graduated in Electronics and Communication Engineering from Punjab Technical University, Jalandhar. He has worked in the field of Wireless and Cellular Communications with leading telecom organisations in India and USA for about 20 years and has managed various VHF/UHF Wireless Communication Projects in reputed companies like HAL, Hyderabad, and PSIDC, Chandigarh. He has also visited Germany on business trips for technical know-how on the transfer of Multi-Access Wireless Communication Systems during 1990–92. In his last association with Flextronics International Inc., Dallas (Texas), USA, he held the position of Senior Network Consultant, offering optimisation solutions in the domain of GSM, and CDMA cellular networks for various clients such as Nokia, AT&T, Nortel, Cingular Wireless, Voice Stream USA during 2000–2002. Since 2003, he has been working with engineering institutes as Senior Faculty of Electronics and Communication Engineering, specialising in Cellular and Mobile Communications, with Punjab Technical University, Jalandhar, India.

His areas of interests include modeling and performance analysis of digital communication systems, seamless mobility in future-generation digital cellular networks, and convergence of wireless and Internet technologies. He has convened international and national conferences on Wireless Networks, and edited proceedings of conferences. Besides these, he has also presented several technical research papers in international and national conferences.

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To

My Parents Shri K R Singal and Smt. Kaushalya Devi

My Wife Pinki

Daughter Ritu

Son Pankaj

Contents

Preface

The first decade of the 21st century belongs to a new wireless world indeed! The rapid growth of cellphones, wireless LANs, and recently the wireless Internet, in short, wireless communications is driving the whole world towards greater integrity with wireless communications. Whether it be metropolitan congested urban areas, or modern industrialised economic zones, or even vast rural fields; wireless voice/data communications is the need of the hour in every walk of life. Wireless communication systems and networks are being installed and commissioned in many towns and cities every day throughout the world. Local residents as well as visitors at any location use handheld wireless devices and laptops to remain connected to the rest of the world.

The availability of more and more RF spectrum in higher bands (1 GHz and more) for mobile communications, mobility awareness in civilised societies, global standardisation of wireless devices and products, expansion of business using information technology and many more such factors are leading towards huge growth rates in wireless communication systems. The advent of new mobile applications and data-transmission techniques will make ubiquitous multimedia mobile computing a reality sooner than later.

Objective

For more than two decades, I have been involved in the design, development, testing, system integration, and commissioning of full-duplex VHF/UHF wireless communication systems, and analog/digital cellular systems with leading telecom companies. With the objective of imparting practical knowledge gained during my industrial experience in the field of wireless and cellular communications, embedded with theoretical classroom teaching to the engineering aspirants, I started my academic profession about seven years ago. During my interaction with the students of Electronics and Communication Engineering on the course(s) of Cellular and Mobile Communications, and Wireless Communication Systems and Networks, I strongly felt that there was a need of a comprehensive textbook which deals with fundamentals of mobile communications, basic concepts of cellular design, and then moves towards imparting a thorough understanding of cellular systems and emerging wireless networking technologies. Based on my rich experience in design, development and system integration of various wireless and cellular systems in telecom industries, I was motivated to share the industries' needs with the aspirants of Electronics and Communication Engineering through a concise but comprehensive and easy-to-understand book on 'Wireless Communications', which also includes my teaching notes and answers to queries raised during the teaching–learning phase.

About The Book:

Wireless Communications is a comprehensive text designed to cover the major topics in the field of wireless and mobile communications, cellular architecture and technology, cellular communication systems and networks, and emerging wireless networking technologies. Most of the contents specified in the curriculum of wireless communications and allied subjects at the undergraduate as well as postgraduate level of Electronics and Communication Engineering in different technical universities and institutes are covered here. There is no single book available in the market presently which caters to all these courses. An overview of the latest trends in the advancement of wireless and cellular communication technologies is also presented in this book, which would enable the reader to revise the curriculum of related courses in future. In general, the chapters and sections of the chapters are structured in a modular fashion to provide sufficient flexibility in the design of the course.

The text is easy-to-read, logical, and has a step-by-step approach that enables the reader to follow new and complex ideas clearly. Throughout the book, an emphasis is put on both wireless technology fundamentals and standard wireless systems/networks. The book also provides a comprehensive guide to understanding specific wireless standards, such as those developed by ITU, IEEE and other similar organisations. Typical pedagogical features such as summary, important equations, key terms, self-test quiz, review questions, analytical problems and references provided at the end of each chapter will enable aspirants to prepare for various competitive exams such as GATE, IES, PhD entrance, amongst others. It will also serve as a reference guide for technical interviews in the field of telecom and cellular networks.

Moreover, a reasonable exposure to integration of cellular networks and wireless LANs encourage students to take up research activities. The book provides a good background for planning and design of wireless communication systems. Therefore, it will serve as a reference text and is suitable for self-study purpose for the practising professionals in the field of wireless and cellular communications.

Target Audience

This book is primarily intended to serve as a textbook for undergraduate and postgraduate students, specialising in wireless communications. The reader is supposed to have a background in electronics and communication engineering and be familiar with the theory of signals and systems, electromagnetic field theory, analog and digital communications, antennas and wave propagation. These courses are offered in most Electronics and Communication Engineering undergraduate programmes. The content is useful for wireless/mobile communications, cellular and mobile communications, wireless communication systems and networks, and advanced communication systems.

The material in this book can be taught in two different courses. The first course may cover chapters 1 to 8, which deal with the principles of wireless/mobile communications, cellular concept and designs. The second course could cover the remaining chapters, which include basic cellular system, wireless communication systems, and emerging cellular and wireless networking technologies. Since each chapter is rather comprehensive on the topics treated and is relatively self-contained, the reader can select to read only those chapters that are of interest to him/her or as per the specified syllabus.

Roadmap for Various Target Courses

The whole book is structured in chapterwise and topicwise modular forms. Different courses based on wireless communications prescribed at undergraduate and postgraduate engineering levels can be planned in a systematic manner. The following mapping of chapters for different courses provides guidelines for compulsory reading of the topics, with options of selective coverage of the topics from other chapters as per the requirement.

Course→ Chapter↓	*Wireless/mobile communications*	*Cellular and mobile communications*	*Wireless communication systems and networks*	*Advanced communication systems*
1	√	√	√	√
2	√	√	X	√
3	√	√	X	X
4	X	√	X	√
5	√	√	X	X
6	X	√	X	√
7	√	√	X	X
8	√	√	√	√
9	X	√	√	√
10	X	X	√	X
11–14	X	X	√	√

Salient Features

The book presents a detailed understanding on fundamentals of wireless communications. It provides a good background for planning and design of cellular communication systems, and an insight of wireless networking technologies.

Following are the main features of this book:

- Fundamentals of Wireless Communications presented in easy language
- Comprehensive coverage on Cellular Communications and Antenna System Design Considerations
- Detailed description of Digital Cellular Technologies including GSM, GPRS, CDMA
- Brief overview of emerging trends like Wi-Max, Wi-Media UWB & RFID
- Includes discussion on emerging wireless technologies like IEEE 802.11x, IEEE 802.16x , WMAN, MANETs, WSN
- Rich pool of pedagogy including solved problems, practice questions, objective type questions and short answer type questions
- Model test papers with answers given at the end of the book will help the student get an idea about composition of exam papers.

Organisation of the Book

The chapter and material sequence in this book have been adapted from various undergraduate and postgraduate level course(s). The book is organised into 14 chapters. Chapter 1 gives an overview of the history of evolution of wireless communications, wireless network generations, and a wide range of applications of wireless data communications. Chapters 2 and 3 present the fundamentals of mobile communication engineering including various propagation path-loss models under different operating conditions and their impact on received signal strengths at the mobile end. Chapters 4–7 deal with the principles of cellular communications such as frequency reuse, cochannel interference, antenna system design considerations including cell sectoring, frequency management and channel assignment techniques, and extensive analysis of design trade-offs of cellular system designs. Chapter 8 describes multiple channel access techniques as well as packet radio and carrier-sense-based multiple access protocols. Chapter 9 introduces the architecture and operation of a basic cellular system. It also includes performance criteria to evaluate and compare different cellular systems. Chapters 10–12 explain different design and evaluation aspects of first-generation and second-generation cellular systems such as GSM and CDMA. Chapter 13 provides an overview of 3G cellular technology based on UMTS as well as CDMA2000.

Chapter 14 presents an insight into the emerging wireless network technologies such as IEEE 802.11 WLAN, HIPERLAN, IEEE 802.15 WPAN, IEEE 802.16 WMAN, Mobile Ad-hoc Networks (MANETs), RFID technology, and Wireless Sensor Networks (WSNs). A brief overview of mobile IP and TCP as well as security requirements for wireless networks is also presented here.

Web Supplements

There are a number of supplementary resources which is available on the Website and updated from time to time to support this book. These can be accessed at http://www.mhhe.com/singal/wc. The website includes the following:

For Instructors:

- Instructor's Solution Manual *The solution manual (password protected) contains hints and solutions to analytical problems provided at the end of each chapter. The instructors would be able to access it through an assigned password. This will assist the instructors to prepare tutorial work and assignments.*

- PowerPoint Slides *A complete set of Microsoft PowerPoint slides for each chapter is available for the instructors. These slides are meant to be used as a teaching aid for classroom presentations which can be made available to students on the intranet for chapter review, or can be printed for classroom distributions. Instructors may add their own slides for further elaboration on any topic.*

For Students:

- Interactive Quiz and Additional Questions *Interactive quiz and additional questions are given on the website for student's reference.*

Acknowledgements

As I began writing the script for this book, neither did I know how much work would be involved in this kind of project, nor how many people who would play roles in different capacities. At the outset, I would like to express my gratitude to Tata McGraw Hill Education for providing me an opportunity to publish this book and I thank the many people responsible for the publication of this book. To begin with, my thanks to Aman Taneja, for appreciating my willingness to write the script, and introducing me to the TMH team who took up this project. My sincere thanks are also due to Suman Sen and Manish Choudhary for their continuous encouragement to include the reviewers' feedback and comments during the preparation of the text. I also want to thank Vibha Mahajan, P L Pandita, Sohini Mukherjee, Rachna Sehgal and other staff members of Tata McGraw Hill Education for their excellent coordination during copyediting and production stages of the book.

The dream of my beloved parents, who wished me to be a mentor for aspiring young engineering students, is fulfilled through the publication of this book. Their blessings are similar to those bestowed by the Almighty. I thank my wife, Pinki; daughter, Ritu; and son, Pankaj for their inspiration, patience and continuous support. It is pertinent to mention the useful contribution made by Pankaj using his expertise with computer work, in spite of being so busy in his engineering studies. Finally, I am grateful to peer reviewers who evaluated each chapter and provided very useful suggestions and comments. Their names are given as follows.

Rajender Kumar	*National Institute of Technology (NIT) Kurukshetra, Haryana*
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Sanjeev Jain	*Government Engineering College, Bikaner, Rajasthan*
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Poonam Singh	*National Institute of Technology (NIT) Rourkela, Orissa*
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Suraiya Tarannum	*AMC Engineering College, Bangalore, Karnataka*
A Sesha Rao	*Vignan Institute of Information Technology Vishakhapatnam, Andhra Pradesh*

Reviewers' Comments

Some of the comments offered by the reviewers on the typescript (which provided immense motivation and encouragement) are reproduced below.

'It is a very good book on 'Wireless Communications' covering fundamentals of wireless networks to the evolution of 3G and 4G wireless networks. The propagation models, which are the fundamental of any 2G, 3G and 4G wireless network, are presented in detail. The inclusion of summary, important equations, key terms, short-answer-type questions with answers, self-test quiz, review questions, analytical problems and references is excellent. The book will be helpful to UG and PG students.'

'The author has made a fairly good presentation of fourteen chapters. The most appealing feature of this book is that it leans towards the intuition and motivations around the important topics of the subject. I believe that it is an excellent resource for students to understand wireless communication thoroughly.'

'The author has tried to simplify the subject in the best possible way. One cannot think of writing such a complicated subject in such a simple and interesting manner. This helps the students understand and visualise the subject.'

'The script has given due importance to the propagation models and Antenna Design Considerations.

Feedback

I hope every reader will find this book rich in content as well as pedagogical features. You will certainly enjoy the simplified yet extensive and elaborate approach to every new topic under discussion. I have tried my level best to make it a complete book on Wireless Communication but I believe there is always scope for improvement in all our efforts. Your valuable suggestions, feedback, or any other comments are most welcome at my email id: tarsemsingal@gmail.com. I am sure this will enable me to enhance the content of the next edition.

T L Singal

Publisher's Note

Do you have any feedback? We look forward to receive your views and suggestions for improvement. The same can be sent to tmh.ecefeedback@gmail.com mentioning the title and author's name in the subject line.

List of Important Symbols

γ	path-loss exponent or distance-power gradient
λ_c	wavelength of transmission
ρ	the value of the specified level, normalised to the local rms amplitude of the fading envelope
Ť	average fade duration
'I	transmission delay
h_t	transmitter antenna height
h_r	receiver antenna height
f_c	frequency of transmission
φ	elevation angle
ϑ	incident angle
L_{pO}	Okumura propagation path loss
L_{pf}	free-space propagation path loss
L_{pm}	mobile propagation path loss
L_{pH}	median value of the propagation path loss
α_m	median attenuation relative to free space
α_t	effective base station antenna height gain factor
α_r	effective mobile receiver antenna height gain factor
α_c	correction factor gain
Tx	transmitter
Rx	receiver
L_{PL}	total partition loss
L_{pLog}	log-distance path loss
P_r	received power
P_t	transmitter power
G_r	receiver antenna gain
G_t	transmitter antenna gain
d_f	near-field distance
f_d	Doppler frequency
V_m	mobile speed
B_c	coherence bandwidth
Ŧ$_d$	delay spread
K	cluster size
q	frequency reuse ratio
D	distance between co-channel cells
R	radius of the cell
r	distance between base station transmitter and mobile receiver
A_{eff}	effective area of receiver antenna
C/I	carrier to interference ratio
f_m	modulating frequency
m_a	modulation index AM
m_f	modulation index FM
Δf	peak frequency deviation of the FM signal
k_f	frequency deviation constant
A_m	peak value of the modulating signal
B_t	total spectrum allocation
B_g	guard band
B_c	channel bandwidth
λ	packet arrival rate
A_{av}	average traffic intensity
H	holding time
N_t	total number of channels
N_d	number of data channels per cluster
A_{cell}	geographical area covered by a cell
η_1	overall spectral efficiency in channels/MHz/km^2
η_2	spectral efficiency in Erlangs/MHz/km^2
N_u	total number of sub-bands
J	radio capacity
G_p	processing gain
R_b	bit rate
M	number of mobile users
E_b	energy per bit
N_o	noise power spectral density
σ	intercell interference
Q	number of chips per time period T
G_A	sectorisation gain factor
G_v	voice activity interference reduction factor
Y	interference improvement factor
α	power control accuracy factor
v	voice activity factor

VISUAL TOUR

Introduction

Each chapter begins with an Introduction that gives a brief summary of the background and contents of the chapter.

1

The term 'wireless' is often used to describe all types of devices and technologies that use space as a signal-propagating medium, and are not connected by a wire or cable. Wireless communication may be defined as the transmission of user information without the use of wires. The user information could be in the form of human voice, digital data, e-mail messages, video and other services. Wireless communication is revolutionising almost every aspect of our daily lives. From the ubiquitous mobile phone to automated inventory counting in large retail stores to remote wireless sensors installed in inaccessible locations, wireless-communication technology is poised to continue expanding at a very fast pace. Using wireless communications to send and receive messages, browse the Internet, and access corporate databases anywhere, any time across the world has already become common. A wide array of devices ranging from computers to digital cameras, laser printers, and even household appliances can already communicate without wires. We are moving towards a wireless world to live in a global village. Starting with a brief history of wireless communications, this chapter provides a quick overview of comparison and applications of wireless communications and systems. Knowing about wireless network generations is important in studying the evolution to next-generation networks. This chapter concludes with potential market areas and challenges for research in the field of

Evolution of Wireless Communication Systems

1.1 BRIEF HISTORY OF WIRELESS COMMUNICATIONS

To appreciate the current technology and prepare ourselves to enhance its development in future, it is always interesting to have a quick glance at the history of a technology such as wireless communications. There are always several smaller steps that take place in leading up to the development of a new technology. Tracing the development of these earlier discoveries in brief can help us better understand how this technology actually functions and contributes towards what could be the next development. A brief review of the history of wireless communications covering radio, television, radar, satellite, wireless and mobile, cellular and other wireless networks are presented here.

Table 12.11 *Comparison of CDMA and GSM technologies*

S. No.	*Parameter*	*CDMA (IS-95)*	*GSM*
1.	Allocated spectrum band	800 MHz; 1900 MHz	900 MHz; 1800 MHz (DCS 1800); 1900 MHz (PCS 1900)
2.	Multiple access technique	CDMA	TDMA
3.	RF and channel bandwidth	12 MHz with 1.25 MHz per carrier channel for spread spectrum	25 MHz with 200 kHz per carrier channel, 8 time slots per channel with frequency hopping
4.	Modulation technique	QPSK with DSSS	GMSK
5.	Multipath phenomenon	Used as an advantage with rake receivers	Causes fading and interference which degrade performance
6.	Use of SIM card at MS	No	Yes
7.	Data bit rate	9.6 kbps or 14.4 kbps	9.6 kbps
8.	Voice codec rate	8 kbps or 13 kbps	13 kbps
9.	SMS feature	Up to 120 text characters	Up to 160 text characters
10.	Nominal MS transmit power	2 mW to 200 mW	125 mW to 2 W
11.	Hand-off mechanism	Soft hand-off	Hard Hand-off
12.	System capacity	Flexible and better than GSM	Fixed and limited
13.	Network planning	PN code planning	Frequency planning with reuse concept

Tables

The facts and figures are arranged in Tables at appropriate places in various chapters.

Benefit:

Will help in comparative analysis of features and technical specifications.

Facts to Know!

Wherever appropriate, additional useful information related to the topic under discussion is inserted and highlighted in a box.

Benefit:

This can be basis to extra reading material for interested readers.

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cellular system and an analytical approach is desirable. For example, it has been observed that for a fixed-channel-to-dynamic-channel ratio of 3:1, the hybrid channel-assignment scheme leads to better service than the fixed channel-assignment scheme for traffic up to 50%. Beyond 50% traffic load, the fixed channel-assignment scheme performs better.

Facts to Know!

Hybrid channel-assignment scheme has the advantages of both static and dynamic channel-assignment schemes. It has low overhead of channel management, lower run-time overhead, reduction in hot spot, and improved channel-load balancing. On the other hand, if hybrid channel assignment is not coordinated properly, the benefits of both static and dynamic channel assignment strategies are lost.

To assign a channel from a dynamic channel set effectively, a centralised control is used with up-to-date traffic-pattern information. There are two different methods used in assigning dynamic channels: scheduled and predictive. In scheduled dynamic channel assignment, a priori estimates about variation in traffic in terms of location and time are needed to schedule dynamic channels at predetermined peaks of traffic change. In a predictive dynamic channel assignment, the traffic intensity and blocking probability is monitored in each cell at all the time so that dynamic channels can be assigned to each cell according to its needs.

Channel borrowing in hybrid channel assignment has similar issues as in dynamic channel assignment. The primary performance concerns include channel locking, channel unlocking and system overheads. Because of these difficulties, dynamic channel assignment and hybrid channel assignment generally perform less satisfactorily than fixed channel assignment under heavy traffic conditions.

Cellular Antenna System Design Considerations | 133

When a cell-site is chosen on the map, there is a 50% chance that the site location can be acquired. So an antenna location can be found within a quarter of the size of the cell $R/4$ where R is the radius of the cell. This is termed as the quarter-radius rule. For example, if the radius of a cell is 12 km, the cell-site antenna can be located within a 3-km radius ($R/4$). The change in the cell-site antenna within a 3-km radius around the centre of the cell would not affect the coverage pattern at a distance 12 km away. If the radius of the cell is 3 km then the cell-site antenna can be located within a 0.75-km radius from its centre. However, the quarter-radius rule can be applied only on relatively flat terrain, not in a hilly area.

There is a need to check the cell-site antennas regularly. Air-pressurised RF coaxial cable is generally used to connect the base-station equipment with the cell-site antennas. This enables to prevent moisture from entering the RF coaxial cable and causing excessive signal attenuation. The RF power delivered to the antenna terminal can be checked with the help of an *RF Power Meter*. Alternatively, VSWR can be checked at the bottom of the RF cable antenna

Fig. 5.5 | *Cell-site antenna tower with various antennas mounted on it*

Real-Life Images

Other than conventional figures and illustrations, selective Real-Life Images are used to demonstrate the practical aspect.

Benefit:

This will help the reader in relating and visualising the concept learnt with the real-life scenarios.

Solved Examples

Adequate number of step-by-step Solved Examples to support theory given in each chapter.

Benefit:

The step by step approach helps in easy understanding of the concepts and associated problems.

EXAMPLE 4.3 | Cell size and system capacity

(a) Assume a cellular system of 32 cells with a cell radius of 1.6 km, a total spectrum allocation that supports 336 traffic channels, and a reuse pattern of 7. Calculate the total service area covered with this configuration, the number of channels per cell, and a total system capacity. Assume regular hexagonal cellular topology.

(b) Let the cell size be reduced to the extent that the same area as covered in Part (a) with 128 cells. Find the radius of the new cell, and new system capacity.

Comment on the results obtained.

Solution

(a) *To calculate total service area, number of channels per cell, and system capacity*

Total number of cells in service area = 32 (given)

Radius of a cell, $R = 1.6$ km (given)

Step 1. *To calculate area of a regular hexagonal cell*

Area of a regular hexagonal cell, $A_{cell} = 3\sqrt{3}/2 \times R^2$

Therefore, $A_{cell} = 3\sqrt{3}/2 \times (1.6 \text{ km})^2 = 6.65 \text{ km}^2$

Step 2. *To calculate total service area*

Total service area covered = no. of cells in total area × Area of a cell

Hence, **total service area covered** = 32 × 6.65 = **213 km²**

Important Equations

- ❑ $P_r = P_t\, G_t\, G_r\, [\lambda_c / (4\pi r)]^2$ (3.12)
- ❑ $L_{pf} = (4\pi r / \lambda_c)^2$ (3.13)
- ❑ $L_{pf}(\text{dB}) = 32.44 + 20 \log r\,(\text{km}) + 20 \log f_c\,(\text{MHz})$ (3.14)
- ❑ $P_r = P_t\, G_t\, G_r / L_{pf}$ (3.15)
- ❑ $P_r\,(\text{dBm}) = P_t\,(\text{dBm}) + G_t\,(\text{dBi}) + G_r\,(\text{dBi}) - L_{pf}(\text{dB})$ (3.16)
- ❑ $P_r = P_t\, G_t\, G_r\, h_t^2\, h_r^2 / r^4$ (3.27)
- ❑ $L_{pm} = r^4 / (h_t^2\, h_r^2)$ (3.30)
- ❑ $L_{pm}(\text{dB}) = 40 \log r - 20 \log h_t - 20 \log h_r$ (3.31)
- ❑ $P_r\,(\text{dBm}) = P_t(\text{dBm}) + G_t\,(\text{dB}) + G_r\,(\text{dB}) - L_{pm}(\text{dB})$ (3.32)
- ❑ $d_f = 4\,(h_t\, h_r) / (\lambda_c)$ (3.35)
- ❑ $L_{pH}(\text{dB}) = 68.75 + 26.16 \log f_c - 13.82 \log h_t + (44.9 - 6.55 \log h_t) \log r$ (3.44)

Important Equations

For handy reference, a brief list of the most commonly used equations are given after summary at the end of each chapter.

Benefit:

While solving the analytical problems, the students can directly refer to equations and use them. This will also be useful for quick re-cap before the exam.

Summary

At the end of each chapter, a summary of important concepts are given.

Benefit:

This will come handy quick re-cap before exam.

Summary

In this chapter, various multiple access techniques that are used in analog and digital cellular mobile standards are discussed in detail. The efficient ways to access the limited radio spectrum by number of potential subscribers simultaneously in a wireless-environment span through division of space, frequency, time and code are discussed. For an efficient use of resources by multiple number of subscribers simultaneously, multiple access techniques such as FDMA, TDMA, or CDMA are used in a wireless cellular system. Thus communication channels are used by system subscribers and there are many multiple access techniques that can be used effectively. Their relative advantages and disadvantages have been outlined here and problems and limitations of using such resources have been widely discussed. A number of subscribers need to access the control channel on shared basis at random times and for random periods. Controlling access to a shared medium is important from the point of view that, at any given time, only one subscriber is allowed to talk while the rest of the subscribers listen. An overview of packet radio multiple access technique is also presented here. A functional schematic of basic cellular system along with its operation and performance criteria is the main attention in the next chapter.

Key Terms

- Antenna
- Attenuation
- Beamwidth
- Bit error rate (BER)
- Coherence bandwidth
- Coherence time
- Delay spread
- Diffraction
- Direct wave path
- Diversity
- Doppler shift
- Doppler spread
- Fade duration
- Fade rate
- Fading
- Fast fading
- Flat fading
- Free space
- Frequency selective fading
- Line-of-sight (LOS)
- Multipath interference
- Noise
- Obstructive path
- Radio path
- Rayleigh fading
- Reflection
- Refraction
- Rician fading
- Scattering
- Slow fading
- Wireless

Key Terms

The important terminologies introduced in each chapter are touched upon once again in the form of a list of Key Terms for quick reference.

Benefit:

This will give an idea to the readers about the coverage of focus areas of that chapter.

Short-Answer Type Questions with Answers

A11.1 How does a SIM card provide security against fraudulent use of GSM phone?

A GSM mobile phone is useless without a valid SIM card. SIM cards carry the private information for a user. The SIM can be set up to require the user to enter a four-digit Personal Identification Number (PIN) whenever the mobile phone is switched on. This feature provides some security in case of its fraudulent use from a lost or stolen mobile phone. If the mobile subscriber removes the SIM card when leaving the mobile phone in a vehicle, the mobile phone cannot be used unless another person has a valid SIM.

A11.2 Why are the needs for the wireless and wired media different?

The wireless medium is unreliable, bandwidth limited, and needs to support mobility. As a result, protocols used in the wireless and wired media are different. The BSS provides for the necessary translation among these protocols.

A11.4 List some of the main responsibilities of the Radio Resource Management (RRM) sub-layer of the network layer.

The main responsibilities of the RRM include the assignment of the radio channel and hopping to new channels in implementation of the slow frequency-hopping option, managing hand-off procedure and measurement reports from MS for hand-off decision, to implement power control procedure, and to adapt to timing advance for synchronisation purpose.

A11.5 What is the significance of Temporary Mobile Subscriber Identity (TMSI)?

As all transmissions between MS and BSS are sent through the air interface, there is a constant threat to the security of the information exchanged. A temporary identity TMSI is usually sent in place of International Mobile Subscriber Equipment Identity (IMSEI).

A11.6 What are the main functions of Frequenc

Short-Answer Type Questions with Answers

Sufficient number of Short-Answer Type Questions with Answers are provided at the end of each chapter.

Benefit:

Provides useful information from examination point of view, it will help students to revise and consolidate the topics.

Self-test Quiz

The end-of-chapter assessment begins with a set of Self-test Quiz which includes numerous multiple-choice/objective-type questions with an aim of reinforcing the concepts discussed in the chapter. Answers to all these questions are also given for self evaluation and revision purpose.

Benefit:
The readers will be motivated to solve the multiple choice questions and verify their responses. It is a quick way of revising the key concepts covered in the chapter.

Self-Test Quiz

S6.1 In the 800 MHz band cellular system the duplex separation is specified as
(a) 20 MHz (c) 45 MHz
(b) 25 MHz (d) 80 MHz

S6.2 The original standard non-extended spectrum in US-AMPS analog cellular system has ______ ______ allocated bandwidth on either side of the duplex spectrum.
(a) 10 MHz (c) 25 MHz
(b) 20 MHz (d) 30 MHz

S6.3 The channel spacing in standard US-AMPS analog cellular system is
(a) 10 kHz (c) 30 kHz
(b) 25 kHz (d) 200 kHz

S6.4 In extended spectrum US-AMPS cellular standard, the uplink frequency band (Mobile Tx) is specified as
(a) 824 MHz – 849 MHz
(b) 825 MHz – 845 MHz
(c) 869 MHz – 894 MHz
(d) 870 MHz – 890 MHz

cluster pattern. Assuming the uniform distribution of channels, the number of voice channels in a cell is
(a) 7 (c) 84
(b) 12 (d) 588

S6.7 There are a total of 168 voice channels available in a cellular system configured with a 7-cell, 3-sector cluster pattern. Assuming the uniform distribution of channels in each sector, the number of voice channels in a sector is
(a) 3 (c) 8
(b) 7 (d) 12

S6.8 There are total 120 voice channels available in a cellular system configured with a 4-cell, 6-sector cluster pattern. Assuming the uniform distribution of channels in each sector, the number of voice channels in a sector is
(a) 5 (c) 30
(b) 20 (d) 120

S6.9 In the GSM cellular system, a pair of 25 MHz band is allocated to the uplink and downlink channel to provide full duplex transmission. Each radio carrier uses a 200-kHz bandwidth. If th

Analytical Problems

P12.1 A CDMA mobile measures the received signal strength from its serving cell-site as –100 dBm. What should the mobile transmitter power be set to as a first approximation?

P12.2 Calculate the time for one complete repetition of the Walsh code on the forward channel. Find the Walsh functions for a 32-bit code.

P12.3 Determine the maximum number of mobile users that can be supported in a single cell CDMA system using
(a) omnidirectional cell-site antenna and no voice-activity detection
(b) three-sectors at the cell-site and voice-activity detection factor, $v = 0.75$.
Assume the CDMA system is interference limited. The system uses an RF bandwidth of 1.25 MHz to transmit data @ 9.6 kbps, and a minimum acceptable E_b/N_0 is found to be 10 dB. Use typical values for other factors.

P12.4 The cellular spectral efficiency η in a CDMA system is defined as the total number of bits/s/Hz transmitted by all mobile users in a cell. For a QPSK modulation scheme used in CDMA, assume that the spectral efficiency of a single CDMA user is $2/Q$ bits/s/Hz, where Q is the length of the spreading code. Develop an expression for η that depends on the received I_o/N_o, SINR, and σ, if the receiver requires a specified SINR.

P12.5 Compare the capacity of 2G digital cellular systems: IS-136 TDMA, GSM, and IS-95 CDMA, assuming the allocated spectrum of 1.25 MHz in each case. Assume frequency reuse factor of K = 4

Analytical Problems

A number of Analytical Numerical Problems are given at the end of each chapter for practice and revision purpose.

Benefit:
The readers will be able to critically examine and analyze the application of theoretical concepts by solving these numerical problems.

Review Questions

Each chapter contains a set of Review Questions. These review questions provide the essence of the concepts discussed in the chapter. These can also serve the purpose of preparing assignment/test exercise questions.

Benefit:
This will help students in self-assessment.

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Review Questions

Q3.1 Why is propagation path loss one of the key parameters used in the analysis of radio wave propagation for mobile communication?

Q3.2 How are the service areas classified, depending on the following two criteria?
(a) Natural terrain along the propagation path
(b) Human-made structures along the propagation path

Q3.3 Explain the free-space propagation model and derive an expression for the received signal power. Make suitable assumptions as necessary.

Q3.4 Signal attenuation is greater for mobile communication than for free-space communication. State the various reasons.

Q3.5 In what ways is radio propagation on land different from that in free space?

Q3.6 Derive an expression for mobile point-to-point propagation model (two-ray model) to determine the received signal power. Explain the use of two-ray model to justify mobile radio path loss and antenna height effects.

respectively and are located at heights H_t and H_r above the ground level.

Q3.8 Describe the parameters responsible for signal loss due to foliage. How does foliage loss vary with foliage density, path lengths and frequency of transmission?

Q3.9 Describe the effect on the received signal quality due to change of
(a) cell-site antenna height
(b) location of cell-site antenna
(c) effective antenna height with change of location of mobile unit
Assume that the mobile is driven up a positive slope (up to a high spot).

Q3.10 Differentiate between area-to-area prediction model and point-to-point propagation model used for estimating radio coverage in a mobile radio communication.

Q3.11 Explain how area-to-area prediction curves can be obtained. What role does 1-km intercept point and the path-loss slope play in obtaining the area-to-area prediction curves?

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References

At the end of each chapter, a comprehensive list of useful References are listed for further studies on the related topics covered in the chapter.

Benefit:
With the use of Internet facility and available journals in the library, the students can enhance their knowledge using references given here.

Abbreviations and Acronyms

A comprehensive list of Abbreviations and Acronym is given as Appendix 1 for quick reference.

Benefit:
While studying a topic, the students can directly use this list to know the abbreviation form, if needed.

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COA	Care Of Address	DS/FHMA	Direct Sequence/Frequency Hopped Multiple Access
codec	coder/decoder	DSL	Digital Subscriber Line
COST	Cooperative for Scientific and Technical research	DSSS	Direct Sequence Spread Spectrum
CPCH	Common Packet CHannel	DST	Digital Signaling Tone
CPE	Customer Premises Equipment	DTCH	Dedicated Traffic (Transport) CHannel
CPFSK	Continuous Phase Frequency Shift Keying	DVB	Digital Video Broadcasting
CRC	Cyclic Redundancy Check	E-DCH	Enhanced Dedicated CHannel
CS	Circuit Switched	EDGE	Enhanced Data Rates for GSM Evolution (see GSM also)
CSMA	Carrier Sense Multiple Access	EGPRS	Extended GPRS (see GPRS also)
CSMA/CA	Carrier Sense Multiple Access Collision Avoidance	EIA	Electronic Industries Alliance (Association)
CSMA/CD	Carrier Sense Multiple Access Collision Detection	EIR	Equipment Identity Register
		EIRP	Effective Isotropic Radiated Power
CT2	Codeless Telephone-2	E_b/N_0	Bit Energy-to-Noise power density
CTCH	Common Traffic Channel	EPC	Electronic Product Code
CTCH	Common Transport CHannel	ERMES	European Radio MEssage System
CVSDM	Continuous Variable Slope Delta Modulation	ERP	Effective Radiated Power
		ERP-PBCC	Extended Rate Physical PBCC (see PBCC also)
DA	Destination Address		
DAB	Digital Audio Broadcasting	ESN	Equipment Serial Number
DAMPS	Digital Advanced Mobile Phone System	ESS	Extended Service Set
dB	decibel	ESSID	Extended Service Set Identifier
dBd	dB gain with respect to a half wave dipole	ETACS	European or Enhanced TACS (see TACS also)
dBi	dB gain with respect to an isotropic antenna	ETSI	European Telecommunications Standards Institute
DBPSK	Differential Binary Phase Shift Keying	EY-NPMA	Elimination Yield Non-Preemptive Multiple Access
DCA	Dynamic Channel Allocation		
DCCH	Digital Control CHannel	FA	Foreign Agent
DCF	Distributed Coordination Function	FACCH	Fast Associated Control CHannel
DCH	Dedicated channel	FACH	Forward Access Channel
DCS1800	Digital Cellular System (1800 MHz)	FAF	Floor Attenuation Factor
DECT	Digital Enhanced Cordless Telephone	FAUSCH	FAst Uplink Signaling Channel
DES	Data Encryption Standard	FBF	Feedback Filter
DFE	Decision Feedback Equaliser	FCC	Federal Communications Commission
DFWMAC	Distributed Foundation Wireless MAC (see MAC also)	FCCH	Frequency Correction Channel
		FCDMA	Hybrid FDMA/CDMA (see FDMA and

Glossary of Key Terms

A comprehensive list of Glossary of Key Terms is given.

Benefit:
The glossary of key terms will serve as a dictionary on wireless communication for the students.

Appendix 2 | **635**

AP — **Access Point**
A base station in a wireless LAN. Access points are typically stand-alone devices that plug into an Ethernet hub or server. Like a cellular phone system, users can roam around with their mobile devices and be handed off from one access point to the other.

ARQ — **Automatic Retransmission Request**
An error-correction scheme that continuously retransmits until an acknowledgment is received or timeout value is exceeded.

ASK — **Amplitude Shift Keying**
A binary modulation technique in which 1 bit has a carrier signal while a 0 bit has no carrier signal.

Attenuation — A loss of signal strength.

AuC — **Authentication Centre**
Process of verifying the identity of the user at the other end of a link. Part of HLR in 3G systems, to perform computations to verify and authenticate mobile phone users.

Authentication — A process that verifies that the client device asking to join the piconet has permission to access the network.

Authorisation — Process of deciding if a requesting device is allowed to have access to a service on another device. Authorisation always includes authentication.

Bands — Sections of the radio frequency spectrum.

Bandwidth — In the internet industry, bandwidth refers to the capacity of a connection to carry information, while in radio communications it is the range of frequencies that can be transmitted. It is also the difference in Hz between the upper and lower frequencies of a spectrum.

Baseband — A transmission technique that treats the entire transmission medium as only one channel.

Baud Rate — The number of times that a carrier signal changes per second.

BCCH — **Broadcast Control CHannel**

Beamwidth — Beamwidth of an antenna pattern is the angle between half-power points of the main lobe, referenced to the peak effective radiated power of the main lobe. Usually expressed in degrees.

BER — **Bit-Error Rate**
The probability that a transmitting bit is received in error.

Block Error Correction Code — A technique in which a k-bit block of data is mapped into an n-bit block (n, k) called a code word, using an FEC encoder,

Bluetooth — A wireless standard that enables devices to transmit data at up to 1 Mbps over a maximum distance of 10 meters.

bps — **bits per second**
The basic unit for rate of transfer of data, which signifies the number of bits that can be transmitted per second.

Broadband — A transmission technique that sends multiple signals at different frequencies. General term used for any type of technology that provides a high data rate (greater than 200 kbps up to 100s of Mbps) capability. Broadband services are usually 'always-on'. Capable of supporting a variety of voice and data applications, such as voice telephony, internet access, pay TV and multimedia services.

BS — **Base Station**
A land station at a fixed location at the centre or on the edge of a coverage region supporting radio access by mobile users to a fixed communication infrastructure.

Model Test Paper - Type 1

Wireless Communications

Max. Time: **3 Hours** Max. Marks: **100**

NOTE: Attempt any FIVE questions. All questions carry equal marks.

Q1 (a) Compare and contrast the features of second-generation digital cellular standards—GSM and CDMA technologies.

(b) What are advantages of cellular mobile communication systems over conventional mobile telephone systems? [12 + 8]

Q2 (a) A cellular architecture is configured with regular hexagonal cell geometry. The total service area is divided into cell clusters with frequency reuse. Prove that the distance D between the centres of two closest cochannel cells is given by $D = R \sqrt{\{3 (i^2 + j^2 + i \text{ x } j)\}}$; where R is the cell radius, having same units as D; and i, j are non-negative integers which describe the geometry relation between adjacent cells.

(b) A large city with an area of 1000 km^2 is required to be covered by a finite number of cells with a radius of 2 km each. How many cell-sites would be required, assuming regular hexagonal-shaped cells? [16 + 4]

Q3 (a) Explain fixed, dynamic, and hybrid channel-assignment strategies in a cellular system.

(b) A cellular mobile communication system operates at 900 MHz. Compute the propagation-path loss at a distance 5 km away from the cell-site. Assume the height of the cell-site transmitting antenna is 50 m and the mobile receiving antenna is 1.5 m above ground. [10 + 10]

Q4 (a) Suggest at least five different means to increase the radio coverage of a cell.

(b) On what basis are multiple access radio protocols classified? Distinguish between various types of multiple access protocols. [10+10]

Q5 (a) Describe the step-by-step procedure for placing a call from a calling mobile subscriber to a called landline telephone subscriber.

(b) In a US AMPS cellular system, a service provider is allocated a total spectrum bandwidth of 10 MHz. Out of this, 1 MHz spectrum is reserved for signaling and control channels. If a 7-cell reuse pattern is used, calculate the radio capacity of a cell. [12 + 8]

Q6 (a) Explain the main properties of the basic multiple access techniques—FDMA, TDMA, and CDMA.

(b) Adaptive power control is the solution to the near–far interference problem in a CDMA cellular system. Describe two basic concepts of power control employed for CDMA. [8 + 12]

Model Test Papers

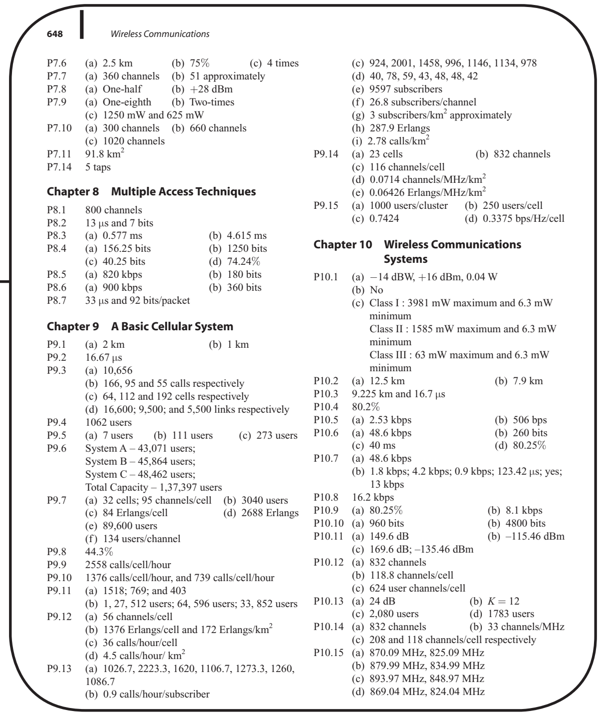
A set of different types of Model Test Papers as per the pattern followed in various universities is provided at the end of the book. The hints to theory questions with reference to text and answers to numerical problems are also given.

Benefit:
The students will be able to get an idea about the composition of exam question paper.

Answers to Analytical Problems

Answers to all the Analytical Problems are given at the end of the book.

Benefit:
This will help in cross-checking the solutions.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P7.6 (a) 2.5 km (b) 75% (c) 4 times
P7.7 (a) 360 channels (b) 51 approximately
P7.8 (a) One-half (b) +28 dBm
P7.9 (a) One-eighth (b) Two-times
(c) 1250 mW and 625 mW
P7.10 (a) 300 channels (b) 660 channels
(c) 1020 channels
P7.11 91.8 km^2
P7.14 5 taps

Chapter 8 Multiple Access Techniques

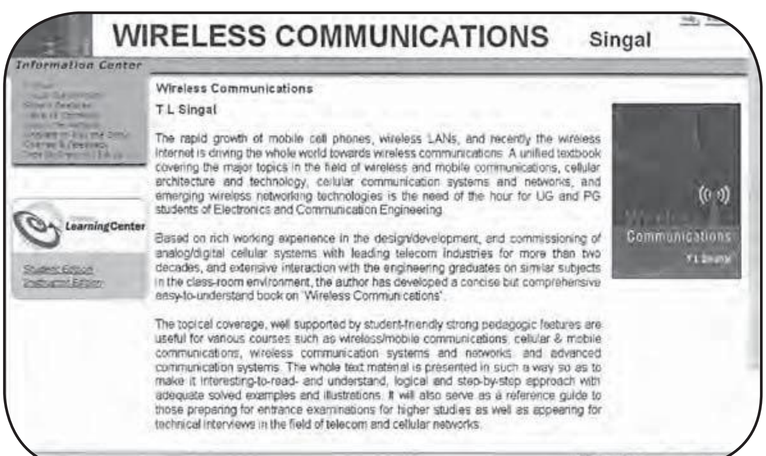
P8.1 800 channels
P8.2 13 μs and 7 bits
P8.3 (a) 0.577 ms (b) 4.615 ms
P8.4 (a) 156.25 bits (b) 1250 bits
(c) 40.25 bits (d) 74.24%
P8.5 (a) 820 kbps (b) 180 bits
P8.6 (a) 900 kbps (b) 360 bits
P8.7 33 μs and 92 bits/packet

Chapter 9 A Basic Cellular System

P9.1 (a) 2 km (b) 1 km
P9.2 16.67 μs
P9.3 (a) 10,656
(b) 166, 95 and 55 calls respectively
(c) 64, 112 and 192 cells respectively
(d) 16,600; 9,500; and 5,500 links respectively
P9.4 1062 users
P9.5 (a) 7 users (b) 111 users (c) 273 users
P9.6 System A – 43,071 users;
System B – 45,864 users;
System C – 48,462 users;
Total Capacity – 1,37,397 users
P9.7 (a) 32 cells; 95 channels/cell (b) 3040 users
(c) 84 Erlangs/cell (d) 2688 Erlangs
(e) 89,600 users
(f) 134 users/channel
P9.8 44.3%
P9.9 2558 calls/cell/hour
P9.10 1376 calls/cell/hour, and 739 calls/cell/hour
P9.11 (a) 1518; 769; and 403
(b) 1, 27, 512 users; 64, 596 users; 33, 852 users
P9.12 (a) 56 channels/cell
(b) 1376 Erlangs/cell and 172 Erlangs/km^2
(c) 36 calls/hour/cell
(d) 4.5 calls/hour/ km^2
P9.13 (a) 1026.7, 2223.3, 1620, 1106.7, 1273.3, 1260, 1086.7
(b) 0.9 calls/hour/subscriber
(c) 924, 2001, 1458, 996, 1146, 1134, 978
(d) 40, 78, 59, 43, 48, 48, 42
(e) 9597 subscribers
(f) 26.8 subscribers/channel
(g) 3 subscribers/km^2 approximately
(h) 287.9 Erlangs
(i) 2.78 calls/km^2
P9.14 (a) 23 cells (b) 832 channels
(c) 116 channels/cell
(d) 0.0714 channels/MHz/km^2
(e) 0.06426 Erlangs/MHz/km^2
P9.15 (a) 1000 users/cluster (b) 250 users/cell
(c) 0.7424 (d) 0.3375 bps/Hz/cell

Chapter 10 Wireless Communications Systems

P10.1 (a) −14 dBW, +16 dBm, 0.04 W
(b) No
(c) Class I : 3981 mW maximum and 6.3 mW minimum
Class II : 1585 mW maximum and 6.3 mW minimum
Class III : 63 mW maximum and 6.3 mW minimum
P10.2 (a) 12.5 km (b) 7.9 km
P10.3 9.225 km and 16.7 μs
P10.4 80.2%
P10.5 (a) 2.53 kbps (b) 506 bps
P10.6 (a) 48.6 kbps (b) 260 bits
(c) 40 ms (d) 80.25%
P10.7 (a) 48.6 kbps
(b) 1.8 kbps; 4.2 kbps; 0.9 kbps; 123.42 μs; yes; 13 kbps
P10.8 16.2 kbps
P10.9 (a) 80.25% (b) 8.1 kbps
P10.10 (a) 960 bits (b) 4800 bits
P10.11 (a) 149.6 dB (b) –115.46 dBm
(c) 169.6 dB; –135.46 dBm
P10.12 (a) 832 channels
(b) 118.8 channels/cell
(c) 624 user channels/cell
P10.13 (a) 24 dB (b) $K = 12$
(c) 2,080 users (d) 1783 users
P10.14 (a) 832 channels (b) 33 channels/MHz
(c) 208 and 118 channels/cell respectively
P10.15 (a) 870.09 MHz, 825.09 MHz
(b) 879.99 MHz, 834.99 MHz
(c) 893.97 MHz, 848.97 MHz
(d) 869.04 MHz, 824.04 MHz



Website

A Website accompanying this book is available. It can be accessed at www.mhhe.com/singal/wc. The Website includes Solution Manual and Power Point Lecture Slides for instructors and an Interactive Quiz for students.

1

Evolution of Wireless Communication Systems

The term 'wireless' is often used to describe all types of devices and technologies that use space as a signal-propagating medium, and are not connected by a wire or cable. Wireless communication may be defined as the transmission of user information without the use of wires. The user information could be in the form of human voice, digital data, e-mail messages, video and other services. Wireless communication is revolutionising almost every aspect of our daily lives. From the ubiquitous mobile phone to automated inventory counting in large retail stores to remote wireless sensors installed in inaccessible locations, wireless-communication technology is poised to continue expanding at a very fast pace. Using wireless communications to send and receive messages, browse the Internet, and access corporate databases anywhere, any time across the world has already become common. A wide array of devices ranging from computers to digital cameras, laser printers, and even household appliances can already communicate without wires. We are moving towards a wireless world to live in a global village. Starting with a brief history of wireless communications, this chapter provides a quick overview of comparison and applications of wireless communications and systems. Knowing about wireless network generations is important in studying the evolution to next-generation networks. This chapter concludes with potential market areas and challenges for research in the field of wireless and mobile communications.

1.1 BRIEF HISTORY OF WIRELESS COMMUNICATIONS

To appreciate the current technology and prepare ourselves to enhance its development in future, it is always interesting to have a quick glance at the history of a technology such as wireless communications. There are always several smaller steps that take place in leading up to the development of a new technology. Tracing the development of these earlier discoveries in brief can help us better understand how this technology actually functions and contributes towards what could be the next development. A brief review of the history of wireless communications covering radio, television, radar, satellite, wireless and mobile, cellular and other wireless networks are presented here.

1.1.1 Radio and Television Communications

In 1874, Marconi performed simple experiments to send signals using electromagnetic waves at short distances of only about 100 metres. Scientists and other experts at that time believed that electromagnetic waves could only be transmitted in a straight line, and the main obstacle to radio transmission was the curvature of the earth's surface. Marconi successfully experimented to prove that electromagnetic wave transmission was possible between two distant points even through obstacles in between.

This paved the way for wireless telegraphy, also known as *radio communications*. The word 'radio' comes from the term 'radiated energy'. In 1901, Marconi set up a transmitting station in England, and a receiving station with larger types of antennas suspended from light kites on the other side of the Atlantic Ocean on the island of Newfoundland. For three hours every day, a signal was transmitted and received at a distance of about 3,500 kilometres! This was due to the presence of the ionosphere, a layer of the upper atmosphere between 60 to 500 kilometres above the earth, which was discovered by an English physicist, Edward Appleton, in 1924. The ionosphere reflects electromagnetic waves which allow a radio signal to travel far distances and is fundamental to all long-distance wireless radio communications. In 1935, Marconi performed distance-based search experiments that eventually led to the invention of radar.

Marconi also studied microwaves and early television technology. In 1927, Farnsworth gave the first public demonstration of the television system, and developed several of the basic concepts of an electronic television system. North America's first television station, W3XK in Wheaton, Maryland, was started in the 1930s. By 1939, widespread commercial electronic television broadcasting started in the United States. The National Broadcasting Company (NBC) started regularly scheduled broadcasts in the New York area to only 400 TV sets. In 1941, the American Federal Communications Authority set the standards for broadcast television. By 1970, television had become the primary information and entertainment medium in the world. Today, it is estimated that there are more than a billion television sets worldwide. However, the standards for television broadcasting are not universal. There are 15 different variations of broadcasting standards used around the world.

Facts to Know!

The word 'radio' originated from the term 'radiated energy'. Radio waves were originally called Hertzian waves, named after the German physicist Heinrich Hertz, who discovered radio waves which are a form of electromagnetic radiation.

1.1.2 Radar Communications

Radar has been recognised as one of the greatest scientific developments of the first half of the 20^{th} century. The development of radar dates back to the discoveries of the 1860s and 1870s, when James Maxwell developed the equations that outlined the behavior of electromagnetic waves, and Heinrich Hertz discovered radio waves. The first successful radio range-finding experiment occurred in 1924, when the British scientist Edward Appleton used radio echoes to determine the height of the ionosphere. The first practical radar system was produced in 1935 by the British physicist Robert Watson-Watt. By 1939, England had established a chain of radar stations along its southern and eastern coasts to detect aggressors in the air or on the sea.

Radar is an active remote-sensing system that operates on the principle of echoes. A radar display shows a map-like picture of the area being scanned. The centre of the picture corresponds to the radar antenna and the radar echoes are shown as bright spots on the screen. Although radar is usually associated with detecting airplanes in the sky or ships on the ocean, it is actually used in a variety of different ways such as to forecast the weather, to scan entire regions for possible archaeological sites from space satellites and airplanes, to study potential hidden dangers in highway tunnels, to locate stagnant pools of water in areas of dense foliage on the earth, and to provide information about the universe. A Doppler radar is being used today by meteorologists to locate tornados and microbursts, which are downdrafts of air traveling at very high speeds. Doppler radar is also used by law-enforcement agencies to locate speeding motorists.

Facts to Know!

Radar was originally called Radio Direction Finder (RDF). The term RADAR was coined in 1941 as an acronym for RAdio Detection And Ranging. The term has since entered the English language as a standard word, radar, losing the capitalisation.

1.1.3 Satellite Communications

A satellite is any object that orbits or revolves around another object. For example, the moon is a satellite of the earth, and the earth is a satellite of the sun. Man-made satellites provide communication capabilities around the world, transmitting television signals, telephone calls, faxes, computer communications, and weather information. Satellites can be sent into space through a variety of launch vehicles. The theory of satellites dates back to 325 years before the first man-made satellite was ever launched. Sir Isaac Newton in the 1720s was probably the first person to conceive the idea of a satellite. Newton illustrated how an artificial satellite could be launched from the earth. He pictured the earth as a high mountain and a cannon on top of the mountain firing shots parallel to the ground. During World War II, the German military made great strides in the development of rocket technology.

In 1945, Arthur C Clarke, a science-fiction author, envisioned a network of communication satellites. Three satellites could be placed into space at about 36,000 kilometres above sea level so as to orbit the planet every 24 hours. These satellites would be able to transmit signals around the world by transmitting in a line-of-sight direction with other orbiting satellites. In 1957, the Soviet Union launched the Sputnik 1 satellite, followed by Sputnik 2 and its passenger, Laika, a dog who has the distinction of being the first living creature to enter the earth's orbit. In 1961, Yuri Gagarin became the first human in orbit. In 1964, an international organisation known as Intelsat was formed, which launched a series of satellites with the goal of providing total earth coverage by satellite transmission. This was achieved by 1969.

Today, Intelsat has 19 satellites in orbit that are open to use by all nations. The Intelsat consortium owns the satellites, but each country owns their earth-receiving stations. The explosive popularity of cellular telephones advanced the idea of always being connected everywhere on the earth. Several companies committed themselves to providing a solution by using satellites in low earth orbit (LEO) at a height of about 650 kilometres. Iridium, sponsored by Motorola, planned to launch 66 satellites into the polar orbit to provide communications services to hand-held phones around the world in 1998.

Facts to Know!

Aryabhatta was India's first satellite, named after the great Indian astronomer. It was built by the Indian Space Research Organisation (ISRO) to conduct experiments related to astronomy. It was launched by the Soviet Union on 19th April 1975 using a Cosmos-3M launch vehicle.

1.1.4 Wireless and Mobile Communications

Based on the nature of wireless transmission, wireless communication systems may be classified as simplex, half-duplex or full-duplex. In *simplex wireless systems*, separate transmitters and receivers operate at the same frequency and communication is possible in only one direction from the transmitter to the receiver at any time. For example, paging and messaging systems are simplex wireless communication systems in which short text or alphanumeric messages are transmitted by fixed paging transmitters and received pagers but the received messages are not acknowledged. *Half-duplex wireless systems* allow two-way communication but a subscriber can only transmit or receive voice information at any given time. The same frequency is used for both transmission and reception, with a push-to-talk feature for enabling transmission only at a time. Walkie-talkie wireless communication sets used by police and paramilitary forces are the examples of half-duplex wireless systems.

ull-duplex wireless communication systems allow simultaneous radio transmission and reception between the calling and called subscribers of the system, either directly or via a base station. They use separate frequency channels (frequency division duplex, or FDD) or different time slots on a single radio channel (time division duplex or TDD) for communication to and from the subscriber. At the base station, separate transmit and receive antennas are used to accommodate the separate forward and reverse frequency channels. However, at the subscriber unit, a single antenna is used for both transmission and reception simultaneously. A device called a *duplexer* is used inside the subscriber unit to enable the same antenna to be used for this purpose. In order to provide sufficient isolation in the duplexer, the transmit and receive frequencies are generally separated by about 5% of the nominal RF frequency of operation. Full-duplex mobile communication systems provide many of the capabilities of the standard telephone for voice communication, with the added convenience of communication on the move.

In Time Division Duplexing (TDD), a portion of the time is used to transfer information data from the base station to the mobile subscriber, and the remaining time is used to transfer information data from the mobile subscriber to the base station on the same frequency channel. Digital transmission formats and digital modulation schemes are used in TDD. It is very sensitive to timing accuracies and needs synchronisation between transmissions and reception of the data at the transmitter and receiver ends respectively. Therefore, TDD has limited applications such as indoor or small-area wireless applications where the physical coverage distances are much smaller than those encountered in conventional cellular telephone systems so as to keep the radio propagation delay within acceptable limits. In the 1930s and 1940s, two-way full-duplex vehicle radios were installed and used by police, utility companies, government agencies, and emergency services.

Facts to Know!

All wireless communication systems need not be mobile (e.g., TVs, microwaves, radios, satellites) whereas all mobile communication systems (e.g., cellularphones) use necessarily wireless communications.

Generally, wireless communications could be simplex, semi-duplex or full-duplex; whereas mobile communication is mostly full-duplex.

1.1.5 Cellular Communications

In 1946, AT&T introduced the first American commercial mobile radio telephone service to private customers. It consisted of a central transmitter with one antenna which could serve a wide area. However, this system could not be used with mobiles because of their limited transmitter power. To overcome this limitation, smaller receivers with antennas were placed on top of buildings and on poles around the city, creating smaller cells. When a person used his mobile, the conversation that he heard was transmitted on one frequency by the central transmitter to the moving vehicle. When the user spoke on his mobile, however, that transmission was sent on a separate frequency that the nearest receiver antenna picked up. The major limitations were availability of limited spectrum and interference. The concepts of using smaller cells and frequency reuse laid the foundation for cellular telephones. In 1969, the Bell System developed a commercial cellular radio operation using frequency reuse.

The first modern cellular telephone systems in the early 1980s used 666 channels. Advanced Mobile Phone Service (AMPS) began setting up analog cellular telephone operations in many parts of the world. Roaming from one city or state in the United States was easy because the US system was based on an analog cellular system. In contrast, it was almost impossible to roam in Europe. During the 1980s, a plan was launched to create a single pan-European digital mobile service with advanced features and easy roaming. This network started operating in 1991.

Cellular mobile communication systems provide full-duplex communication, in which a pair of simplex RF channels with a fixed and known frequency separation (called duplex spacing) is used to define a specific radio channel in the system. For example, in the US AMPS standard and European GSM cellular standard, the forward channel has a frequency that is exactly 45 MHz more than that of the reverse channel. The channel used to transfer traffic data to the mobile subscriber from a base station is called the *forward channel*. The channel used to transfer traffic from the mobile subscriber to the base station is called the *reverse channel*.

1.1.6 Transition from Analog to Digital Systems

In the 1980s, most mobile cellular systems were based on analog design. The GSM system can be considered as the first digital cellular system. The different reasons that explain this transition from analog to digital technology are the following:

> **Facts to Know!**
>
> Today cellular telephone deployment is worldwide, but technology development remains concentrated in Scandinavia, the United States, Europe, and Japan.

System Capacity Cellular systems experienced a very significant growth in the 1980s. Analog systems were not able to cope with this increasing demand. In order to overcome this problem, new frequency bands were allocated for the development of mobile cellular radio and new modulation and coding technologies were introduced. The digital radio was, therefore, the best option to handle the capacity needs in a cost-efficient way.

Quality Aspects The quality of the service can be considerably improved using a digital technology rather than an analog one. In fact, analog systems carry the physical disturbances in radio transmission such as fades, multipath reception, spurious signals or interferences to the receiver. These disturbances reduce the quality of the communication because they produce effects such as fadeouts, crosstalks, hisses, etc.

On the other hand, digital systems avoid these effects, transforming the signal into bits. This transformation combined with other techniques, such as digital coding, improve the quality of the transmission. The improvement of digital systems compared to analog systems is more noticeable under difficult reception conditions than under good reception conditions.

Compatibility with Other Systems such as ISDN The decision of adopting a digital technology for GSM was made in the course of developing the standard. During the development of GSM, the telecommunications industry converted to digital methods. The ISDN network is an example of this evolution. In order to make GSM compatible with the services offered by ISDN, it was decided that the digital technology was the best option. Additionally, a digital system allows, easily than an analog one, the implementation of future improvements and the change of its own characteristics.

> **Facts to Know!**
>
> Analog systems do not fully utilise the signal between the phone and the cellular network because analog signals cannot be compressed and manipulated as easily as a true digital signal. This is the reason why many service providers have switched to digital systems.

1.2 ADVANTAGES OF WIRELESS COMMUNICATIONS

There are many advantages of wireless communications, using wireless communications technology and wireless networking, as compared to wired communications and networks. Some of the major advantages include mobility, increased reliability, ease of installation, rapid disaster recovery and above all lower cost.

Mobility The primary advantage of wireless communications is to offer the user freedom to move about while remaining connected with the network within its coverage area. Many business categories, such as the police department, require its workforce to be mobile instead of fixed at one location. Wireless technology enables many industries to shift toward an increasingly mobile workforce, whether they are in meetings or working on a factory floor or conducting research.

Increased Reliability The most common source of network problems is the failure or damage of network cables due to environment conditions or erosion of metallic conductors. A cable splice that is done incorrectly can cause unexplainable errors and is very difficult to identify. Using wireless technology not only eliminates these types of cable failures, but also increases the overall reliability of the network.

Facts to Know!

Notebook computers and other portable handheld wireless devices allow team members an added collaborative convenience of immediate access to the company's WLAN.

Ease of Installation Wireless communications and networks make it easier for any office to be modified with new cubicles or furniture, without worrying about providing network connectivity through cables. There is no need to first consider the location of the computer jack in the wall when relocating furniture. Instead, the focus can be on creating the most effective work environment without any delay and hassles. The time required to install network cabling may take days or even weeks to complete, thereby disrupting the whole work. Using a wireless LAN eliminates such disruption.

Rapid Disaster Recovery Accidents may happen due to fires, tornados, and floods at any possible location, and that too without any prior warning. Any organisation that is not prepared to recover from such natural disasters will find itself quickly out of business. Since the computer network is a vital part of the daily operation of a business, the ability to have the network up and immediately working after a disaster is critical. A documented disaster recovery plan is a must.

Facts to Know!

Many businesses are nowadays using wireless communications and networking, keeping laptop computers with wireless NICs and access points in reserve along with backup network servers. In the event of a disaster, the office can be quickly relocated and made operational, without any need of finding a new facility with network wiring.

Lower Cost Of course, eliminating the need to install cabling and using wireless communications results in significant cost savings. Installing network cabling in older buildings can be an extremely difficult, slow, and costly task. Facilities constructed prior to the mid-1980s were built without any thought given to running computer wiring in each room. In such cases a wireless LAN is the ideal solution. Historical buildings would be preserved, harmful asbestos would not be disturbed, and difficult drilling could be avoided by using a wireless system.

1.3 DISADVANTAGES OF WIRELESS COMMUNICATIONS

Along with the many advantages of wireless communications and technology, there are some disadvantages and concerns. The most prominent of them are radio signal interference, security issues, and health hazards.

Facts to Know!

External interference from AM or FM radio stations, TV broadcast stations, or microwave and cellular transmitters does not occur because they operate at different frequencies and power levels than wireless LANs.

Radio Signal Interference Signals from other wireless devices can disrupt its radio transmission, or a wireless device may itself be a source of interference for other wireless devices. For example, several commonly used office wireless devices such as cordless telephones, microwave ovens, elevator motors, and other heavy electrical manufacturing machines, transmit radio signals that may interfere with a wireless LAN operation. These may cause errors to occur in the transmission between a wireless device and an access point. Similarly, Bluetooth and WLAN devices both operate in the same radio frequency, potentially resulting in interference between such devices.

Security A wireless communication device transmits radio signals over a wide open area, and hence security becomes a major concern. It is possible for an intruder with a notebook computer and wireless NIC to intercept the signals from a nearby wireless network. Because much of business network traffic may contain sensitive information, this becomes a serious concern for many users.

Some wireless technologies can provide added levels of security with authorisation features prior to gaining access to the network. Network administrators can also limit access for approved wireless devices only. As further protection, data transmitted between the wireless device and the access point can also be encrypted in such a way that only the intended recipient can decode the message.

Health Hazards High power levels of RF energy can produce biological damage. However, it is not yet established accurately as to how much levels of RF can cause adverse health effects. But continuous radiations even at lower levels can be harmful to sensitive body organs. Radio transmitters in wireless communications devices emit radio frequency (RF) energy. Typically, these wireless devices emit low levels of RF while being used. Although some research has been done to address these issues, no clear facts of the biological effects of this type of radiations have emerged to date.

Facts to Know!

Questions exist regarding the safety of handheld cellular phones, the kind with a built-in antenna that is positioned very close to the user's head during normal conversation.

The safety of cordless phones, which have a base unit connected to the telephone wiring in a house and which operate at far lower power levels and frequencies, has never been questioned. It is always wise to be aware of the health concerns and to monitor ongoing scientific research, even though the available science does not conclude either way about the safety of wireless mobile devices.

1.4 WIRELESS NETWORK GENERATIONS

The cellular systems have been classified into three distinct evolution of generations: The first-generation (1G) analog cellular communication systems are voice-oriented analog cellular systems using frequency division multiple access technique. The first-generation systems used large cells and omni-directional antennas in the 800-MHz band. The AMPS and ETACS systems use a seven-cell reuse pattern with provisions for cell-sectoring and cell-splitting to increase capacity when needed. These systems provide satisfactory voice communication on the move with limited traffic-handling capacity.

1.4.1 First-Generation Analog Cellular Systems

The first-generation cellular systems are based on analog transmission technology. The most popular first-generation cellular systems are AMPS (widely deployed in most parts of US, South America, Australia, China), and ETACS (deployed throughout Europe). The systems transmit speech signals employing FM, and important control information is transmitted in digital form using FSK. The entire service area is divided into logical cells, and each cell is allocated one specific band in the frequency spectrum. To explore a frequency reuse pattern, the frequency spectrum is divided among seven cells, improving the voice quality as each subscriber is given a larger bandwidth.

AMPS and ETACS cellular radio systems deploy cell-sites with tall towers that support several receiving antennas and have transmitting antennas that typically radiate a few hundred watts of effective radiated power. Each cell-site has one control channel transmitter that broadcasts on the forward control channel, one control channel receiver that listens on the reverse control channel for any mobile phone to set-up a call, and eight or more FM duplex voice channels.

Facts to Know!

The channel bandwidth in AMPS is 30 kHz in an 800-MHz spectrum whereas the channel bandwidth in ETACS is 30 kHz in a 900-MHz spectrum.

Table 1.1 shows the worldwide 1G analog cellular systems. All these systems use two separate frequency bands for forward (from cell-site to mobile) and reverse (from mobile to cell-site) links. Such a system is referred to as a *frequency division duplex (DD) scheme.* The typical allocated overall band in each direction, for example, for AMPS, and NMT-900, is 25 MHz in each direction. The dominant spectra of operation for these systems are the 800-and 900-MHz bands. In an ideal situation, all countries should use the same standard and the same frequency bands. However, in practice, as shown in Table 1.1, a variety of frequencies and standards are adopted all over the world.

Table 1.1 *Existing 1G analog cellular systems*

Standard	*orward frequency band (MHz)*	*Reverse frequency band (MHz)*	*Channel spacing (kHz)*	*Number of channels*	*Multiple access/ Modulation technique*	*Major region of operation*
AMPS	824–849	869–894	30	832	FDMA/FM	US
ETACS	872–905	917–950	25	1240	FDMA/FM	UK
NMT 900	890–915	935–960	12.5	1999	FDMA/FM	EU
JTACS/ NTACS	915–925 898–901 918.5–922	860–870 843–846 863.5–867	25/12.5 25/12.5 12.5	400/800 120/240 280	FDMA/FM	Japan

AMPS Advanced Mobile Phone System
ETACS Enhanced Total Access Communication System
NMT Nordic Mobile Telephone
JTACS Japanese Total Access Communication System; NTACS: Narrowband JTACS

All the 1G cellular systems use analog frequency modulation (FM) for which the transmission power requirement depends on the transmission bandwidth. On the other hand, power is also related to the coverage and size of the cells. Therefore, one can compensate for the reduction in transmission bandwidth per subscriber by reducing the size of a cell in a cellular network. Reduction in size of the cell increases the number of cells and the cost of installation of the infrastructure. The channel spacing, or bandwidth, allocated to each subscriber is either 30 kHz or 25 kHz or a fraction of either of them.

1.4.2 Second-Generation Digital Cellular Systems

First-generation analog cellular systems were followed by second-generation digital cellular systems. The second-generation (2G) cellular systems represent the set of wireless air interface standards that rely on digital modulation and sophisticated digital signal processing in the handset and the base station. Digital cellular technologies support a much larger number of mobile subscribers within a given frequency allocation, thereby offering higher user capacity, providing superior security and voice quality, and lay the foundation for value-added services (including data) that will continue to be developed and enhanced in future. To have efficient use of the frequency spectrum, time division or code-division multiple access technique is used in 2G digital cellular systems so that low-rate data along with voice can be processed. Table 1.2 summarises the major 2G digital cellular standards.

There are four major standards in this category: the North American Interim Standard (IS-54) that later on improved into IS-136; GSM, the pan-European digital cellular; and Personal digital cellular (PDC) — all of them using TDMA technology; and IS-95 in North America, which uses CDMA technology. Like the 1G analog cellular systems, the 2G digital cellular systems are all FDD and mostly operate in the 800- and 900-MHz bands. The carrier spacing of IS-54/136 and PDC is the same as the carrier spacing of 1G analog cellular systems in their respective regions, but GSM and IS-95 use multiple analog channels to form one digital carrier.

The most popular 2G cellular standards include three TDMA standards and one CDMA standard. Interim Standard 54 or 136 (IS-54 or IS-136), also known as US Digital Cellular (USDC), which supports three time-slotted mobile subscribers for each 30-kHz radio channel in both the cellular 800 MHz and PCS 1900 MHz

Table 1.2 *2G digital cellular standards*

Standard	*orward frequency band (MHz)*	*Reverse frequency band (MHz)*	*Multiple access technique*	*Major region of operation*
IS-54/136	869–894	824–849	TDMA/FDD	US
GSM	935–960	890–915	TDMA/FDD	Europe/Asia
PDC	940–956	810–826	TDMA/FDD	Japan
IS-95	869–894	824–849	CDMA/FDD	US/Asia

bands. Based on the analog AMPS cellular system, the TDMA system IS-54/136 was developed in the US that adds digital traffic channels. IS-54/136 uses dual-mode mobile phones and incorporates associated control channels, authentication procedures using encryption, and mobile assisted handoff. The IS-136 includes digital control channels which enable to provide several additional services such as identification, voice mail, SMS, call waiting, group calling, etc. The USDC systems share the same frequency spectrum, frequency reuse plan, and cell-sites as that of AMPS.

Global System for Mobile (GSM), which supports eight time slotted mobile subscribers for each 200-kHz radio channel in both the cellular and PCS bands; and Pacific Digital Cellular (PDC), a Japanese TDMA standard that is similar to IS-136, are the other two most popular TDMA-based digital cellular standards. The popular 2G CDMA standard (IS-95), also known as cdmaOne, can support up to 64 mobile subscribers that are orthogonally coded and simultaneously transmitted on each 1.25 MHz channel.

Facts to Know!

Three primary benefits of 2G cellular networks over their predecessors are that phone conversations are digitally encrypted, 2G systems are significantly more efficient on the spectrum allowing for far greater mobile phone penetration levels; and 2G introduced data services for mobile, starting with SMS text messages.

The speech-coding technique of all 2G systems operates at about 10 kbps. It is assumed that large cell sizes and a large number of subscribers per cell are available, which necessitates lower speech-coding rates. The peak transmission power of the mobile terminals can be between several hundreds of mW up to 1W, and on the average they consume about 100 mW. All of these systems employ centralised power control, which reduces battery power consumption and helps in controlling the interference level. In digital communications, information is transmitted in packets or frames. The duration of a packet/frame in the air should be short enough, so that the channel does not change significantly during the transmission, and long enough, so that the required time interval between packets is much smaller than the length of the packet. A frame length of around 5 to 40 ms is typically used in 2G cellular networks.

GSM supports eight users in a 200-kHz band; IS-54 and JDC support three users in 30 and 25-kHz bands, respectively. In other words, GSM uses 25 kHz for each user, IS-54 uses 10 kHz per user, and JDC uses 8.33 kHz per user. Therefore, GSM supports 2.5 times less number of subscribers in the given bandwidth. The number of users for CDMA depends on the acceptable quality of service; therefore, the number of users in the 1,250 kHz CDMA channels cannot be theoretically fixed. But this number is large enough to convince the standards organisation to adopt CDMA technology for next-generation 3G systems.

1.4.3 Evolution from 2G to 3G Cellular Networks

There are two steps of 3G evolution paths from present 2G technologies based on GSM and IS-95 CDMA respectively. An evolution path from second generation digital cellular GSM network to third generation network is depicted in Fig. 1.1.

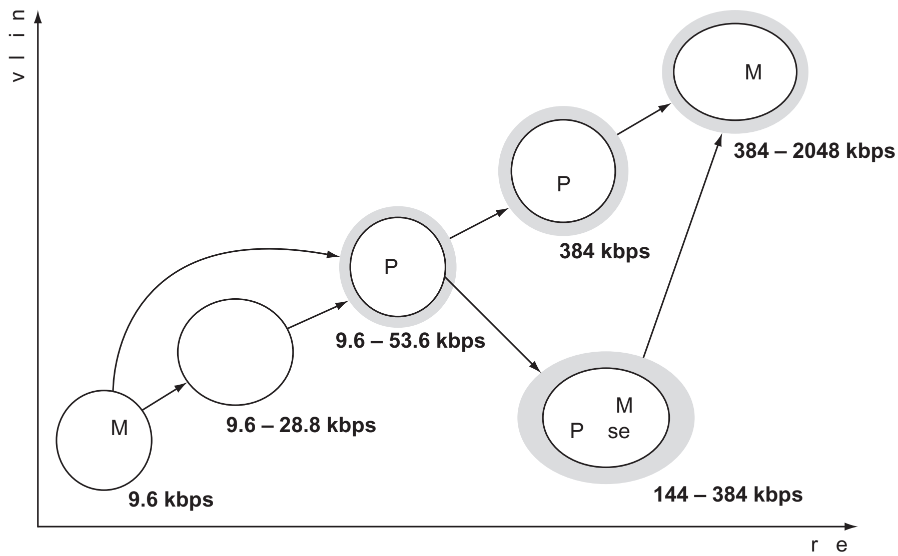


Fig. 1.1 *An evolution path from GSM to 3G network*

GSM is an open, digital cellular technology which supports voice calls and data transfer speeds of up to 9.6 kbps, together with the transmission of SMS (Short Message Service). GSM operates in the 900 MHz and 1.8 GHz bands in Europe and the 850 MHz and 1.9 GHz bands in the US. GSM provides international roaming capability that enables users to access the seamless services when travelling abroad. HSCSD (High Speed Circuit Switched Data) enables data to be transferred more rapidly than the standard GSM system by using multiple channels. GPRS is a very widely deployed wireless data service, available now with most GSM networks. GPRS offers throughput rates of up to 53.6 kbps, so that users have a similar access speed to a dial-up modem, but with the convenience of being able to connect from almost anywhere. Further enhancements to GSM networks are provided by Enhanced Data rates for GSM Evolution (EDGE) technology or EGPRS, which offers up to three times the data capacity of GPRS. Various mobile data services such as multimedia messaging, high-speed Internet access and e-mail are possible on the move. EDGE allows it to be overlaid directly onto an existing GSM network with simple software-upgrade. WCDMA is the air interface for third-generation mobile communications systems. It enables the continued support of voice, text and MMS services in addition to richer mobile multimedia services. UMTS offers data speeds up to 384 kbps on the move and 2.048 Mbps stationary. Chapters 11 and 13 gives detailed descriptions of GSM based cellular technologies.

> **Facts to Know!**
>
> The range of GSM and CDMA technology is different, and they also have different rates and modulation schemes, and that is why handsets are different between the two technologies. GSM uses SIM cards, whereas CDMA based phones do not

Besides GSM, CDMA is the most popular mobile communication standard. The initial evolution of CDMA started in 1991 as IS-95A cdmaOne 2G digital cellular technology for voice communication as well as data and multimedia services because it could allow multiple users to communicate within the spectrum, avoiding interference or jamming among users. Code division ensures that each user's signal remains separate in the spectrum. An evolution path from second generation digital cellular CDMA networks to third generation networks is depicted in Fig. 1.2.

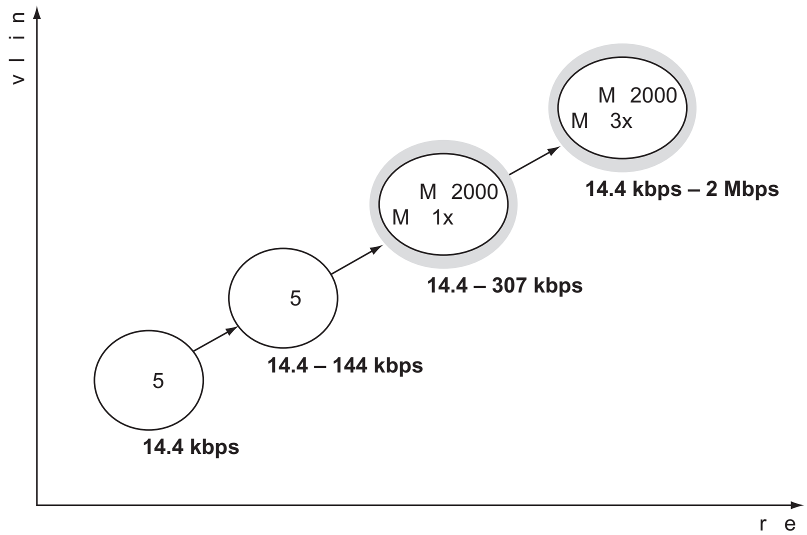


Fig.1.2 *An evolution path from CDMA to 3G network*

IS-95A describes the structure of the wideband 1.25 MHz CDMA channels, power control, call processing, hand-offs, and registration techniques for system operation. In addition to voice services, many IS-95A operators provide circuit-switched data connections at 14.4 kbps. The IS-95B or cdmaOne, categorised as a 2.5G technology, defines a compatibility standard for 1.8 to 2.0 GHz CDMA PCS systems, offers up to 144 kbps packet-switched data, in addition to voice services. CDMA2000 Multi-Carrier (MC) delivers improved system capacity and spectrum efficiency over 2G systems and it supports data services at minimum transmission rates of 144 kbps in mobile (outdoor) and 2 Mbps in fixed (indoor) environments. Chapters 12 and 13 gives the detailed description of CDMA-based cellular technologies.

1.4.4 Third-Generation Digital Cellular Systems

The fundamental purpose of the 3G mobile communications system is to provide a globally integrated wireless communication system combining different incompatible network technologies already deployed across the world. All 2G and 2.5G cellular communications systems and mobile phones will eventually evolve towards a global standard, which is referred to IMT-2000. While no one common standard for the air interface has been approved, the number of different standard specifications includes one FDMA standard, one TDMA standard, and one CDMA standard with three variations. The IMT-2000 system incorporates three variations of CDMA. The modes differ in how duplexing is accomplished and how many carriers are used. All variations operate in a 5-MHz channel, as compared to 1.25 MHz for cdmaOne systems. Figure 1.3 illustrates how these different standards are evolved into one standard IMT-2000.

The need for a capacity increase necessitates a greater spectrum allocation (1885 MHz–2025 MHz and 2110 MHz–2200 MHz) for 3G systems. The key features of the IMT-2000 system defining the ITU's view of 3G cellular network capabilities are as follows:

(a) High degree of worldwide commonality of design
(b) Compatibility of services with fixed networks and within IMT-2000
(c) More efficient use of the available spectrum
(d) Voice quality comparable to that of PSTN

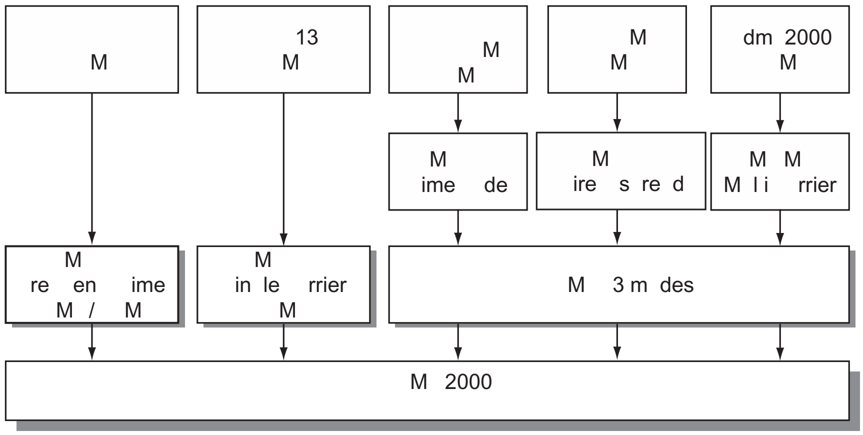


Fig. 1.3 *Evolution of IMT-2000 standards*

(e) 144–kbps data rate available to users in high-speed vehicles over large areas
(f) 384 kbps available to pedestrians standing or moving slowly over small areas
(g) Support for 2-Mbps data rate for office use
(h) Symmetrical and asymmetrical data-transmission rates
(i) Support for both circuit-switched and packet-switched data services
(j) Support for wide variety of mobile phones for worldwide use including pico, micro, macro, and global cellular/satellite cells
(k) Worldwide roaming capability
(l) Capability for multimedia applications and a wide range of services
(i) Flexibility to allow the introduction of new services and technologies

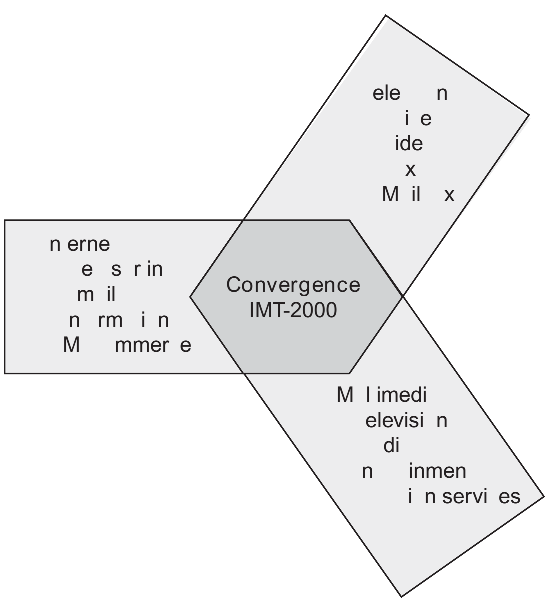


Fig. 1.4 *Convergence of services in IMT-2000*

The third generation aims to combine telephony, Internet, and multimedia into a single device. A convergence of all these applications in IMT-2000 is depicted in Fig. 1.4. This entails an additional requirement that it supports the Internet protocols and be based on a packet-switched network backbone.

To achieve the convergence of various services, IMT-2000 systems have been designed with six broad classes of service in mind. Cellular service providers will be able to offer whatever data rates the mobile users want up to a maximum 2 Mbps or so, and the mobile users will also have flexibility to choose the required data-rate service. Three of the service classes are already present on 2G networks to some extent, while three more service classes are new and involve mobile multimedia. In order of increasing data rate requirements, these services are the following:

(a) Voice 3G systems will offer speech quality at least as good as the fixed telephone network. Voicemail will also be eventually integrated fully with email service through computerised voice recognition and synthesis techniques.

(b) Switched data This includes dial-up access to corporate networks or fax service or the Internet access that doesn't support a fully packet-switched network.

(c) Messaging This is an extension of paging, combined with Internet e-mail service. Unlike the text-only messaging services built into some 2G systems, 3G systems will allow e-mail attachments. It can also be used for payment and electronic ticketing.

(d) Multimedia Messaging Service (MMS) The MMS is designed to allow rich text, colour, icons and logos, sound clips, photographs, animated graphics, and video clips. It works over the broadband wireless channels in 3G networks.

(e) Immediate messaging MMS features push capability that enables the message to be delivered instantly if the called mobile user is active. It avoids the need for collection from the server. This always-on characteristic of the mobile users opens up the exciting possibility of multimedia chat in real time.

(f) Medium multimedia This is likely to be the most popular 3G service. Its downstream data rate is ideal for web surfing, games, location-based maps, and collaborative group working.

(g) High multimedia This can be used for very high-speed Internet access, as well as for high-definition video and CD-quality audio on demand. Another possible application is online shopping for intangible products that can be delivered over the air such as a software program for a mobile computer.

(h) Interactive high multimedia This can be used for high-quality videophones, videoconferencing or a combination of videoconferencing and collaborative working.

(i) Sending multimedia postcards A clip of a holiday video could be captured through the integral video cam of a user's mobile handset or uploaded via Bluetooth from a standard camcorder, then combined with voice or text messages and mailed instantly to any other mobile user.

1.4.5 Wireless Networking Technologies

The use of radio signals for data transmission during World War II by the US Army inspired a group of researchers in 1971 at the University of Hawaii to create the first packet-based radio communications network called ALOHAnet, the very first wireless local area network (WLAN). It consisted of 7 computers that communicated in a bi-directional star topology. The first generation of WLAN technology used an unlicensed ISM band of 902–928 MHz. To minimise the interference from small appliances and industrial machinery, a spread spectrum was used which operated at a 500-kbps data rate. In 1990, the IEEE 802 Executive Committee established the 802.11 Working Group to create the WLAN standard. The standard specified an operating frequency in the 2.4-GHz ISM band. In 1997, the group approved IEEE 802.11 as the world's first WLAN standard with data rates of 1 and 2 Mbps. Like cellphones, wireless-equipped laptops within range of a given access point have the ability to communicate with the network. A single access point can communicate with multiple wireless-equipped laptops. Many systems allow roaming between access points. Despite their limited range (up to 100 m) and lower data rates (as compared to 1 Gbps offered by wired Ethernets), WLANs have become the preferred Internet access method for e-mail and Web browsing applications, in many offices, homes, campus environments, and public places.

Facts to Know!

The challenge for future WLANs is to support many users simultaneously with extended range for bandwidth-intensive applications such as video.

A wireless personal area network (WPAN), such as Bluetooth IEEE 802.15.1, enables wireless communication between devices, ranging from computers and cell phones to keyboards and headphones, and operates in ISM 2.4 GHz band. WiMAX (WMAN based on the IEEE 802.16 family of standards) will soon offer wireless broadband Internet access to residences and businesses at relatively low cost.

1.5 COMPARISON OF WIRELESS SYSTEMS

Wireless communication systems primarily comprise of a fixed-base transceiver station and a number of fixed/mobile subscriber transceiver equipments. The base station as well as the mobile subscriber of various types of mobile or portable wireless communication systems can be compared for the types of services, functionality (Transmitter Tx only or Receiver Rx only or Transceiver), operating carrier frequency range, the level of infrastructure needed, configuration complexity, hardware cost, and radio coverage range. Table 1.3 gives a brief account of the comparison of three most commonly used household wireless communication systems, that is, paging systems, cordless phone systems, and cellular telephone systems. The details of all these systems are covered in Chapter 10.

Table 1.3 *Comparison of wireless communication systems*

Service type	*unctionality*	*Operating frequency*	*Level of infrastructure*	*Complexity*	*Hardware cost*	*Range*
Paging system	BS: Tx only MS: Rx only	< 1 GHz	High	BS: High MS: Low	BS: High MS: Low	High
Cordless phone system	Transceivers	1–3 GHz	Low	BS: Low MS: Medium	BS: Medium MS: Low	Low
Cellular phone system	Transceivers	< 2 GHz	High	High	BS: High MS: Medium	High

Virtually, all these wireless communication systems aim to connect a mobile subscriber (vehicle-installed or handheld or portable) to a fixed wireless base transceiver system having antennas mounted at reasonably high towers. The user expectations vary widely among the type of services needed. The infrastructure costs are dependent upon the required coverage area. The radio link between the cordless phone base station and the portable cordless handset is designed to behave identically to the coiled cord connecting a traditional wired telephone handset to the telephone base. For example, cordless telephones use fixed base stations so that they may be plugged into the existing standard telephone line.

Similarly, in case of low power, hand-held cellular phones, a large number of cell sites are required to ensure that any mobile phone is in close range to a cell site within its service area. If cell sites area is not within the radio coverage range, a high transmitter power would be required at the mobile phone which is limited by the battery life.

Facts to Know!

The cellular systems have been evolved from the first generation of analog cellular systems standards through second-generation digital cellular standards, followed by more advanced third-generation digital cellular standards providing multiple user services including voice, high-speed data and multimedia applications.

Table 1.4 summarises the most common cellular systems standards used in North America, Europe, and Japan. The details of AMPS, ETACS, USDC-IS 54/136, GSM, PDC, and IS-95 cellular systems are covered in Chapters 10–12.

The first-generation analog cellular systems use frequency modulation scheme for speech transmission.

Table 1.4 *Major cellular communication systems standards*

Analog or digital	*Cellular systems standard*	*Frequency band*	*Multiple access technique*	*Modulation scheme*	*Region*
Analog	AMPS	824–894 MHz	FDMA	FM	North America
Analog	NMT-900	890–960 MHz	FDMA	FM	Europe
Analog	ETACS	900 MHz	FDMA	FM	Europe
Analog	JTACS	860–925 MHz	FDMA	FM	Japan
Digital	USDC/IS-54 or 136	824–894 MHz	TDMA	π/4-DQPSK	North America
Digital	IS-95	824–894 MHz; 1.8–2.0 GHz	CDMA	QPSK/ OQPSK	North America
Digital	GSM	890–960 MHz	TDMA	GMSK	Europe
Digital	PDC	810–1501 MHz	TDMA	π/4-DQPSK	Japan

Individual calls use different frequencies and share the available spectrum through frequency division multiple access technique. The world's earlier cellular systems include North America's Advanced Mobile Phone System (AMPS) operating in the 800-MHz band (50 MHz allocated spectrum: 824–849 MHz uplink and 869–894 MHz downlink), with 832 full-duplex channels, having a one-way channel bandwidth of 30 kHz for a total spectrum occupancy of 60 kHz for each duplex channel. The AMPS system uses a seven-cell reuse pattern with provisions for three-sectoring per cell and cell splitting to increase capacity when needed. The analog AMPS system requires that the received signal strength be at least 18 dB above the co-channel interference to provide acceptable call quality.

In Europe, the earlier cellular systems include the Nordic Mobile Telephone system (NMT 900), developed for the 900–MHz band. The European Total Access Communications System (ETACS) operates with a 25-MHz band in the uplink (890–915 MHz), and a 25-MHz band in the downlink (935–960 MHz), with a total capacity of 1000 full-duplex channels, having a one-way channel bandwidth of 25 kHz for a total spectrum occupancy of 50 kHz for each duplex channel. The smaller bandwidth channels result in a slight degradation of signal-to-noise ratio and coverage range.

The United States Digital Cellular (USDC) standards IS-54 and then IS-136 allowed cellular operators to replace gracefully some single-user analog channels with digital channels which support three subscribers in the same 30-kHz bandwidth. In this way, US cellular service providers gradually phased out AMPS analog mobile phones as more subscribers accepted USDC digital mobile phones. The USDC system uses digital modulation (π/4 differential quadrature phase-shift keying), speech coding, and time division multiple access (TDMA) in place of analog FM and FDMA as in the case of AMPS. The capacity improvement offered by USDC is three times that of AMPS.

Facts to Know!

North America's IS-136 digital cellular systems are eventually being replaced by IS-95 based CDMA technology. It supports a variable number of subscribers in 1.25-MHz wide channels using direct-sequence spread-spectrum modulation scheme.

CDMA systems can operate at much larger interference levels because of their inherent interference resistance properties. The ability of CDMA to operate with a much smaller signal-to-noise ratio than conventional narrowband FM techniques allows CDMA systems to use the same set of frequencies in every cell. It provides a large improvement in the overall system capacity. Unlike other digital cellular systems, the IS-95 system uses a variable rate vocoder with voice-activity detection that considerably reduces the required data rate and

Facts to Know!

The European GSM digital cellular standard is commonly referred to as digital TDMA cellular system. In Japan, the Pacific Digital Cellular (PDC) standard provides digital cellular coverage using a system similar to North America's USDC.

also the battery drain by the mobile phone. The North American IS-136 is commonly referred to as *digital TDMA cellular systems* and the IS-95 standard is called *digital CDMA cellular systems.*

The Pan European digital cellular standard GSM (Global System for Mobile) is a 900-MHz band. The GSM standard has gained worldwide acceptance as the first universal digital cellular system with modern network features extended to each mobile subscriber, and is the leading digital air interface for PCS services above 1800 MHz throughout the world.

The channel bit rate of GSM is 270 kbps whereas that of IS-54 and JDC is 48 kbps and 42 kbps, respectively. Higher channel bit rates of a digital cellular system allow simple implementation of higher data rates for data services. By assigning several voice slots to one user in a single carrier, one can easily increase the maximum supportable data rate for a data service offered by the cellular network. Similarly, the 1228.8 kcps channel chip rate of IS-95 provides a good ground for integration of higher data rates into IS-95. This fact has been exploited in IMT-2000 systems to support data rates up to 2 Mbps.

1.6 EVOLUTION OF NEXT-GENERATION NETWORKS

After the successful creation of an environment for second-generation digital mobile cellular communications systems, demand for further evolution in mobile and wireless systems is now emerging. Users place emphasis on the integration of mobile communication applications, particularly accommodating multimedia applications, higher performance, global and seamless coverage of wired and wireless services as well as greater customisation of services.

At this stage, many uncertainties remain as for the concrete specification of the next-generation wireless network, but the process of discussion among market drivers and within the standardisation bodies such as 3GPP2, UMTS Forum, and ETSI among others has been launched. There are several areas which have been identified for a global approach towards regulation, standardisation, frequency allocation, R&D efforts and international cooperation. The objective is to create a framework which leads to greater choice, improved quality and lower prices for all users of mobile services, while ensuring full competition within an environment which fosters the competitiveness that the sector achieved so far.

The recent evolution of the regulatory environment in view of the full liberalisation of telecommunications has provided a comprehensive framework for both fixed and mobile telecommunications systems. It will be important to assess the flexibility of that framework against the new demands with the emergence of the concept of next-generation network (NGN). The same applies for the decision on priorities for standardisation and choices concerning the most suitable allocation of frequency bands in the spectrum so far reserved for 3G technology.

Recent years have witnessed the rapid evolution of commercially available mobile computing devices and networks. There are several viable but non-interoperable wireless networking technologies—each targeting a mobility environment and providing a distinct quality of service.

Figure 1.5 depicts the vision of next-generation networks.

This would also mean enhancement of present 2G/2.5G/3G cellular systems already in operation across the world. The usage environment would demand high-speed wireless access techniques in hot-spots, that is, inadequate signal reception areas or indoor multi-structured buildings, and would also require ultra short-range connectivity in order to access laptop or desktop terminals with a remote wireless digital device using technology such as Wi-Fi or Bluetooth.

With the advancement in mobile information technologies like ultra high-speed transmission, wireless Internet protocol IPv6, user-controllable software defined radios, the potential mobile users would be able to

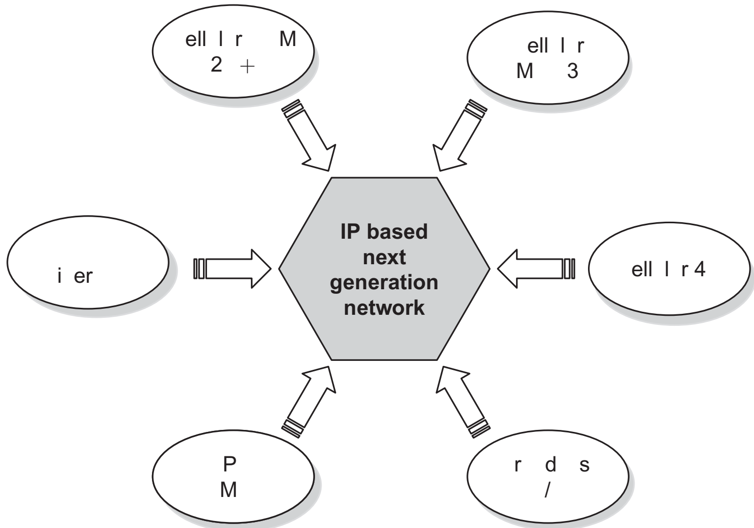


Fig. 1.5 Vision of next-generation networks

- access the Internet as they do in the office—anywhere, anytime while on move
- use cellphones or laptop computers or any other PDA as mobile communication terminal
- choose freely the services, applications and service-providing networks
- exploit advanced mobile E-commerce applications with higher levels of data security and integrity during business transactions

1.6.1 Functional Requirements

Very High-speed and High-quality Transmission Next-generation mobile communication systems should be able to handle a large volume of multimedia information like downloading a full song or sending a complete data file or several video clips. This would be possible by various means like transmitting data at 50 Mbps–100 Mbps, having asymmetric data speeds in up and down links, having continuous coverage over a large geographical area, applying QoS mechanisms (for example, efficient encoding, error detection and correction techniques, echo cancellers, voice equalisers) at low, affordable and reasonable operating costs.

> **Facts to Know!**
>
> The lack of a uniform set of standards, the inconsistency in the quality of service, and the diversity in the networking approaches make it difficult for a mobile computing environment to provide seamless mobility across different wireless networks.

Open Platform Next-generation mobile communication systems should be open regarding mobile phone platform, service nodes, and mobile network mechanisms. That would mean that the user can freely select protocols, applications and networks. Advanced service and content providers can extend their services and contents independent of operators. Location and charging information can be shared among networks and applications.

Flexible and Varied Service Functions Next-generation mobile communication networks should be seamless with regard to the medium, whether it is wireless or optical fiber or satellite or wireline, with regard to corresponding hosts or service providers as well as have interconnectivity with other networks like GSM or CDMA or other telecom networks.

1.6.2 Seamless Mobility

There is a powerful trend towards seamless mobility in the cellular world, where mobile professionals today and eventually all users in the future would like to communicate and be able to do their routine business anytime, anywhere. As a result, there is real demand for ubiquitous connectivity between a wide variety of mobile devices and access technologies, which include WLANs and WWANs. Roaming and communications among these technologies are therefore must-haves for seamless mobility to occur.

> **Facts to Know!**
>
> The future network infrastructures will consist of a set of various networks using IP as a common protocol so that users are in control to choose every application and environment.

The next-generation wireless networks intend to provide acces to information anywhere, anytime, with a seamless connection to a wide range of information and services, and receive a large volume of information, data, images, video, and so on.

To realise the potential of seamless mobility and ensure continued profitability, wireless mobile service providers have to focus as equally on WLAN implementations as they do on their cellular WWANs. Wi-Fi and other wireless services are add-ons that can exist and succeed together and provide users what they want, and when they want it. Users will utilise these technologies for different reasons and at different times. The 2.5G and 3G technologies such as GPRS, EDGE, CDMA 1xRTT, and CDMA 1xEV-DO will be used for applications requiring instant access of bursty data like e-mail, text messaging, and multimedia message service, among others. But WLANs will be used in specific locations where users need access to their corporate files and Intranets.

One of the most important requirements of the 3G system is that it provides a seamless path of migration from present-day digital wireless networks that it is capable of inter-working. All major providers of wireless network systems, services and terminals agree that next-generation networks should evolve from the core infrastructures contained in today's digital networks. A comparison of key aspects of all generations of wireless and cellular network is depicted in Fig. 1.6.

Seamless mobility is the necessity as a result of extensive primary and secondary research on a variety of industry participants including cellular service providers, Internet service providers, electronic component manufacturers, equipment suppliers, and software providers. Technology drivers and obstacles that must be addressed to achieve growth in the WLAN market such as roaming, security, seamless authentication, handovers, and billing are equally important.

The handoff latency of many new access technologies such as wireless LAN devices is very large, of the order of hundreds of milliseconds. During this period, mobile nodes cannot receive or transmit packets.

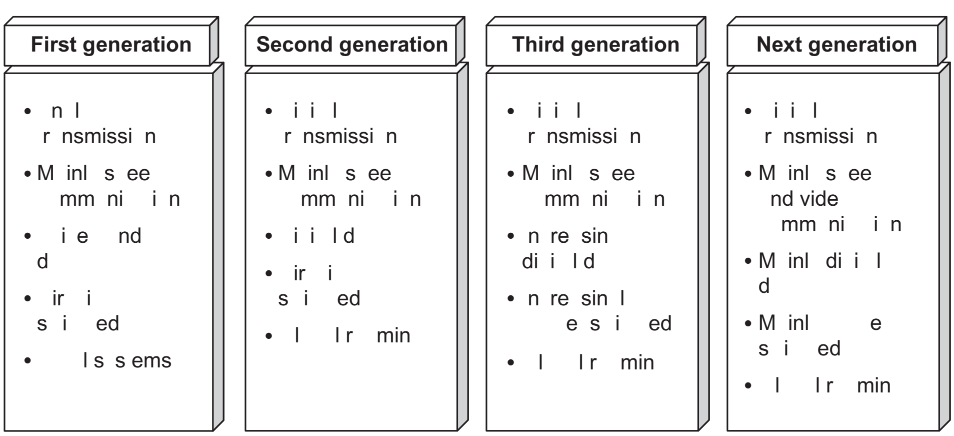


Fig. 1.6 *Comparison of cellular network generations*

This results in significant performance degradation during hand-off operation. Furthermore, while handing off across subnets, network layer hand-off can be initiated only after link layer hand-off is complete. This increases the latency even further. Present cellular systems designed to handle mobility resolve the latency issue by adding necessary interfaces and intelligence to the network. However, in the IP-based architectures, the access technologies do not support the level of hand-off coordination that cellular systems provide. Though packets can be buffered in a local mobility agent and then retransmitted to eliminate packet losses during hand-offs with some latency.

Based on the developing trends of mobile communication, next-generation wireless networks will have broader bandwidth, higher data rate, and smoother and quicker hand-off and will focus on ensuring seamless service across a multitude of wireless systems and networks. Seamless handoffs can be designed for homogeneous networks (IEEE 802.11 WLAN and cellular) at the MAC layer and with limited participation of the mobile node in the decision. For IP to mobile node applications where there is a multiplicity of potential wireless access technologies, the requirement of functionality in the mobile node and wireless access router to facilitate seamless transfer (network or mobile node initiated), and the need of Quality of Service (QoS), Authorisation/ Authentication/ Accounting (AAA), as well as the need of security infrastructure changes are to be established.

1.6.3 Wireless Data Communications Technologies

As new wireless communications technologies are introduced, they will become even more integral to our day-to-day requirements and will continue to change how we live. Wireless data communications technologies include cellular, wireless LAN, wireless MAN, Bluetooth, RFID, and satellite. Table 1.5 summarises the key features of various wireless data communications technologies.

Table 1.5 *Summary of wireless data communications technologies*

S. No.	*Wireless technology*	*Data speed*	*Radio coverage*
1.	2G digital cellular	10 kbps	Nationwide through roaming
2.	2.5G digital cellular	Up to 384 kbps	Nationwide through roaming
3.	3G digital cellular	Up to 2 Mbps	Nationwide through roaming
4.	WLAN 802.11b	11 Mbps	Building/campus, 100–120 metres
5.	WLAN 802.11g	54 Mbps	Building/campus, 90 metres
6.	WMAN 802.16 WiMax	75 Mbps	Metro city area, 56 kilometres
7.	Bluetooth	1 Mbps	Room/house, within 10 metres
8.	Ultra Wide Band (UWB)	100 Mbps	Auditorium, 50 metres
9.	RFID	Few kbps	Small area within a store room, 2.5 cm to 100 m
10.	GPS and satellites	250 ms delay	Worldwide global

Figure 1.7 illustrates a pictorial view of comparing the capabilities of these wireless data communication technologies in terms of data speed and range.

Users are constantly demanding more functionality from their computers and laptops, and, as a result, wireless devices such as cellular phones, personal digital assistants (PDAs), and Smartphones are being combined into a single digital wireless device. Wireless networks play an important role in digital convergence as users demand to be connected to their voice and data networks at all times at any place. Digital convergence refers to the power of digital devices such as computers and mobile phones to combine voice, video, and text-processing capabilities, as well as to be connected to home and business networks and to the Internet. In the

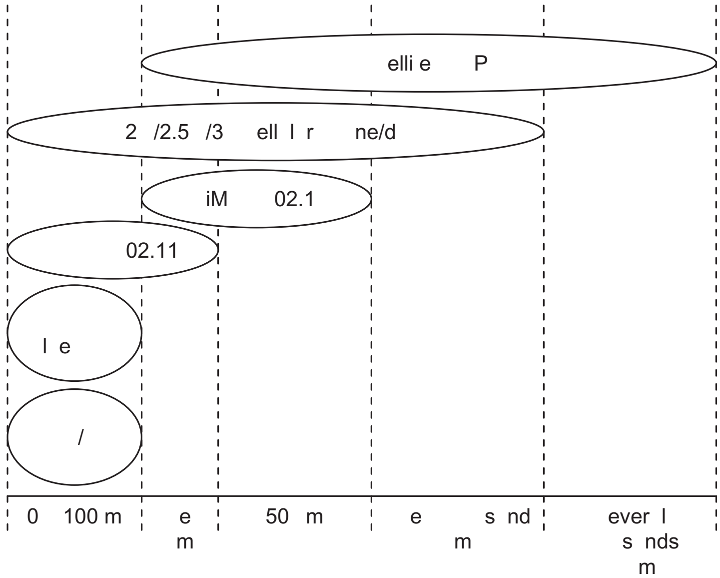


Fig. 1.7 *Comparative capabilities of wireless data communication technologies*

same way, the development of Voice over IP networks use the same protocols and both wired and wireless media to carry voice conversations as well as data.

The future wireless technologies will see more and more mobile devices, the merging of classical voice and data-transmission technologies, and the extension of today's Internet applications onto mobile and wireless devices. New applications and new wireless mobile networks will bring ubiquitous multimedia computing to the common user; PDAs, wireless laptops and mobile phones will converge and many different functions will be available on one device, operating on top of Internet technologies. Figure 1.8 depicts the pictorial view of a typical fixed wireless network.

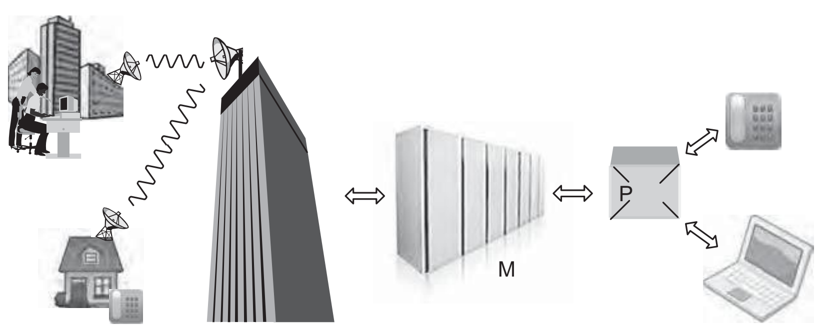


Fig. 1.8 *A typical fixed wireless network*

The next-generation wireless network promises to fulfill the goal of PCC (Personal Computing and Communication), a vision that affordably provides high data rates seamlessly over a wireless network. Seamless mobility in wireless networks intends to integrate, from satellite broadband to high-altitude platforms to 3G cellular and 3G systems to WLL and Fixed Wireless Access to WLAN and WPAN, all with IP as the integrating mechanism. With this, a range of new services and models will be available. In addition, next-generation wireless networks will be fully IP-based wireless Internet. Future mobile communication systems will certainly and surely achieve the concept of a Global Village'. Both, the recent and short-term future advances in wireless standards and technologies discussed in this book will enable an ever-wider range of applications for wireless devices.

1.7 APPLICATIONS OF WIRELESS COMMUNICATIONS

Today, more and more people use mobile phones than traditional fixed telephones. These trends create ever-increasing requirements for well-educated qualified telecom engineers and technocrats who understand the developments and possibilities of wireless and mobile communications. Cellular communication systems mainly rely on judicious frequency reuse planning and multiple access techniques to maximise system capacity. Cellular systems have evolved from analog techniques to the more flexible advanced digital techniques that are currently employed. Future developments are aimed at further enhancing these digital techniques to integrate voice, messaging, and high-speed data.

Wireless networks can be made available where regular wired networks cannot. Wireless applications, the use of wireless communications technologies in conducting day-to-day business activities, can be found in every industry. Different standards for these systems have been developed for operation in different regions of the world such as North America, Europe, and Japan at the same time.

> **Facts to Know!**
>
> The most widely used wireless communication systems can be broadly categorised into various distinct classes depending on their popular application such as Paging and Messaging Systems, Cellular Telephone Systems, Cordless Telephone Systems, Wireless Local Loop, Wireless LAN, Wireless PAN and many others.

The eventual goal of personal communication systems is to allow each individual user to have one personal mobile phone and phone number which will take the place of home, office, vehicle, and portable handheld phones. The third-generation personal cellular systems feature higher maximum data rates, greater capacity for voice calls, and the ability to work with a wide range of cell sizes and types. It is more standardised than the second generation. Technologically, the increased capacity is achieved by using extra spectrum and new modulation techniques such as 8-PSK and spread-spectrum that squeeze higher data rates from a given spectrum. The 3G standards are backward compatible so that mobile phones can maintain a connection while moving between cells based on the earlier and the new technologies. Global roaming will be possible with special multimode mobile phones. A fourth-generation (4G) or next-generation network, with data rates of 150 Mbps and more, is already under development.

The main areas of wireless applications can be broadly categorised as follows.

Office and Household Environments Typically, an office space is wired with computer cables for network connections and telephone wires for telephones. With wireless technologies such as WLAN and Bluetooth, that expensive cabling infrastructure is no longer necessary. This means that an office can be created in a very short period of time with minimum down time and infrastructure at almost no additional cost. During office renovations or reorganisation, employees can move to another location in the building and can continue working as usual with immediate access to the wireless network. In addition to the accessibility of networked data, wireless technologies allow businesses to create an office where the traditional infrastructure doesn't already exist.

Industrial Control In the construction industry, instant information from the job site including shortage of man power or materials, could be relayed back to the main office for rescheduling of workers to other sites to prevent idle time. Construction equipment such as bulldozers and earth graders are fitted with wireless terminals and GPS which can provide accurate location information on a colour-coded map to guide the operator as it digs. Implementing wireless technology is a key feature for many warehouse operations. By equipping all of the warehouse's machinery and personnel with wireless networking devices, managers can use warehouse management system (WMS) software to manage all of the activities from receiving through shipping. And since this network is connected into the front office computer system, managers can have current stock statistics.

Many large suppliers implement RFID in all the products and some highly sophisticated warehouses are operating with fully automated pallet machines and forklifts that can process the storing and retrieving of products completely without human intervention. Because of their huge size and complexity, large manufacturing facilities, such as automotive assembly plants, are using wireless remote sensors called motes connected to a WLAN, which collect data and transmit it to a central location. Technicians in a control room can monitor the status of every machine or device and take corrective measures.

Facts to Know!

As an example, a hotel conference room that may not have the infrastructure to support a wired network can quickly be turned into a wireless networked office environment. Wireless-enabled devices have the ability to control lights, air conditioners, refrigerators, and other household appliances.

Facts to Know!

Smart equipment can be connected through wireless transmissions back to the remote office, which tracks engine hours and equipment location. Wireless terminals in the engine's diagnostic system can send an alert when the oil needs to be changed or other maintenance operations are due.

Education Sector Nowadays, most of the university and college campuses are equipped with Wi-Fi technology. The instructors and students carry their notebooks and laptops with in-built wireless devices. They can access the campus network wirelessly from almost any location on the campus. This wireless education model makes computing resources available to students from anywhere (including hostel, cafeteria) and at any time. An instructor can create a classroom presentation on the notebook computer at his place and then carry it right into the classroom or lab. Without the need to plug or unplug any cables to connect to the campus computer network, the notebook automatically gets connected to the classroom network. Teachers can also distribute handouts directly to students who have brought their own wireless devices to class. Wireless technology translates into cost savings for university and colleges as well. There is no need to have traditional classrooms or computer labs with expensive wiring and infrastructure.

Health Services Even telephones are connected to hospital WLANs, employing VoIP technology. Doctors and nurses no longer have to be paged over the public address system. Doctors can also consult with specialists (located somewhere else) while they are at a patient's bedside. Cellular phones cannot be used in health-care facilities, but handsets that can connect to WLAN and use VoIP are allowed, and these improve efficiency of health care services in hospitals. Notebook computers on mobile carts or handheld PDAs with bar code scanners or RFID tags and a wireless connection enable doctors and nurses on duty to document a patient's medication administration immediately in the computer as they move from one place to another place. Nurses first identify themselves to the computer system by scanning their own personal bar-coded ID badge. The patient's bar-coded armband is then scanned and all medications that are currently due for that particular patient are brought up on the screen. The medications to be administered are sealed in bar-coded pouches. Nurses scan RFID tags before opening the package. An alert immediately appears on the screen if the wrong medication or incorrect dose is identified. After administration, the nurse indicates through the wireless network that the medication has been given, essentially electronically signing the distribution form.

All hospital personnel have real-time access to the latest medication and patient status information from their place of work.

> **Facts to Know!**
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> Wireless point-of-care computer systems installed in medical care centres and hospitals allow medical staff to access and update patient records immediately. Select medical groups provide their physicians with a PDA, printer, and prescription-writing software. This technology is intended to reduce errors associated with illegible handwritten prescriptions.

Government and Military Operations Government offices deploy a broadband wireless network to enable their employees and contractors at remote construction sites to access data stored in a central database. Police officers can both download and upload streaming video to help them tackle road accidents and crime. Wireless technology is being used to provide free Internet access to visitors and business people in public places. Using cellular and satellite communications, military personnel can talk, access the Internet, and receive full-motion video through their wireless handset. They can also connect with other wireless devices using the Bluetooth wireless protocol, or to a WLAN for numerous defense applications in the field.

Event and Travel Management Several large public auditoriums, arenas and sports stadiums are now equipped with wireless systems to facilitate the process of distribution of valid tickets and overall control of events. Entry tickets are printed with a unique RFID tag that is scanned at the venue's point of entry using a handheld or integrated wireless device connected to a wireless network. The network instantly validates the ticket and thus prevents the use of counterfeit or stolen tickets. In addition, wireless technologies are changing the latest information and entertainment experience itself. For example, wireless transmissions of in-progress game statistics are available to any one in the stadium with a wireless device such as a notebook computer or PDA. Because wireless technology creates mobility, the travel industry makes use of its advantages to plan and manage the itinerary. Wireless global positioning systems (GPS) that tie into emergency roadside assistance services have become standard features. Airport terminals transmit wireless signals that passengers can pick up on their wireless notebook computers or PDAs while waiting for their flights. For a nominal fee, they can also surf the Internet or read their e-mail. Even the airplanes themselves are being equipped with wireless data access, offering wireless Internet capabilities to passengers on flights.

Home Entertainment FM radio, TV, and satellite radio serves the common man for news and entertainment services uninterrupted day and night using wireless communications. Several large computer manufacturers have introduced specialised media PCs that enable audio and movie enthusiasts to download, distribute, and control all forms of digital entertainment from anywhere. These PCs are equipped with wireless networking hardware and software to simplify the processing of sound, video, and pictures. One can send music, movies, or pictures to a stereo receiver, portable device, or PC located anywhere in the building. The files can be downloaded to digital media portable devices, such as MP3 and video players that can be used while roaming anywhere.

Environmental and Industrial Research Scientists are now using small, battery- or solar-cell-powered WLAN transmitter-equipped wireless smart sensors in difficult places such as deep caves or on mountain tops or at the tops of tall trees to monitor the effects on dense forests caused by ultraviolet rays due to the holes in the ozone layer. The data recorders can be installed in electric power- or generator-operated

> **Facts to Know!**
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> Third-generation (3G) cellular mobile communications systems are critical to the wireless Internet services, offering permanent access to the Web, interactive video, and voice quality that sounds more like a CD player than a normal cell phone. 3G systems provide ISDN speeds for everyone, equipped with the normally used mobile phones anywhere.

Facts to Know!

Wireless wide area networks will enable companies of all sizes to interconnect their offices without the high cost charged by telephone carriers for their landline connections. WLAN applications are found in a wide variety of industries and organisations

nearby locations, and can communicate with the sensors using wireless technology. This capability has proven to be a major breakthrough in many different scientific fields and research. Wireless sensors are capable of communicating using wireless technologies, and are widely used in large manufacturing facilities to monitor equipment and for scientific research.

1.8 POTENTIAL MARKET AREAS

The market for mobile telephones for person-to-person communication has grown rapidly across the globe. One of the major growth areas for wireless mobile communications is referred to as Machine-to-Machine, or Man-to-Machine (M2M) communications over cellular radio networks. Wireless communications devices are integrated within other equipments and are used in many different telemetry applications for making Short Message Service (SMS) calls or automatically sending data and voice messages to a control centre. For example, they are being integrated in automobiles, fleet management systems, fire and security alarms, vending machines, public utility meters and other industrial equipment. Software upgradations have also been made to allow the seamless transport of data and information. The wireless devices are being built into host appliances, and controlled through a computerised link.

Wireless communications devices will increasingly be built into a wide variety of equipment, including domestic appliances, industrial machinery, metering equipment and all kinds of vehicles, and will soon be integrated in microchips. Water, electricity and gas meters are already being read digitally and reported automatically by wireless communication. There are houses, or business premises, where a break-in or fire is reported to the authorities without human intervention. Vehicles are being developed which not only call for help automatically after an accident, but which can also identify with pinpoint accuracy where they are, and even give some information about what type of incident has occurred.

Figure 1.9 depicts the potential usage of a variety of common applications and the related data-rate requirement from the wireless technology.

Wireless communications devices will thus automate a wide range of existing functions and will stimulate the development of many others, which may be beyond imagination as on today. This market is potentially significantly larger than the person-to-person wireless market. The challenge of marketing M2M technology is its more-or-less unlimited applications, the only boundary being human imagination. The potential for wireless device communication could be summed up in the concept of at least one device per house, and one device per vehicle.

Similarly, a single utility or security company or the builder may provide the first device to the household or other premises, and automotive manufacturers may supply a single device to a vehicle for a dedicated purpose. Once this device is in place, it can then be used for a variety of different purposes, and customers will come to expect more and more communication systems to be channeled through a wireless device. Many domestic appliances, such as refrigerators, will be routinely equipped with them. These devices will report, probably via a LAN network, to a central gateway device, which will provide the link between the dedicated and the public wireless network.

Facts to Know!

In principle, all communication systems could be channeled through the wireless device, both for home and for vehicle

As regards the size of this potential market; a tenfold growth in vehicles is probable worldwide over the next decade, to a total of 500 million vehicles, with an annual increase of 50 million.

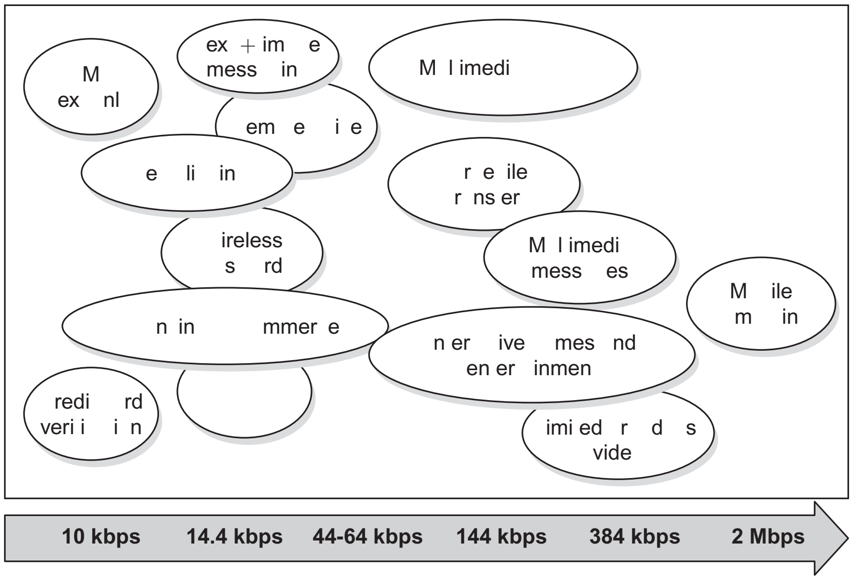

Fig. 1.9 Potential market areas versus data-rate requirement

In addition, there are millions of households, vending machines, alarms and elevators; each of these has a potential need for wireless communication. On top of this, there are all types of metering systems, plus many other types of equipment in which the device has applications. In conjunction with the development of this technology, there will be emergence of new multi-service providers; organisations or businesses which will act as intermediaries to households in supplying electronic services by wireless communication, including many forms of entertainment, Internet access, telephone services, security systems and utility services. There can be mergers between media–entertainment and telecommunications companies and the emergence of multi-utilities' companies which provide gas, water and electricity services.

Facts to Know!

The scope for wireless devices is even greater in the business-to-business world, than for these domestic applications, and applications such as fleet management, remote diagnostics, vending, monitoring of servers, etc., and will grow faster and be more significant.

Key Marketing Issues Two developments are key to unleashing the potential of the wireless–device market. First, decreases in air-time costs on public networks and, second, increases in their bandwidth, to enable a much broader range of communication services to be delivered via the public wireless channel. There are also a number of secondary issues concerning standards, reliability, privacy and legal requirements.

Reliability/Signal Quality Several of the potential wireless-device applications will require greater bandwidth than is currently available via public networks. These bottlenecks will, however, soon be overcome by new generations of devices and wider bandwidth standards which are currently being developed. The increasing use of wireless communication for all types of emergency alarm systems will obviously put greater demands on its reliability as a communications channel. Standards and roaming are also important issues in the automotive and fleet-management markets.

Privacy/Security The increasing use of wireless communications raises the question of privacy, both for individuals and for companies. For instance, making the location of people and vehicles traceable by unauthorized third parties can clearly be regarded as undesirable. On the other hand, the growing use of digital signals in itself significantly reduces the possibility of breach in privacy. If the privacy issue is considered absolutely critical, then encryption is always a possibility. However, information will always be accessible to the authorised personnel who operate these systems. Ultimately, they have to be trusted not to disclose this information in illegitimate ways.

1.8.1 Target Business Areas

(a) The Automotive Industry Market The potential for wireless communication in vehicles includes entertainment systems, climate control, mechanical status reports to dealers or vehicle maintenance centres, satellite navigation via GPS (Global Positioning Systems); traffic information including map guidance and advice on traffic congestion, etc. There are other possibilities such as immediate accident reporting using the triggering of airbags to signal an alert to the emergency services, to provide an exact location using GPS; seat sensors could provide the number of passengers, and the details of which airbags have been triggered could provide some information about the type of accident, for example, whether it was a head-on collision. At present, similar products are being developed for the high end of the automobile market, and soon they could become standard specification.

(b) The Fleet Management/Vehicle Positioning Market For this application, each vehicle in a fleet will be equipped with a wireless device capable of checking the vehicle's position, monitoring and reporting for security including alarms, and as a backup for the driver's mobile phone. The device can also be used for vehicle diagnostics.

> **Facts to Know!**
>
> The major market driver in this industry is the increasing introduction of deregulation, privatisation, and opening up to competition.

(c) The Utilities Market In this market, wireless devices will be used for remote metering of the consumption of the various utilities—gas, electricity and water. Other applications in the utilities business area include infrastructure monitoring, involving observation and reporting from water-pumping stations, electricity-distribution sub-stations, and remote switchgear, as well as service and maintenance functions. The issue of billing and charging becomes very important, with the opportunity to change rapidly between suppliers.

(d) The Security Systems Market This market can be categorised in two sectors. The *auto alarm sector* involves the retrofitting for alarms, fitted into vehicles after they have been manufactured and sold. The *household and building sector* covers building alarms for households, commercial premises of various types, and other buildings. Security companies are interested in wireless communication because they provide a more secure alarm-reporting system, given that physical landlines can be cut or damaged and, secondly, it involves much less installation cost.

> **Facts to Know!**
>
> As an example, if a household has a wireless communication device in it, many other opportunities are opened up; the device can be used for many other applications including monitoring the well-being of elderly people. This is a step towards the concept of an 'intelligent house'.

(e) Vending Machines The potential for this market may include the vending-machine manufacturers, the makers of the coin systems for the machines, the vending operators and the owners of the point of sale. Installing a wireless device in a vending machine offers a large number of benefits, such as it enables enormous price flexibility. For example, offers on certain items can be made automatically and remotely depending on the local competition for a particular machine. Other payment options can be introduced, for example, 'micro-payments'—people could order and pay for goods from a vending machine via their mobile phone. Such payments could be added to their mobile phone bill.

In addition, the wireless device can provide an enormous database, giving rich and accurate information on the usage of each machine. This data can be used to optimise the positions of machines, the distances between machines and the stock each one contains. The manufacturer would need to form a partnership with an application-service provider who would manage this specific communication service for vending machines. The manufacturer would then sell or lease the wireless device and other equipment, to the machine operators. There is a need to recruit and coordinate the activities of several other players including a telecom operator, access provider, a content and application provider to support the service, plus an Internet payment provider to manage the financial transactions, leading to a combined Wireless Application Service Provider (WASP) to be the entrepreneur.

Facts to Know!

Just as the number of different wireless devices are on a tremendous increase, so is the number of job opportunities for professionals, wireless engineers, wireless network managers, and wireless technical support personnel to build and support the ever-growing wireless communication technology.

1.9 CHALLENGES FOR RESEARCH

Wireless communications has been one of the major technological areas for research as well as industrial applications. The world of wireless communications has come of age where true wireless multimedia services can be offered to highly mobile subscribers seamlessly on a global arena. Future wireless communication networks are expected to support diverse IP multimedia applications to allow sharing of common resources among diverse subscribers. Mobility is the most important feature of a cellular wireless communication system. The potential users of wireless telecommunication systems require establishing voice as well as data communication with other users while on the move. This results into the deployment of mobile communication systems to cover large geographic regions. There has been a tremendous growth rate in wireless mobile communication systems and networks in recent times, mainly due to increasing mobility awareness among people.

While traditional communication paradigms deal with fixed wireline or wireless networks, mobility raises a new set of wireless communication techniques, complexity, challenges and solutions. Due to the lack of an appropriate fixed communication infrastructure in many countries, mobile communication is the only viable alternative.

Cellular communication systems mainly rely on judicious frequency reuse planning and multiple access techniques to maximise system capacity. Cellular systems have evolved from analog techniques to the more flexible and advanced digital techniques that are currently employed. Future developments are aimed at further enhancing these digital techniques to integrate voice, messaging, and high-speed data. The rapid advancements in the field of VLSI hardware technology as well as embedded software have paved the way for reliable mobile communication systems.

Building on these current developments, the communications industry is developing a strategic vision of the next generation of digital mobile systems. Details of future service concepts and of future user requirements need to be addressed in order to formulate regulatory, frequency and standardisation responses at the global level.

The information and communication technologies industries are a critical component of the world economy in that they are a major and growing part of industrial activity. They are also one of the keys to future competitiveness of industrial processes, products and services and a platform for the emerging Information Society. There are two major challenges that need to be addressed—meeting the changing user needs for mobile services; and further developing the conditions for industrial competitiveness within the mobile communications sector. Achieving these goals will be important not only for the telecommunications sector, but also for the broader economy.

One of the major issues concerning fair market competition relates to the issue of roaming. At least in the initial stages of deployment, UMTS systems may have to seek roaming arrangements with GSM networks in order to provide a sufficient degree of national and international coverage. Being based on commercial agreements between operators, the refusal to enter into roaming agreements with UMTS operators would be a major deterrent to the use of UMTS systems. At a later stage, roaming will take on a new dimension as different mobile and/or fixed service segments converge. This will require a careful monitoring of roaming arrangements.

As the basic shape of UMTS or other third-generation systems emerges, ETSI is likely to continue to play a central role in close cooperation with the UMTS Forum and other interested parties in determining the precise requirements for standardisation. GSM is a comprehensive and open standard allowing for the evolution towards different service forms while maintaining the integrity of the GSM platform. To what extent such an approach can be applied to the UMTS concept is an issue. It needs to be ensured that the development of UMTS strikes the right balance between formal, open standardisation, and industry-developed de-facto standards of a more proprietary nature which may often develop more at a pace commensurate with the rapid life cycle of the information technologies and services which UMTS is likely to offer.

1.10 OUTLINE OF THE BOOK

The objective of this book is to provide a comprehensive technical overview of wireless communications, with an emphasis on cellular communication principles and systems—one of the fastest growing technological application areas in the telecom arena. The knowledge may lead towards convergence of wireless communication techniques, mobility, networking and information technologies. The various forms of wireless data communications using cellular communications, wireless LAN, Bluetooth, fixed broadband, and wireless WAN communications technologies are covered. The book is organised to provide fundamental treatment about many theoretical and practical aspects that form the basis of wireless and cellular communications, as well as emerging wireless networking technologies. Technical concepts which are at the core of design, planning, evaluation, and inventions of cellular communication systems are presented in an order that is conducive to understanding and grasping general concepts, as well as those specific to current and emerging cellular systems standards and wireless networks. To accommodate a wide range of previously related technical knowledge levels, important concepts throughout the text are developed from the basic principles and well supported with facts and examples.

Chapter 1: Evolution of Wireless Communication Systems An overview of historic evolution of wireless communications, wireless network generations, and a wide range of applications of wireless data communications is provided. The chapter follows the way the industry is classifying wireless data communications today and looks at the advantages and disadvantages of wireless data communications.

Chapter 2: Mobile Communication Engineering This chapter presents the fundamental theoretical concepts of wave propagation in wireless media. The important aspects of mobile communication engineering such as signal attenuation in wireless media, effects of multipath propagation on radio signals, small-signal fading phenomenon including simulation of wireless fading channels are introduced.

Chapter 3: The Propagation Models This chapter covers various propagation path loss models under different operating conditions such as free-space, mobile environment, and the signal attenuation due to foliage. The impact on received signal strengths at the mobile receiver is well supported with easy-to-follow examples.

Chapter 4: Principles of Cellular Communication This chapter deals with the principles of cellular radio communications such as frequency reuse and cochannel interference, which are at the core of providing wireless communication service to subscribers on the move using limited radio spectrum while maximising system capacity. A detailed analysis of hexagonal-geometry based cellular structure including the cluster formation and methods of locating cochannel cells is provided. The effect of cochannel interference on signal quality and system capacity is discussed with an introduction to various methods to reduce cochannel interference.

Chapter 5: Cellular Antenna System Design Considerations This chapter introduces the basic characteristics of cellular antennas applicable to cell-site antennas as well as mobile antennas. The antenna system design considerations in the fully grown hexagonal-pattern cellular architecture are covered which are based on omnidirectional antennas and directional antennas at the cell-site. The concept of cell sectoring in order to achieve desired signal quality in an interference-limited system is presented along with analytical evaluation of the performance of the system in terms of signal quality.

Chapter 6: Frequency Management and Channel Assignment This chapter provides a concise coverage of frequency management and channel assignment methodology in different cells of a cellular radio network. The example of channel numbering and channel grouping used in AMPS and GSM cellular systems is included. Various fixed, dynamic and hybrid channel assignment schemes are elaborated in order to clearly understand the implementation complexities of the frequency reuse concept in practical cellular systems and resultant improvement in capacity performance.

Chapter 7: Cellular System Design Trade-offs There is an inter-dependence of performance criteria such as cell coverage, signal quality, and traffic capacity for successful operation of the cellular systems. This chapter describes an extensive analysis of design trade-offs of cellular systems. Topics in this chapter are system parameters to increase cell coverage, coverage hole fillers and leaky feeders, system parameters to reduce interference, methods to increase traffic capacity including cell splitting, review of modulation, equalisation, diversity, speech and channel-coding techniques, as well as hand-off mechanisms.

Chapter 8: Multiple Access Techniques Several multiple-channel access techniques as well as packet radio and carrier-sense-based multiple-access, protocols are explained in this chapter. This justifies why the standard medium-access schemes from fixed networks are not suitable if used in a wireless environment. Multiple access techniques based on division of frequency, time, code, or space are described along with comparative pros and cons of their usage. It also explains how each access method can accommodate a large number of mobile users and demonstrates their impact on capacity and the network infrastructure of a cellular system. Packet radio and carrier sense based multiple access protocols are included along with their respective throughput performance.

Chapter 9: A Basic Cellular System An architecture and operation of a basic cellular system is introduced in this chapter. A brief account of various system-performance criteria such as voice quality, trunking and grade of service, spectral efficiency, radio capacity, service quality and special features are described with an objective of appreciating the design and implementation techniques of wireless and cellular networks.

Chapter 10: Wireless Communications Systems Various wireless mobile communication systems such as paging system, cellular telephone, cordless telephone and wireless local loop are discussed. The technical details of 1G analog cellular systems such as AMPS and ETACS, followed by 2G digital cellular systems and standards IS-54/IS-136 and PDC are also covered in this chapter.

Chapter 11: Global System for Mobile (GSM) This chapter presents the vital design and planning aspects of the most popular 2G cellular system (GSM). The various topics include air-interface standards, logical channels, frame formats, and signal processing. An overview of the speech-coding scheme, security aspects, and special services and features as applicable to GSM systems is provided here.

Chapter 12: CDMA Digital Cellular Standards The details of spread-spectrum techniques leading to another 2G digital cellular standard (CDMA) is covered in this chapter. The various topics include architecture, forward and reverse logical channels, and basic call processing. Unique features to CDMA technology such as power control, soft hand-off, cell breathing, and rake receiver concept are also presented here. The performance evaluation of CDMA systems for specified parameters is described too.

Chapter 13: 3G Digital Cellular Technology This chapter provides an overview of the major modern cellular communication systems such as 2.5G GPRS, EDGE, and cdmaOne technologies, followed by 3G cellular technologies such as UMTS, W-CDMA, and Cdma2000.

Chapter 14: Emerging Wireless Network Technologies This chapter presents an insight into the emerging wireless network technologies such as IEEE 802.11 WLAN technology including the most popular Wi-Fi, IEEE 802.15 WPAN Technology including Bluetooth, IEEE 802.16 WMAN Technology, Mobile Ad-hoc Networks (MANETs), and Wireless Sensor Networks (WSNs). Finally, a brief overview of mobile IP and TCP as well as security requirements for wireless networks is given.

Key Terms

- Broadband
- Convergence
- Data rate
- Digital convergence
- Enhanced Data rates for GSM Evolution (EDGE)
- First generation (1G)
- Frequency Division Duplexing (FDD)
- Full-duplex transmission
- General Packet Radio Service (GPRS)
- Global Positioning System (GPS)
- Global System for Mobile (GSM)
- Half-duplex transmission
- IMT-2000
- Radar
- Satellite
- Second generation (2G)
- Second generation plus (2.5G)
- Simplex transmission
- Third generation (3G)
- Time Division Duplexing (TDD)
- Universal Mobile Telecommunications System (UMTS)
- Wideband CDMA (W-CDMA)
- Wireless
- Wireless Application Service Provider (WASP)
- Wireless communications

Summary

Wireless communications have become common use today. Cellular mobile communications, remote wireless Internet connections, and wireless networks are making many network-based business activities faster and more convenient. Wireless networks and devices are found everywhere. Home users can implement wireless local area networks to connect different devices, while Bluetooth and UWB can connect devices over short distances. Fixed broadband wireless is used to transmit data at distances up to 56 kilometres, while digital cellular networks are used to transmit data at up to 2 Mbps. WLAN applications are found in a wide variety of industries and organisations, including the education, business, entertainment, travel, and health-care service. Mobility, increased network reliability, easier and less expensive installation, and support for disaster recovery are the major advantages of wireless communications. Radio-signal interference, security issues, and health risks are its potential disadvantages. Wireless communications has a lot of market potential as well as challenges for further innovations and research.

Short-Answer Type Questions with Answers

A1.1 List and describe the three types of data flow in wireless communication.
Simplex - When data moves from the transmitter to the receiver only. The receiver cannot reply.
Half-duplex - When data can move in both directions but not simultaneously. When one is transmitting, the other must listen until the transmission is finished.
ull-duplex - Data can move in both directions simultaneously.

A1.2 Discuss the advantages and disadvantages of standards.
Standards ensure that devices interoperate with those from other vendors, and create competition. Other advantages are

- Lower prices not only for consumers but also lower manufacturing costs
- No large research investments
- Standards also help protect consumer investment in technology by creating a migration path

The disadvantages are

- International standards can be a threat to industries in large countries since they benefit manufacturers in small countries and allow them to make and sell their products to the originating country, often at a much lower cost
- Some countries create their own standards instead of adopting an international one which often forces manufacturers to build more expensive equipment that complies with multiple standards and the consumer ultimately pays the higher price

A1.3 What are the main benefits of IEEE 802.11e WLAN standards
IEEE 802.11e adds QoS capabilities and enhances the MAC protocol. It also permits 802.11 WLANs to prioritise voice and video traffic.

A1.4 What are some advantages and disadvantages of IEEE 802.11a WLANs
IEEE 802.11a WLANs are faster than IEEE 802.11b and g. They operate on a 5-GHz band and are not subject to interference from cordless phones, microwaves or Bluetooth. The disadvantages are that they are not backward compatible with 802.11b, provide shorter range than 802.11b, and also do not support QoS.

A1.5 What is authentication What is the disadvantage of the authentication included in the IEEE 802.11 standards and what kind of authentication should be considered for WLANs
Authentication is the process of establishing whether a user is authorised to communicate on the network. IEEE 802.11 standard authenticates the device only, but not the user. RADIUS authentication should be considered to increase the security of WLANs.

A1.6 List the three major ways in which satellites are used. Which one is used for fixed broadband wireless
Satellites are used in scientific and environment research, weather forecast, military applications, and as reflectors. Satellites as reflectors are used for fixed broadband wireless purpose.

A1.7 List some potential applications for RFID technology.
The applications include people location within a building, parking-lot access cards, laboratory experiment tracking, timber-grade monitoring, permanent identification of valuable items such as art, and security-guard monitoring (rounds monitoring), among many other applications.

A1.8 List three specific challenges associated with RFID system implementation.
Increased network traffic, increased storage requirement, network availability, and security are some specific challenges associated with RFID system implementation.

A1.9 Which methods can be used to temporarily or permanently disable the RFID tag
Physical destruction, the kill command, locking, and blocker tags are some of the methods which can be used to temporarily or permanently disable the RFID tag.

A1.10 What is a wireless access service provider (WASP) What type of services are provided by a WASP

A WASP can design and create a wireless application and even deliver the hardware, software, security and networks as one complete package. Because many of the wireless devices, languages, and applications are so new and diverse, a WASP may have the expertise needed to get the project up and running quickly. Many WASPs may even host the application on their own wireless network, in which case the services are subscribed to, rather than being purchased.

A1.11 List some issues that can be answered by performing a wireless site survey prior to making a decision on WLAN implementation.

Issues such as location of a wireless access point and antennas, throughput required, type of client adapters, power requirements, expansion of the wireless network and impact on current design, security features and policies required, number of radio channels required, radio signal range can be resolved.

Self-Test Quiz

S1.1 UHF 900-MHz frequency band is commonly used in cellular mobile communications mainly due to
(a) regulations and standards
(b) non-availability of band below 900 MHz
(c) line-of-sight propagation available in operating area
(d) line-of-sight and reflected signals ensure the reception at mobile

S1.2 GSM cellular mobile communication service uses
(a) FDMA for multiple users
(b) FDMA for multiple channel access and TDMA for multiple users
(c) TDMA for multiple channel access
(d) different uplink and downlink modulation methods

S1.3 WLAN services use
(a) short-distance communications at high data rate
(b) short-distance communications at low data rate
(c) long-distance communications at high data rate
(d) long-distance communications at low data rate

S1.4 IEEE 802.11g WLAN devices can be as far as 100 metres apart and can send and receive data at rates up to Mbps
(a) 75
(b) 54
(c) 11
(d) 1

S1.5 Bluetooth devices provide
(a) short-distance (1–100 metres) communications up to 1 Mbps data rate
(b) short-distance (1–100 meters) communications up to 1 Gbps data rate
(c) short-distance (1–100 meters) communications up to 2 Gbps data rate
(d) short as well as long distance (1m–1km) communications up to 1 Mbps

S1.6 Bluetooth devices communicate using small radio transceivers called that are built onto microprocessor chips.
(a) transponders
(b) radio modules
(c) receivers
(d) transmitters

S1.7 Ultra Wide Band technology is used primarily for
(a) connecting wireless devices inside a home at very high speeds
(b) displaying Web pages on a cellular phone
(c) transmitting data at distances of up to 56 kilometres
(d) finding the location of a vehicle within a small city

S1.8 Security issues in a wireless device are encountered due to
(a) virus attacks and hacking of data
(b) virus attacks, hacking of data, and eavesdropping
(c) jamming of the received signals
(d) virus attacks, hacking, eavesdropping, jamming, and forcefully exhausting the energy resources

S1.9 A wireless security mechanism should provide
(a) secure access of a mobile device to the service provider
(b) authentication, integrity, and privacy of services
(c) integrity and wired-equivalent privacy of services
(d) authentication, integrity, and confidentiality of services

S1.10 GPRS is a
(a) circuit-switched-cum-packet-oriented service for mobile users
(b) packet-oriented service for mobile users
(c) asynchronous packet-oriented service for mobile users
(d) synchronous packet-oriented service for mobile users

Answers to Self-Test Quiz

S1.1 (d); S1.2 (b); S1.3 (a); S1.4 (a); S1.5 (a); S1.6 (b); S1.7 (a); S1.8 (d); S1.9 (c); S1.10 (b)

Review Questions

Q1.1 Mention the frequency bands utilised for various wireless communication standards such as AMPS, NMT, TACS, GSM, DECT, and PCS.

Q1.2 List the significant improvements introduced in the first-, second-, and third-generation standards of cellular communication systems.

Q1.3 Compare the technical parameters of US, European and Japanese cellular standards.

Q1.4 List and describe the three classifications of AMPS cellular standards.

Q1.5 Explain the key differences between 1G, 2G, 2.5G, 2.5G+, and 3G mobile communication standards.

Q1.6 Explain how a wireless LAN can be used in a classroom.

Q1.7 Describe how wireless networks can reduce installation time and cost.

Q1.8 Explain how implementing a wireless network can be helpful in case of disaster recovery in a small business.

Q1.9 A wireless device transmits radio signals over a wide open area. What are the security concerns with using wireless communications technology?

Q1.10 What are the additional features which the next-generation wireless networks is likely to have over and above the present 3G wireless technologies?

Q1.11 How is a 3G wireless network different from a 2G CDMA network?

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2

Mobile Communication Engineering

The mobile radio channel is extremely random in nature, difficult to analyse, and places fundamental limitations on the performance of wireless communication systems. Therefore, for design of wireless and mobile communication systems, it is very important to understand the fundamental features of mobile radio propagation. The study of signal propagation is vital to wireless communications because it provides prediction models for estimating the received signal power in mobile radio environment. It also helps in development of compensating techniques for the impairments introduced through wireless transmission. In this chapter, an insight into the basic propagation mechanisms and the behaviour of mobile radio channels is presented. This provides a basic understanding of how a signal propagates in the wireless medium and what effects the medium has on the signal. An overview of simulation of wireless fading channels is also included.

2.1 INTRODUCTION

The wireless medium is quite unreliable, has a relatively low bandwidth, and is of broadcast nature. But it is the wireless medium alone that supports the mobility of users. All wireless transmissions share the same medium, that is, air. It is the frequency of operation and the legality of access to the available frequency band that differentiates a variety of alternativs possible for wireless communications. For example, cellular communication networks operate around a licensed 1-GHz band, and Personal Communication Systems operate around 2-GHz band. For frequencies up to a few GHz, the radio signals can penetrate through the concrete structures, allowing indoor applications with minimal wireless infrastructure inside a building. At higher frequencies, a radio signal that is generated outdoors does not penetrate into buildings, and the signal generated indoors stays confined to it. This phenomenon imposes restrictions on the selection of a suitable frequency spectrum for wireless applications. Thus, with the wireless medium, the system capacity is limited due to allocation of a limited frequency spectrum for operation in a certain area for a specific application until some special techniques are deployed.

The three most important radio-propagation characteristics used in the design, analysis, and installation of mobile communication systems are the achievable signal coverage, the maximum data rate that can be supported by the wireless channel, and the rate of signal fluctuations in the wireless channel. For efficient data communications on wireless channels, the maximum data rate that can be supported becomes a critical design parameter. The multipath structure of the wireless channel and the fading characteristics of the multipath component of the signals being transmitted strongly influence the data-rate capabilities. This also results in the need of a proper signaling scheme and receiver design. Depending on the nature of the operating environment, and the data rates that need to be supported by an application, some characteristics are much more important than others. For example, signal coverage and slow fading are more important for low data-rate narrowband systems such as cellular voice and low-speed data applications. The multipath delay spread becomes important for high-data-rate wideband systems such as systems designed using spread-spectrum modulation schemes as CDMA and 3G cellular services.

Facts to Know!

Radio waves can travel large distances, unlike heat waves. They can also penetrate non-metallic objects, unlike light waves. Radio waves are free of some of the limitations that heat and light waves experience. Heat waves and visible light waves can be seen and felt, but radio waves are invisible. Thus, radio waves are an excellent means to transmit data without wires.

In any mobile communication system, the cell-site, or the base station, is always fixed but the subscribers are mostly on the move. Due to movement of the mobile subscriber unit, which is surrounded by many types of obstacles during operation, it is always under the influence of a varying multipath signal environment. The received signal is quite weak in such a mobile radio-operating environment which experiences multiple reflections from nearby buildings and other moving vehicles. There may also be locations where the direct signal from the transmitter received at the mobile unit is obstructed by a nearby tall building. The movement of the transmitter, receiver, or objects in between mainly causes the rate of signal fluctuations in the wireless channel in the mobile communication scenario. This particular aspect is characterised by the Doppler spread of the received signal. Due to this effect, there is great difficulty in achieving much-needed accurate timing and synchronisation between the transmitted and received radio signals, as well as the phase recovery at the receiver end.

Facts to Know!

The signal propagation path changes with the movement of the mobile unit, and/or the movement of the surroundings and environment. Even the smallest and slowest movement causes a time-variable multipath, thus random time-variable signal reception.

2.2 THE RADIO PATHS

In general, a radio path is a path traveled by the radio signal in the wireless medium from the transmitter to the receiver. The transmission path between the transmitter and the receiver can vary from a simple line-of-sight to one that is severely obstructed by natural terrain, buildings, other nearby moving vehicles, and the presence of heavy foliage (vegetation). Even the speed of the mobile unit impacts how rapidly the signal level fades as it moves. In a mobile radio environment, the following types of radio paths are generally considered.

Direct wave Path It is a radio signal path from the transmitter to the receiver that is clear from the terrain contour.

Line-of-Sight (LOS) Path It is the shortest direct radio signal path between the transmitter and receiver, a path clear from in-between buildings. In urban areas, the line-of-sight condition is generally not met because buildings and other terrain features usually block the radio signal path.

Obstructive Path It is a radio signal path between the transmitter and receiver when the terrain contour blocks the direct wave path. In indoor applications, walls, floors, and interior objects within buildings obstruct line-of-sight communications. The signal strengths of radio signal paths depend on the distance they have traveled, the obstacles they have reflected from or passed through, the location of the objects surrounding the transmitter and the receiver, and the architecture of the wireless environment. The signals may also encounter diffraction phenomenon resulting into shadow or diffraction loss in the received radio signal.

Facts to Know!

Assume that the mobile user is sitting in a car in a parking lot, near a busy highway. Although the mobile user is relatively stationary, but his/her surrounding is moving at 100 km/h. The vehicles on the highway become reflectors of radio signals. If during transmission or reception, the user also starts driving at 100 km/h, the randomly reflected signals vary at a faster rate.

When a mobile unit is nearer to the cell-site, a line-of-sight condition might be encountered. Under this situation, the average received signal at the mobile unit is higher, although the 40-dB/decade path-loss slope still exists. So the received signal at the mobile unit is a combination of a strong line-of-sight path, a ground-reflected wave, plus many weak reflected waves from surrounding buildings. This results into short-term *Rician fading*.

When a mobile unit is far away from the cell site, an out-of-sight condition is normally encountered. Still, the 40-dB/decade path-loss slope remains, however, all reflected waves from surroundings become dominant. The short-term received signal at the mobile unit observes the most severe Rayleigh fading.

2.3 THE PROPAGATION ATTENUATION

In general, the propagation path loss increases with frequency of transmission, f_c as well as the distance between the cell site and mobile, R. The radio frequency of operation affects radio-propagation characteristics and system design. For example, at frequencies lower than 500 MHz in the radio spectrum, the received signal strength loss is much less with distance. Although the available bandwidth is also not adequate, and the antenna sizes required are unusually large and thus quite impractical for large-scale system deployment. On the other hand, adequate bandwidth is available at much higher frequencies (around 1 GHz and greater than a few GHz). However, at such frequencies, the radio signals suffer a greater signal strength loss at shorter distance, and also suffer larger signal strength losses while passing through obstacles such as walls.

In a real mobile radio environment the propagation path-loss Lp varies as directly proportional to R^{γ}, where R is the distance between the transmitter and receiver, and γ is the path loss exponent, which varies between 2 and 6, depending on the actual atmospheric conditions. Table 2.1 lists typical path-loss exponent values obtained in various mobile radio environments.

Facts to Know!

Attenuation can also be caused by precipitation such as rain or snow at certain frequencies. The density of air and water vapour decreases at higher altitude. Due to this, signal attenuation decreases.

Since the propagation path loss and the received signal power are reciprocal to each other, assuming all other factors constant, we can say that the received carrier signal power, C is inversely proportional to R^{γ}, that is,

$$C \propto R^{-\gamma} \tag{2.1}$$

where C = received carrier signal power

R = distance measured from transmitter to the receiver

Let the received carrier signal power at a distance R_1 be C_1 and at a distance R_2 be C_2. The ratio in received carrier signal powers at two different distances is

Table 2.1 *Typical path-loss exponents in different environments.*

Mobile radio environment	*Path-loss exponent, γ*
Free space condition	2
Flat rural area	3
Rolling terrain rural area	3.5
Typical urban areas	2.7 to 3.5
Suburban with low-rise buildings	4
Shadowed urban areas	3 to 5
Dense urban with high-rise buildings	4.5
In building — line-of-sight conditions	1.6 to 1.8
In building — obstructed conditions	4 to 6
In factories — obstructed conditions	2 to 3
Typical mobile radio environment	4

$$(C_2 / C_1) = (R_1 / R_2)^{\gamma} \tag{2.2}$$

$$\Delta C \text{ (in dB)} = C_2 \text{ (in dB)} - C_1 \text{ (in dB)} = 10 \log (C_2 / C_1)$$

Or,
$$\Delta C \text{ (in dB)} = 10 \log (R_1 / R_2)^{\gamma} = 10\, \gamma \log (R_1 / R_2) \tag{2.3}$$

In free-space condition, $\gamma = 2$.

Therefore, $C \propto R^{-2}$ (free space)

And $\Delta C \text{ (in dB)} = C_2 \text{ (in dB)} - C_1 \text{ (in dB)} = 10 \log (C_2 / C_1)$

$$= 10 \log (R_1 / R_2)^2 = 20 \log (R_1 / R_2) \tag{2.4}$$

EXAMPLE 2.1 Signal attenuation in free-space propagation

Calculate the change in received signal strengths (in dB) in free-space propagation condition at two different distance points
(a) when the second distance point is twice the distance of the first point
(b) when the second distance point is ten times the distance of the first point
Comment on the results obtained.

Solution

Step 1. *Let the received carrier signal power at a distance R_1 be C_1 and at a distance R_2 be C_2. The change in received signal strengths (in dB), ΔC in free space propagation, between the distance points R_2 and R_1 is given by*

$\Delta C \text{ (in dB)} = 20 \log (R_1 / R_2)$

(a) *To calculate the change in received signal strengths*

Step 2. *Here, $R_2 = 2\,R_1$ (given)*

Therefore, $\Delta C \text{ (in dB)} = 20 \log (R_1 / 2\,R_1) = 20 \log (1/2)$

Hence, $\mathbf{\Delta}\text{C} = -6$ dB

Comments on the result This implies that signal strength decays at the rate of 6 dB/octave in the free space propagation environment condition.

(b) *To calculate the change in received signal strengths*

Step 3. *Now, $R_2 = 10\,R_1$ (given)*

Therefore, ΔC (in dB) $= 20 \log (R_1 / 10\,R_1) = 20 \log (1 / 10)$

Hence, $\Delta C = -20$ dB

Comments on the result. This implies that signal strength decays at the rate of 20 dB/decade in the free space propagation environment condition.

In mobile radio environment, the attenuation increases much more quickly with distance; values ranging from $\gamma = 3$ to a more typical value of $\gamma = 4$. That is, attenuation is roughly proportional to the fourth power of distance because of reflections and obstacles.

Therefore, $C \propto R^{-4}$ (mobile radio environment)

And ΔC (in dB) $= C_2$ (in dB) $- C_1$ (in dB)

Or, $= 10 \log (C_2 / C_1)$

Or, $= 10 \log (R_1 / R_2)^4 = 40 \log (R_1 / R_2)$ (2.5)

EXAMPLE 2.2 Signal attenuation in mobile radio propagation

Calculate the change in received signal strengths (in dB) in mobile radio propagation condition at two different distance points
(a) when the second distance point is twice the distance of the first point
(b) when the second distance point is ten times the distance of the first point
Comment on the results obtained.

Solution

Step 1. *Let the received carrier signal power at a distance R_1 be C_1 and at a distance R_2 be C_2. The change in received signal strengths (in dB), ΔC in mobile radio propagation, between the distance points R_2 and R_1 is given by*

$$\Delta C \text{ (in dB)} = 40 \log (R_1 / R_2)$$

(a) *To calculate the change in received signal strengths*

Step 2. *Here, $R_2 = 2\,R_1$ (given)*

Therefore, ΔC (in dB) $= 40 \log (R_1 / 2\,R_1) = 40 \log (1 / 2)$

Hence, $\Delta \boldsymbol{C} = -12$ dB

Comments on the result This implies that signal strength decays at the rate of 12 dB/octave in the mobile radio-propagation environment condition.

(b) *To calculate the change in received signal strengths.*

Step 3. *Now, $R_2 = 10\,R_1$ (given)*

Therefore, ΔC (in dB) $= 40 \log (R_1 / 10\,R_1) = 40 \log (1 / 10)$

Hence, $\Delta \boldsymbol{C} = -40$ dB

Comments on the result This implies that signal strength decays at the rate of 40 dB/decade in the mobile radio-propagation environment condition.

2.4 BASIC PROPAGATION MECHANISMS

With any communication system, the received signal always differs from the transmitted signal due to various transmission impairments. Radio propagation in free space and without any obstacles is the most ideal situation. But in mobile communication applications, the ideal situation is rarely achieved. Radio signals with frequencies above 800 MHz have extremely small wavelengths compared with the dimensions of buildings and other obstacles, so electromagnetic waves can be treated simply as optical rays. In a wireless signal-propagation environment, apart from direct waves, the receiver will get a number of reflected waves, diffracted waves and scattered waves. The mechanisms behind electromagnetic wave propagation are diverse, but can generally be attributed to mainly three basic radio propagation mechanisms, namely, reflection, diffraction, and scattering. The vectorial addition of these waves constitutes the resultant wave which will vary in strength in real time. A typical propagation effect in a mobile radio environment is illustrated in Fig. 2.1.

As shown in Fig.2.1 h_t is the height of the cell-site antenna from the earth's surface, h_r is the height of the mobile antenna from the earth's surface, and r is the distance between the cell-site and the mobile unit.

The three basic propagation mechanisms are reflection, diffraction, and scattering which influence signal propagation in a mobile communication environment are briefly described now.

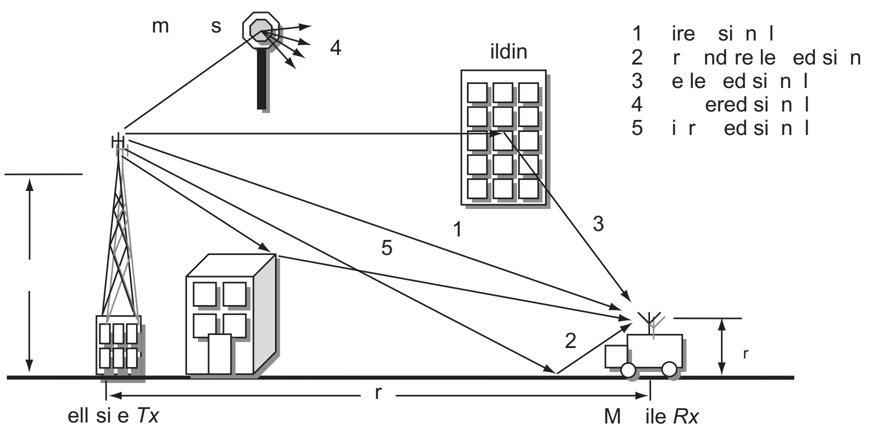

Fig. 2.1 *Radio propagation mechanisms in a mobile communication*

2.4.1 Reflection

Reflection occurs when incident electromagnetic waves are partially reflected when they impinge on obstructions of different electrical properties. A propagating electromagnetic wave impinges on objects the sizes of which are large compared to its wavelength, such as the surface of the earth, buildings, walls, etc. The electromagnetic radio waves get reflected from tall building structures which have a good amount of conductivity. Reflection can also occur due to metal reinforcement. The extent of reflection of radio waves depends on the composition and surface characteristics of the objects. The angle of reflection is equal to the angle at which the wave strikes the object and is measured by the Fresnel reflection coefficient. Upon reflection, the signal strength of the radio wave gets attenuated that depends on many factors like the frequency of the radio waves, the angle of incidence, and the nature of the medium including its material properties, thickness, homogeneity, etc. Generally, higher frequencies reflect more than lower frequencies.

As an instance, let a ground-reflected wave near the mobile unit be received. Because the ground-reflected wave has a 180^{o} phase shift after reflection, the ground wave and the line-of-sight wave may tend to cancel each other, resulting in high signal attenuation. The vector sum of the phases of the multipath received signals may give a resultant zero amplitude at certain time instants and large signal amplitude at some other time. Most of the times, the vectorial addition of these multipath reflected signals produce an undetectable signal. Further, because the mobile antenna is lower than most human-made structures in the operational area, multipath interference occurs. These reflected waves may interfere constructively or destructively at the receiver. In outdoor urban areas, the reflection mechanism often loses its importance because it involves multiple reflections that reduce the strength of the signal to negligible values. However, reflection mechanisms often dominate radio propagation in indoor applications. The reflections are a source of multipath signals which cause low strength in signal reception. Reflection results in a large-scale fading of the radio signals.

> **Facts to Know!**
>
> The term 'multipath distortion' originates from the fact that as electromagnetic waves arrives at different times, they are out of phase with one another. Since the amplitudes of multiple signals either get added or subtracted from one another, the resulting signal at the input of the receiver gets distorted.

EXAMPLE 2.3 | Effects of reflection on signal propagation

A wireless communication transmitter transmits a signal at 900 MHz. A receiver located at a distance of 1 km away from transmitter receives two signals — one directly as a line-of-sight signal and another indirectly via reflection from a building (having a height more than 10 metres), as shown in Fig. 2.2.

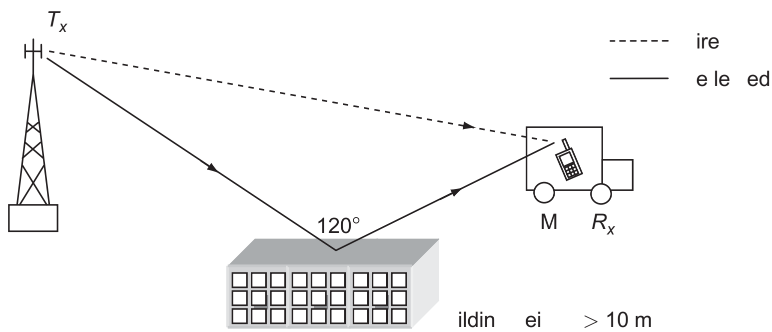

Fig. 2.2 | *Reflected and direct signals*

Give reason(s) to justify that the reflected signal causes delay in the reception. Calculate the amount of delay in the reflected signal with respect to the direct signal at the receiver.

Solution

Frequency of transmission, $f_c = 900$ MHz (given)

Step 1. *To find the wavelength of transmission,* λ_c

We know that $\lambda_c = c / f_c$

Or, $\lambda_c = 3 \times 10^8$ m/s / 900×10^6 Hz

Therefore, $\lambda_c = 0.33$ m

Step 2. *To justify that reflected signal causes delay.*

The height of the building 10 m (given)

Thus, the given height of the building is much greater than the wavelength of the transmission.

It implies that the radio signal is reflected from the surface of the obstacle of size much greater than λ_c of the radio transmissions. The reflected signal suffers a delay in reaching the receiver.

Step 3. *To find the time taken by the direct path, t_{direct}*

Distance between transmitter and receiver = 1 km or 1000 m (given)

We know that t_{direct} = distance traveled by direct path / speed of radio wave

Or, $t_{direct} = 1000 \text{ m} / 3 \times 10^8 \text{ m/s}$

Therefore, $t_{direct} = 3.33 \ \mu\text{s}$

Step 4. *To find the time taken by the reflected path, $t_{reflected}$*

Assuming that the reflected path is approximately equal to 1000 m

Angle between incident and reflected path = 120 (given)

Thus, incident angle = 120 /2 = 60

Therefore, $t_{reflected} = 1000 \text{ m} / \ (3 \times 10^8 \text{ m/s}) \times \sin 60$

Or, $t_{reflected} = 3.85 \ \mu\text{s}$

Step 5. *To calculate the delay in a reflected signal*

Delay $= t_{reflected} \quad t_{direct}$

Hence, delay = 3.85 μs 3.33 μs = 0.52 s

2.4.2 Diffraction

Diffraction is referred to the change in wave pattern caused by interference between waves that have been reflected from a surface or a point. It is based on Huygen's principle which states that all points on a wavefront can be considered as point sources for production of secondary wavelets that can combine to produce a new wavefront in the direction of propagation of the signal. Diffraction occurs when the radio path between a transmitter and receiver is obstructed by a surface with sharp irregular edges. Waves bend around the obstacle, even when a line-of-sight condition does not exist. It causes regions of signal strengthening and weakening irregularly. Diffraction can also occur in different situations such as when radio waves pass through a narrow slit or the edge of a reflector or reflect off from two different surfaces approximately one wavelength apart. At higher frequencies, diffraction depends on the geometry of the object, as well as the amplitude, phase, and polarisation of the incident wave at the point of diffraction. Figure 2.3 depicts a simple case of diffraction of a radio signal.

Diffraction is a description of how a radio signal propagates around and over an obstruction, and is measured in dB. Diffraction often results in *small-signal fading*. In effect, diffraction results in propagation into shadow regions because the diffracted field can reach a receiver, which is not in the line-of-sight of the transmitter. Because a secondary wavelet is created, it suffers a signal loss much greater than that experienced via reflection. Although the received signal strength decreases rapidly as a receiver moves deeper into the shadow region, the diffraction field still exists and often produces useful signal strength.

Consequently, diffraction is an important phenomenon of propagation impairment in outdoor applications such as in micro-cellular areas where signal transmission through buildings is virtually impossible. It is less consequential in indoor applications where a diffracted signal is extremely weak compared to a reflected signal or a signal that is transmitted through a relatively thin wall.

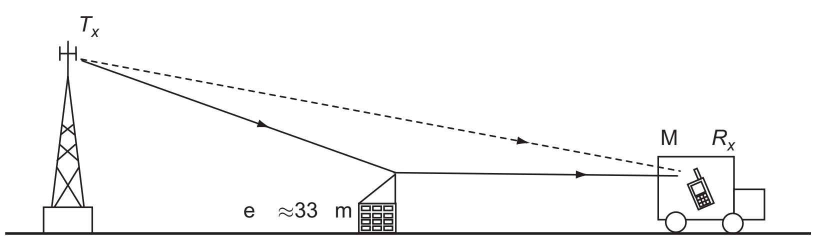

Fig. 2.3 *Diffraction of a radio signal*

In mobile communication systems, diffraction loss occurs from the blockage of secondary waves such that only a portion of the energy is diffracted around an obstacle. Most cellular systems operate in urban areas where there is no direct line-of-sight path between the transmitter and the receiver (either from the cell-site to the mobile unit or vice-versa), and where the presence of high-rise buildings causes severe diffraction loss. In many practical situations, the propagation path may consist of more than one obstruction. For example, in hilly terrains, the total diffraction loss must be computed due to all of the obstacles.

Facts to Know!

Shadowing is caused mainly by terrain features of the land-mobile wireless propagation environment. It imposes a slowly changing average signal on the Rayleigh fading statistics.

2.4.3 Scattering

Scattering is a special case of reflection caused by irregular objects such as walls with rough surfaces, vehicles, foliage, traffic signs, lamp posts, and results in many different angles of reflection and scatter waves in all directions in the form of spherical waves. Thus, due to availability of numerous objects, scattering effects are difficult to predict. Scattering occurs when the size of objects is comparable or smaller than the wavelength of the propagating radio wave, and where the number of obstacles per unit volume is large. Figure 2.4 depicts a typical case of scattering of a radio signal.

Propagation in many directions results in reduced received-signal power levels, especially far from the scatterer. So an incoming radio signal is scattered into several weaker outgoing radio signals. As a result, the scattering phenomenon is not significant unless the receiver or transmitter is located in a highly noisy environment. In a mobile radio environment, scattering provides additional radio energy levels at the receiver to what has been predicted by reflection and diffraction models alone. In radio channels, knowledge of the physical location of large distant objects, which induce scattering, can be used to accurately predict scattered signal strength levels. In a mobile radio environment, heavy foliage often causes scattering. Scattering too results in small-scale fading effects.

These three impairments to free-space propagation influence system performance in various ways depending on local conditions and as the mobile unit moving within a cell in a cellular system.

- If a mobile unit has a clear line-of-sight condition with the cell-site then only reflection may have a significant effect whereas diffraction and scattering have minor effects on the received signal levels.
- If there is no clear line-of-sight condition, such as in an urban area at busy street level, then diffraction and scattering are the primary means of signal reception.

One major adverse effect of multipath propagation is that multiple copies of a signal may arrive at different phases. If these phases add destructively, the signal level relative to noise declines, making signal detection at the receiver much more difficult and unreliable.

The second major effect of multipath propagation is increase in received data errors due to intersymbol interference in digital transmission. As the mobile unit moves, the relative location of various objects also

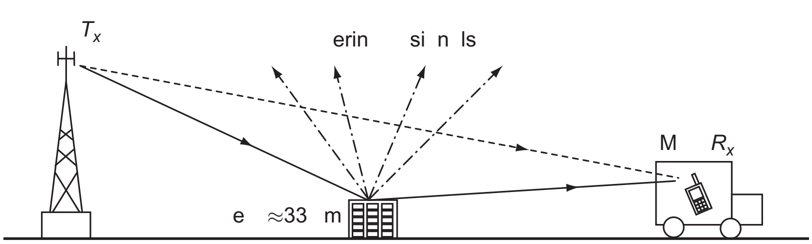

Fig. 2.4 *Scattering of radio signal*

changes; hence intersymbol interference increases to the extent that makes it difficult to design signal processing techniques that will filter out multipath effects in order to recover the intended signal with fidelity.

An extreme form of signal attenuation is blocking or shadowing of radio signals, which is caused by obstacles much larger in size than the wavelengths of the operating signals such as a small wall, trees, or a large vehicle on the street.

Another form of propagation effect is the effect of refraction. Refraction occurs because the velocity of the electromagnetic waves depends on the density of the medium through which it travels. Waves that travel into a denser medium are bent towards the medium. This is the reason for line-of-sight radio waves being bent towards the earth since the density of the atmosphere is higher closer to the earth.

2.5 MOBILE RADIO CHANNEL

In a mobile communication system, a signal experiences multipath propagation which causes rapid signal-level fluctuations in time, called *fading*. Mobile radio channels introduce noise, fading, interference, and other distortions into the signals that they transmit. Fading effects that characterise mobile radio communication are large-scale fading and small-signal fading. If there is a large number of multiple reflective paths with no line-of-sight signal path, it is *Rayleigh fading*. The Rayleigh flat-fading channel model assumes that the channel induces amplitude which varies in time according to Rayleigh distribution. When there is a dominant non-fading signal component present, the small-signal fading envelope is described by a *Rician fading*. Small-signal fading results into signal dispersion and time-variant behaviour of the channel.

Rayleigh and Rician fading phenomena include multipath scattering effects, time dispersion, and Doppler shifts that arise from relative motion between the transmitter and receiver. The major paths result in the arrival of delayed versions of the signal at the receiver. In addition, the radio signal undergoes scattering on a local scale for each major path. Such local scattering is typically characterised by a large number of reflections by objects near the mobile. These irresolvable components combine at the receiver and give rise to the phenomenon known as *multipath fading*. As a result, each major path behaves as a discrete fading path.

Typically, the fading process is characterised by a Rayleigh distribution for a non-line-of-sight path and a Rician distribution for a line-of-sight path. In mobile radio channels, the Rayleigh distribution is commonly used to describe the statistical time varying nature of the received envelope of a flat fading channel, or the envelope of an individual multipath components. The relative motion between the transmitter and receiver causes Doppler shifts. Local scattering typically comes from many angles around the mobile. This scenario causes a range of Doppler shifts, known as the Doppler spectrum. The maximum Doppler shift corresponds to the local scattering components whose direction exactly opposes the mobile's trajectory.

Facts to Know!

The effects of multipath include constructive and destructive interference, and phase shifting of the signal. This causes Rayleigh fading, and its standard statistical model gives Rayleigh distribution. Rayleigh fading with a strong line-of-sight content is said to have a Rician fading, or Rician distribution.

2.5.1 Multipath Fading

Fading of signal received by the mobile unit is an inherent problem in mobile communication. As the location of the mobile unit keeps on changing in real time, the resultant radio signal incident on its antenna varies continuously. Multipath in the mobile communication channel creates small-scale fading effects such as rapid changes in signal strength over a small time interval or small distance traveled by a mobile; random frequency modulation due to varying Doppler shifts on different multipath signals; and time dispersion caused by multipath propagation delays. Fading is the rapid fluctuation of a radio signal's amplitude in a short time or over a short distance.

> **Facts to Know!**
>
> In wireless communications, fading may either be due to multipath propagation, referred to as multipath induced fading, or due to shadowing from obstacles affecting the propagation of electromagnetic waves, sometimes referred to as shadow fading.

In reality, the received signal rapidly fluctuates due to the mobility of the mobile unit causing changes in multiple signal components arriving via different paths. These multiple waves can combine constructively or destructively. Multipath waves are also generated because the antenna height of the mobile unit is lower than its typical surrounding structures such as in built-up urban areas of operation, and the operating wavelength is much less than the sizes of the surrounding structures at the mobile unit. The sum of multipath waves causes a signal-fading phenomenon. The rapid fluctuation of the signal amplitude is referred to as small-signal fading, and it is the result of movement of the transmitter, the receiver, or objects surrounding them. Over a small area, the average value of the received signal is considered to compute the propagation path loss and received signal strength. But the characteristics of the instantaneous signal level are also important in order to design receivers that can mitigate these effects.

In fact, there are two main reasons that contribute to the rapid fluctuations of the signal amplitude. The first, caused by the addition of signals arriving via different paths, is referred to as multipath fading. The second, caused by the relative movement of the mobile unit towards or away from the cell-site transmitter, is called *Doppler effect*. Other factors that influence small-scale fading include multipath propagation, speed of the mobile, speed of the surrounding objects, and the transmission bandwidth of the signal. For a particular service area, the fading effects of the received signal at the mobile unit need to be analysed towards the effort of designing a reliable mobile communication system. Suitable diversity reception or signal-processing techniques need to be provided to minimise the impact of fading.

Multipath fading results in fluctuations of the signal amplitude because of the addition of signals arriving with different phases. This phase difference is caused due to the fact that signals have traveled different path lengths. Because the phase of the arriving paths are too changing rapidly, the received signal amplitude undergoes rapid fluctuation that is often modeled as a random variable with a particular distribution, called *Rayleigh distribution*. The multipath waves at the mobile receiver bounce back and forth due to the surrounding buildings and other structures, as shown in Fig. 2.5. When a mobile unit is stand-still, its receiver only receives a signal strength at that spot, so a constant signal is observed. When the mobile unit is moving, the fading structure of the wave in the space is received. It is a multipath fading which becomes fast as the vehicle moves faster.

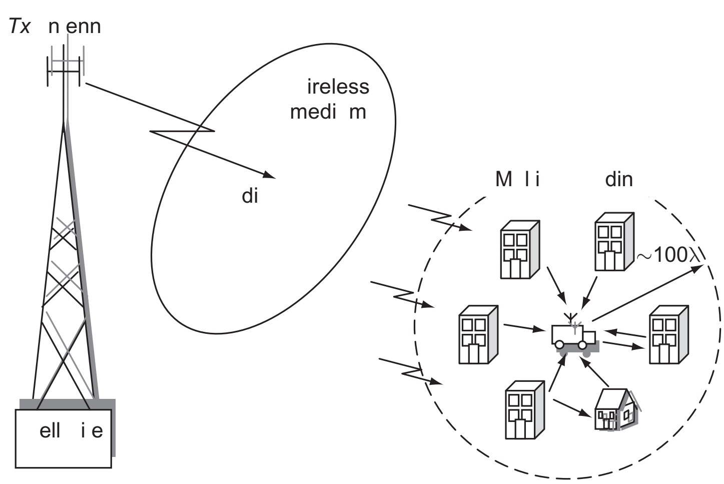

Fig. 2.5 *Multipath fading in a mobile radio environment*

2.5.2 Types of Small-Scale Fading

The type of fading experienced by a signal propagating through a mobile communication channel depends on the nature of the transmitted signal with respect to the characteristics of the wireless channel, the speed of the mobile, and the direction of motion of the mobile with respect to the incoming received signal from the cell-site transmitter. Fading effects in a mobile radio environment can be classified as

– fading effects due to multipath time delay spread; and
– fading effects due to Doppler spread.

Due to multipath time-delay spread, fading effects can also be classified as flat fading and frequency selective fading. ***Flat fading***, or *non-selective fading*, is that type of fading in which all frequency components of the received signal fluctuates in the same proportions simultaneously. Flat fading occurs when the radio channel has a constant gain and linear phase response but its bandwidth is greater than that of the transmitted signal. It implies that the desired signal bandwidth is narrower than, and completely covered by, the spectrum affected by the fading. In flat fading, the multipath structure of the channel is such that the spectral characteristics of the transmitted signal are preserved at the receiver. However, the strength of the received signal changes with time due to fluctuations in the gain of the channel caused by multipath.

In a flat fading channel, sometimes referred to as a narrowband channel, the bandwidth of the transmitted signal is much larger than the reciprocal of the multipath time-delay spread of the channel. The bandwidth of the applied signal is narrow as compared to that of the wireless channel. The distribution of the instantaneous gain of flat-fading channels can be best described by Rayleigh distribution and is important for designing wireless communication links. Typical flat-fading channels cause deep fades, and can be best corrected by increasing the transmitter power by 20 or 30 dB in order to achieve low bit-error rates during times of deep fades as compared to systems operating over non-fading channels.

Facts to Know!

The effect of time-delay spread can also be interpreted as a frequency-selective fading effect. This effect may impose a limit on the BER performance of high-speed digital communication systems due to severe wave form distortions in the demodulated signal at the receiver.

Frequency-selective fading affects unequally the different spectral components of a radio signal. Selective fading is usually significant only relative to the bandwidth of the overall wireless communication channel. If the signal attenuates over a portion of the bandwidth of the signal, the fading is considered to be selective in frequency domain. Frequency selective fading on the received signal occurs when a radio channel has a constant gain and linear phase response, but the channel bandwidth is less than that of the transmitted signal. Under such conditions, the channel impulse response has a multipath delay spread which is greater than the reciprocal bandwidth of the transmitted signal. The received signal includes multiple versions of the transmitted signal which are faded and delayed in time, and hence the received signal is distorted. Frequency selective fading is due to time dispersion of the transmitted symbols within the channel, and the channel induces intersymbol interference. Because this effect varies by frequency, fading is different at different frequencies and it is extremely difficult to counter its impact or compensate for the signal loss. Frequency-selective fading channels are also known as *wideband channels* since the bandwidth of the transmitted signal is wider than the bandwidth of the channel impulse response.

As an example, suppose a mobile receiver moves directly away from the transmitting antenna but toward a reflecting surface. This particular scenario is depicted in Fig. 2.6.

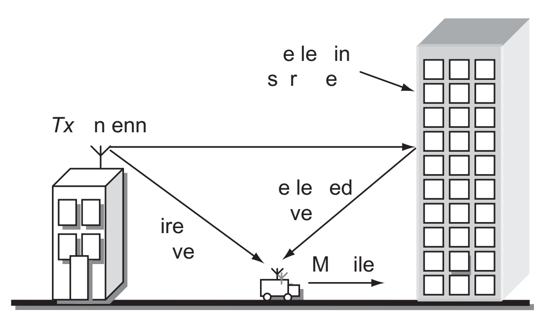

Fig. 2.6 *Fast fading in a mobile environment*

> **Facts to Know!**
>
> Frequency-selective fading channels are also dispersive, which means that the signal energy associated with each symbol is spread out in time. This causes transmitted symbols that are adjacent in time to interfere with each other. Equalisers are often used to compensate for the effects of the intersymbol interference.

If the two signals are in phase at a given point, they will add. As the mobile moves forward a distance of $\lambda_c/4$, the direct path is increased and the reflected path is reduced by the same amount, resulting in a total phase shift of 180 degrees, resulting into partial cancellation of the signal (which means the signal may fade up to 50 dB in worst cases). When the mobile moves another distance of $\lambda_c/4$, the signals are once again in phase. Thus, the fades occur each time the mobile moves a distance of $\lambda_c/2$. Given the frequency of the the signal and the speed of the mobile, it is easy to estimate the time between fades. The time between fades is given by

$$T_f = (\lambda_c /2) / V_m = \lambda_c / (2V_m) = c / (2 f_c V_m) \tag{2.6}$$

Due to Doppler spread, fading effects can also be classified as fast fading and slow fading. Depending on how rapidly the transmitted baseband signal changes as compared to the rate of change of the channel, a wireless channel is classified as a fast fading or slow-fading channel. In ***fast-fading*** conditions, as the mobile unit moves down a street in an urban environment, rapid fluctuations in received signal strength occur over distances of about one-half a wavelength. The channel impulse response changes rapidly within the symbol duration. That is, the coherence time of the channel is smaller than the symbol period of the transmitted signal. This causes frequency dispersion, also called *time-selective fading*, due to Doppler spreading. This results into signal distortion which increases with increasing Doppler spread relative to the bandwidth of the transmitted signal. Therefore, a signal undergoes fast fading if the symbol period $T_s > T_f$, the time between fades. Fast fading occurs when the rate of change of the channel characteristics is faster than the rate of change of the information data signal, and results in distortion.

EXAMPLE 2.4 | Slow or fast fading

A mobile subscriber travels at a uniform speed of 60 km/h. Compute the time between fades if the mobile uses
(a) a cellphone operating at 900 MHz
(b) a PCS phone operating at 1900 MHz
Comment on the results obtained.

Solution

Speed of the mobile, $V_m = 60$ km/h (given)
$= 60 \times 10^3$ m / 3600 s $= 16.7$ m/s

Time between fades is given by the expression, $T_f = c / (2 f_c V_m)$
where c is the speed of radio waves $= 3 \times 10^8$ m/s

(a) *To compute time between fades for a mobile operating at 900 MHz*

Step 1. *Frequency of operation, $f_c = 900$ MHz or 900×10^6 Hz (given)*

Time between fades, $T_f = (3 \times 10^8$ m/s / $(2 \times 900 \times 10^6$ Hz $\times$ 16.7 m/s)
Hence, time between fades at 900 MHz = 10 ms

(b) *To compute time between fades for a mobile operating at 1900 MHz*

Step 2. *Frequency of operation, $f_c = 1900$ MHz or 1900×10^6 Hz (given)*

Time between fades, $T_f = (3 \times 10^8$ m/s / $(2 \times 1900 \times 10^6$ Hz $\times$ 16.7 m/s)
Hence, time between fades at 1900 MHz = 4.7 ms

Comments on the results It is observed that the rapidity of the fading increases with the frequency of the transmission at the same speed of the mobile vehicle.

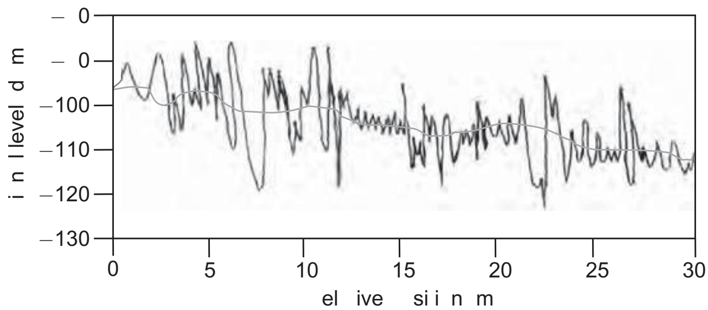

Fig. 2.7 *Typical fast and slow fading in an urban mobile radio environment*

In mobile cellular applications operating at 900 MHz (corresponding to a wavelength of 0.33 m) in an urban environment, a typical example of the spatial variation of a received signal amplitude is illustrated in Fig.2.7.

The signal may fade in a range of about 40 dB (10 dB above and 30 dB below the average signal) over a short distance. If the mobile unit moves fast, the rate of fluctuation of the radio signal also increases. This type of rapidly changing fading phenomenon, known as fast fading, not only affects mobile phones in automobile vehicles, but even a mobile phone user walking through an urban street.

In a ***slow fading channel***, the channel impulse response changes at a rate much slower than the transmitted baseband signal. As the mobile subscriber covers distances well in excess of a wavelength, the urban environment changes, as the mobile subscriber crosses buildings of different heights, busy intersections, vacant spaces, and so forth. Over these longer distances, there is a change in the average received power level about which the rapid fluctuations occur. This is indicated by the slowly changing waveforms. In the frequency domain, this implies that the Doppler spread of the channel is much less than the bandwidth of the baseband signals. Therefore, a signal undergoes slow fading if the symbol period $T_s << T_f$.

Depending on the environment and the surroundings, and the location of objects, the received signal strength for the same distance from the transmitter will be different. In fact, the actual received signal strength will vary around the mean value of the signal. This variation of the signal strength due to location is often referred to as ***shadow fading,*** which is similar to slow fading. Shadow fading is typically modeled by attenuation in signal amplitude that follows a log-normal distribution. The variation in shadow fading is specified by the standard deviation of the logarithm of this attenuation.

The problem caused by shadow fading is that all locations at a given distance may not receive sufficient signal strength for detecting the data accurately. In order to achieve sufficient signal coverage, the technique employed is to add a fade margin to the path loss or the received signal strength. This is more important at the edge of a cell or near the fringe cell areas. This fading margin can be applied by increasing the transmit power and keeping the same cell size, or reducing the cell size.

Facts to Know!

The term 'shadow fading' arises due to the fact that very often the fluctuations around the mean value are caused due to the signal being blocked from the receiver by buildings in outdoor areas, walls inside buildings, and other objects in the operating environment. It is similar to slow fading because the signal variations are much slower with distance than fading caused due to multipath.

2.5.3 Effects of Multipath Fading

Fading may cause rapid changes in volume, random frequency modulation, echoes, distortion, or a dropped call. The listener notices all these effects of fading. Small-signal fading results in very high bit-error rates. In order to overcome the effects of small-signal fading, it is not possible to simply increase the transmit power because this will require a huge increase in the transmit power. A variety of techniques are used to mitigate the effects of small-signal fading such as error control coding with interleaving, diversity schemes, and using directional antennas at the cell-site. Diversity techniques are useful to overcome the effects of fast fading by providing multiple copies of the signal at the receiver. Because the probability that all these copies suffer fading is small, the receiver is able to correctly decode the received data. *requency hopping* is another technique that can be used to combat fast fading. Because all frequencies are not simultaneously under fade, transmitting data by hopping to different frequencies is an approach to combat fading.

2.5.4 Multipath Delay Spread

Multipath interference is the reflection of radio signals from concrete structures that results in multiple copies of the received signal. Multipath interference can allow radio signals to reach hard-to-reach areas. It can also create some problems such as delay spread which occurs when several signals reach a receiver at different times due to different lengths of transmission paths. Delay spread also occurs due to Rayleigh fading which results from the signal's amplitude and phase being altered by reflections.

In a digital communication system, the delay spread along with fading causes intersymbol interference, thereby limiting the maximum symbol rate of a digital multipath channel. If the multipath delay spread is comparable to or larger than the symbol duration, the received waveform spreads into neighbouring symbols and produces intersymbol interference. The intersymbol interference results in irreducible errors that are caused in the detected signal. Figure 2.8 shows the multiple signals received at different multipaths.

Since each radio signal path has a different path length, the time of arrival for each signal path is different. The smearing or spreading-out effect of the received radio signal is called delay spread. For a low bit-error-rate (BER) performance of a digital transmission, the transmission data rate, T_r should be

$$T_r < 1 / (2\, T_d) \tag{2.7}$$

where T_d is the delay spread.

The average delay spread is typically about 3 microseconds for an urban area and up to 10 microseconds in hilly terrain. A measure of the data rate that can be supported over the channel without additional receiver techniques is determined by the RMS multipath delay spread values. The RMS delay spread varies depending on the type of the operating environment. In urban microcells, the RMS delay spread is of the order of a few microseconds. In indoor applications, it could be as small as 30 nanoseconds in residential areas or as large

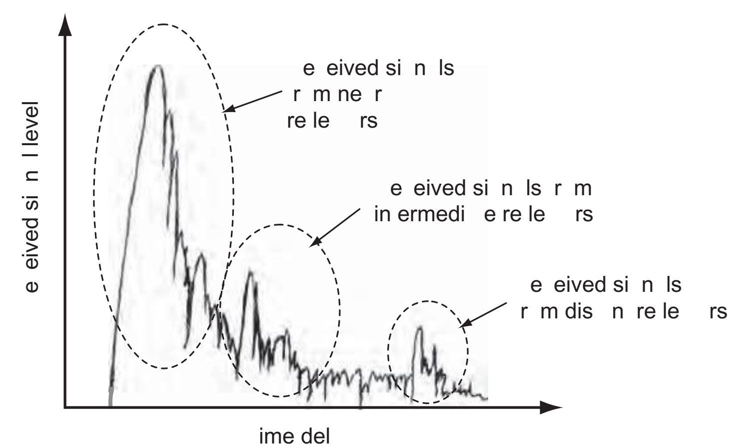

Fig. 2.8 *Delay spread of a received signal*

as 300 nanoseconds in factory environments. This means that the maximum data rate which can be supported in outdoor applications is about 50 kbps (at 4 microseconds of RMS delay spread) and in indoor applications, about 6.7 Mbps (at 30 nanoseconds of RMS delay spread). Table 2.2 shows the typical values of delay spread in different operating environments.

Table 2.2 *Delay spread in different operating environments*

Type of environment	*Delay spread (s)*
Inside the building	< 0.1
Open area	< 0.2
Sub-urban area	0.5
Urban area	3

2.5.5 Doppler Shift

There is always a relative motion between the cell-site transmitter and the mobile receiver. As a result, Doppler effect occurs in the shift of the received carrier frequency. Doppler spectrum is the spectrum of the fluctuations of the received signal strength. Multipath fading provides the distributions of the amplitude of a radio signal. It is important to know for what time a signal strength will be below a pre-defined threshold value, that is, the duration of fade, and how often it crosses a threshold value, that is, frequency of transitions or fading rate. Doppler effect results in the inaccurate operation of the system. Proper compensation technique needs to be implemented to minimise this effect. A study of Doppler spectrum is important to design the coding and interleaver schemes for efficient performance.

Thus, multipath propagation, speed of mobile unit, speed of reflecting objects, and Doppler shift are the main causes of fading. Multipath propagation can result in a positive or negative Doppler shift. As mobile unit moves around, the resulting multipath reception of waves reflected from different objects can also result in a positive or negative Doppler shift. The Rayleigh distribution is often used to model the received envelope of such a signal in a statistical, time-varying way. Rayleigh fading is also called multipath fading in the mobile radio environment. Speed of reflecting objects can induce their own Doppler shift in the reflected wave. Doppler frequency or Doppler shift is given by

$$f_d = (1/\lambda_c)\, V_m \cos\theta \tag{2.8}$$

where λ_c is the wavelength of the carrier signal, V_m is the relative velocity of the mobile, the angle θ is between the motion of the mobile and direction of arrival of the scattered waves, and $V_m \cos\theta$ represents the velocity component of the motion of the mobile in the direction of the incoming signal.

The maximum Doppler frequency will be obtained when the mobile unit is moving in line with the direction of the received signal, that is, $\theta = 0^o$ or $\cos\theta = 1$. Then from Eq. (2.8), the maximum Doppler frequency is given by

$$f_{dm} = V_m / \lambda_c = V_m\, f_c / c \tag{2.9}$$

where f_c is the frequency of transmission in Hz, V_m is the speed of the mobile and c is speed of light in same units.

When a pure sinusoidal carrier signal having frequency f_c is transmitted, the received signal spectrum, called the Doppler spectrum, will have components in the range $f_c - f_d$ to $f_c + f_d$, corresponding to whether the direction of motion of the mobile is away from or towards the direction of the received signal respectively. This simply means that Doppler shift will be positive or negative depending on whether the mobile receiver is moving toward or away from the base station transmitter.

EXAMPLE 2.5 Doppler shift frequency

Consider a base-station transmitter operating at 900 MHz carrier frequency. For a mobile moving at a speed of 72 km/h, calculate the received carrier frequency if the mobile is moving
(a) directly away from the base-station transmitter
(b) directly towards the base-station transmitter
(c) in a direction which is 60 degrees to the direction of arrival of the transmitted signal
(d) in a direction perpendicular to the direction of arrival of the transmitted signal

Solution

Carrier frequency of base station transmitter, $f_c = 900$ MHz (given)
Speed of the mobile, V_m = 72 km/h (given)
Or, $V_m = (72 \times 10^3)/3600 = 20$ m/s

(a) *To calculate received carrier frequency when the mobile is moving directly away from the base-station transmitter*

Step 1. *In the given case,* $\theta = 180^o$, $\cos\theta = \cos 180^o = -1$
So the Doppler shift is negative.

Step 2. *Doppler frequency, or Doppler shift, is given by*

$f_d = (1/\lambda_c)\, V_m$ where $\lambda_c = (c / f_c)$
Or, $f_d = (f_c / c)\, V_m = (900 \times 10^6 \text{ Hz} / 3 \times 10^8 \text{ m/s}) \times 20$ m/s
Or, $f_d = 60$ Hz

Step 3. *The received carrier frequency at the mobile* $= f_c - f_d$
$= 900 \times 10^6$ Hz – 60 Hz
$= 899.99994$ MHz

(b) *To calculate the received carrier frequency when the mobile is moving directly towards the base-station transmitter.*

Step 4. *In this case,* $\theta = 0^o$, $\cos\theta = \cos 0^o = +1$.
So the Doppler shift is positive.

Step 5. *Doppler frequency or Doppler shift is given by*

$f_d = (f_c / c)\, V_m = (900 \times 10^6 \text{ Hz} / 3 \times 10^8 \text{ m/s}) \times 20$ m/s
Or, $f_d = 60$ Hz

Step 6. *The received carrier frequency at the mobile* $= f_c + f_d$
$= 900 \times 10^6$ Hz + 60 Hz
$= 900.00006$ MHz

(c) *To calculate received carrier frequency when the mobile is moving in a direction which is 60 degrees to the direction of arrival of the transmitted signal*

Step 7. *In this case,* $\theta = 60^o$, $\cos\theta = \cos 60^o = 0.5$.
So the Doppler shift is positive.

Step 8. *Doppler frequency or Doppler shift is given by*

$$f_d = (f_c / c)\, V_m \cos 60^o$$
$$= (900 \times 10^6 / 3 \times 10^8) \times 20 \times 0.5$$
$$= 30 \text{ Hz}$$

Step 9. *Hence, the received carrier frequency at the mobile* $= f_c + f_d$
$= 900 \times 10^6$ Hz + 30 Hz
$= 900.00003$ MHz

***(d)** To calculate received carrier frequency when the mobile is moving in a direction perpendicular to the direction of arrival of the transmitted signal*

***Step 10.** In this case, $\theta = 90^{o}$, $\cos\theta = \cos 90^{o} = 0$.*

So there is no Doppler shift.

***Step 11.** The received signal frequency is the same as the transmitted frequency.*

Hence, the received carrier frequency = 900 MHz

Facts to Know!

The rate of variations of the signal in a mobile environment is frequently described as Doppler's spread.

In mobile radio applications, the Doppler spectrum or Doppler spread for a Rayleigh fading channel is usually modeled by the following expression:

$$D(\lambda) = (0.16 / f_{dm}) \times \; 1 - (\lambda_c / f_{dm})^{2 \; -0.5} \quad \text{for} \quad -f_{dm} <= \lambda_c <= f_{dm} \qquad (2.10)$$

where f_{dm} is the maximum Doppler frequency possible and is related to the velocity V_m of the mobile via the expression $f_{dm} = V_m / \lambda_c$ and λ_c is the wavelength of the radio signal.

EXAMPLE 2.6 Vehicle speed versus Doppler spread

Determine the maximum speed of a vehicle in a mobile communication system experiencing a maximum Doppler frequency shift of 70 Hz and a frequency of transmission 900 MHz.

Solution

The frequency of transmission, f_c = 900 MHz (given)

The maximum Doppler frequency shift, f_{dm} = 70 Hz (given)

We know that $f_{dm} = V_m / \lambda_c$, where V_m is the maximum speed of the vehicle.

Or, $V_m = f_{dm} \times \lambda_c$

***Step 1.** To calculate the wavelength of the transmission signal, λ_c*

$\lambda_c = c / f_c = (3 \times 10^8 \text{ m/s}) / (900 \times 10^6 \text{ Hz}) = 0.333$ m/s

***Step 2.** To calculate the maximum speed of the vehicle, V_m*

Therefore, $V_m = 70 \text{ Hz} \times 0.333 \text{ m/s} = 23.3$ m/s or 84 km/h

Hence, the maximum speed of the vehicle = 84 km/h

EXAMPLE 2.7 Doppler frequencies versus mobile antenna beamwidth

A mobile receiver is tuned to a transmission at 800 MHz and receives signals with Doppler frequencies ranging from 10 Hz to 50 Hz when moving at a uniform speed of 80 km/h. What is the beamwidth of the mobile antenna?

Solution

Carrier frequency of transmission, f_c = 800 MHz (given)

Doppler frequency 1, f_{d1} = 10 Hz (given)

Doppler frequency 2, f_{d2} = 50 Hz (given)

Speed of the mobile, V_m = 80 km/h (given)

Or, $V_m = (80 \times 10^3)/3600 = 22.222$ m/s

We know that $f_d = (1/\lambda_c)\, V_m \cos\theta$

Or, $\cos\theta = (\lambda_c \times f_d) / V_m$

***Step 1.** To calculate the wavelength of the transmission signal, λ_c*

$\lambda_c = c / f_c = (3 \times 10^8 \text{ m/s}) / (800 \times 10^6 \text{ Hz}) = 0.375$ m/s

Step 2. *To calculate θ_1 at Doppler frequency*

$\cos\theta_1 = (\lambda_c \times f_{d1}) / V_m$

$\cos\theta_1 = (0.375 \times 10) / 22.222 = 0.16875$

Or, $\theta_1 = \cos^{-1}(0.16875) = 80.285^\circ$

Step 3. *To calculate θ_2 at Doppler frequency*

$\cos\theta_2 = (\lambda_c \times f_{d2}) / V_m$

$\cos\theta_2 = (0.375 \times 50) / 22.222 = 0.84376$

Or, $\theta_2 = \cos^{-1}(0.84376) = 32.461^\circ$

Step 4. *To calculate beamwidth of mobile antenna*

Hence, beamwidth of mobile antenna $= \theta_1 - \theta_2 = 47.824^\circ$

It is possible to relate the time rate of change of the received signal to the signal level and velocity of the mobile. The level crossing rate and average fade duration of a Rayleigh fading signal are two important statistics which are useful for designing error control codes and diversity schemes to be used in mobile communication systems.

The ***level-crossing rate*** is defined as the expected rate at which the Rayleigh fading envelope, normalised to the local RMS signal level, crosses a specified threshold level in a positive-going direction. The average number of level crossings per second at a specified level L is given by

$$N_L = 2.5\, f_{dm}\, \rho\, e^{-\rho^2} \tag{2.11}$$

$e^{-\rho^2}$

where f_{dm} is the maximum Doppler frequency given by V_m / λ_c.

ρ is the value of the specified level L, normalised to the local rms amplitude of the fading envelope, that is, L/L_{rms}.

Or,
$$N_L = 2.5\, (V_m / \lambda_c)\, \rho\, e^{-\rho^2} \tag{2.12}$$

Thus, the level-crossing rate is a function of the mobile speed V_m. There are few crossings at both high and low levels, with the maximum rate occurring at $\rho = 0.707$ (that is, at a level of 3 dB below the rms level).

EXAMPLE 2.8 Doppler frequency in a Rayleigh fading channel

Consider a Rayleigh fading signal experiencing a maximum Doppler frequency of 20 Hz. The carrier frequency is 900 MHz. Compute

(a) the positive-going level-crossing rate for $\rho = 1$

(b) maximum velocity of the mobile for the given Doppler frequency

Solution

Carrier frequency of transmission, $f_c = 900$ MHz (given)

Maximum Doppler frequency, $f_{dm} = 20$ Hz (given)

(a) *To compute the positive-going level crossing rate, N_L*

Step 1. *Normalized specified level, $\rho = 1$ (given)*

Average number of level crossings per second at a specified level is given by:

$$N_L = 2.5\, f_{dm}\, \rho\, e^{-\rho 2}$$

Therefore, $N_L = 2.5 \times 20 \times 1 \times e^{-1} =$ **18.39 crossings per second**

(b) *To compute maximum velocity of the mobile, V_m*

Step 2. *We know that $f_{dm} = V_m / \lambda_c$*

Or, $V_m = f_{dm} \times \lambda_c = f_{dm} \times c / f_c$

where $c = 3 \times 10^8$ m/s
Therefore, $V_m = (20 \text{ Hz} \times 3 \times 10^8 \text{ m/s}) / (900 \times 10^6 \text{ Hz})$
Hence, $V_m = 6.67$ m/s or 24 km/h

From the Doppler spread, it is possible to obtain the fade rate as well as the fade duration for a given mobile velocity. ***Fade rate*** is defined as the number of times that the signal envelope crosses the threshold value in a positive-going direction per unit time. Usually, the fade rate is related to the carrier wavelength λ_c, the velocity of the mobile user V_m, and the number of multipaths. The average fade rate is given by $2\ V_m / \lambda_c$, that is,

$$\text{average fade rate} = 2\ V_m / \lambda_c \tag{2.13}$$

The ***average fade duration*** is defined as the average period of time for which the received signal is below a specified level L. Fade duration is defined for which the signal is below a given threshold value. It is a random variable and usually, average fade duration is used. For a Rayleigh fading signal, the average fade duration as a function of ρ and f_{dm} can be expressed as

$$\text{Average fade duration } \check{T} = 0.4\,(e^{\rho^2} - 1) / (f_{\text{dm}}\,\rho) \tag{2.14}$$

where ρ is the value of the specified level L, normalised to the local rms amplitude of the fading envelope, that is, L/L_{rms}, and f_{dm} is the maximum Doppler frequency given by V_m / λ_c.

There is a fade margin built into the link budget of the mobile communication system. The average fade duration of a received signal enables one to determine the most likely number of signaling bits that may be lost during a fade. It primarily depends upon the speed of the mobile, and decreases as the maximum Doppler frequency becomes large. It is appropriate to evaluate the receiver performance by determining the rate at which the input received signal falls below a given level L, and on the average how long it remains below the level. Similarly, depth of fading is defined as the ratio between the mean square value and the minimum value of the fading signal. It is also a random variable and usually, the average depth of fading is used. This is useful for establishing the relationship between the signal-to-noise ratio during a fade to the instantaneous available BER.

EXAMPLE 2.9 | Fade duration

Assume that a bit error occurs whenever any portion of a bit encounters a fade for which $\rho < 0.1$. For a given maximum Doppler frequency of 20 Hz,

(a) What is the average fade duration for threshold levels $\rho = 0.01$, $\rho = 0.1$, $\rho = 0.707$, and $\rho = 1$?

(b) For a binary digital modulation with a data rate of 50 bps, is the Rayleigh fading slow or fast corresponding to $\rho = 0.707$?

(c) What is the average number of bit errors per second for the given data rate of 50 bps?

Solution

Maximum Doppler frequency, $f_{dm} = 20$ Hz (given)
It is given that a bit error occurs whenever any portion of a bit encounters a fade for which $\rho < 0.1$.

(a) *To compute the average fade duration for $\rho = 0.01, 0.1, 0.707$, and 1*

We know that average fade duration, $\check{T} = 0.4\,(e^{\rho^2} - 1) / (f_{dm}\,\rho)$

Step 1. *For $f_{dm} = 20$ Hz, $\rho = 0.01$;*

$$\check{T} = 0.4\,(e^{0.0001}\quad 1) / (20 \text{ Hz} \times 0.01) = 199\quad \text{s}$$

Step 2. *For $f_{dm} = 20$ Hz, $\rho = 0.1$;*

$$\check{T} = 0.4\,(e^{0.01}\quad 1) / (20 \text{ Hz} \times 0.1) = 2 \text{ ms}$$

Step 3. *For $f_{dm} = 20$ Hz, $\rho = 0.707$;*

$$\check{T} = 0.4\,(e^{0.5}\quad 1) / (20 \text{ Hz} \times 0.707) = 18 \text{ ms}$$

Step 4. *For $f_{dm} = 20$ Hz, $\rho = 1$;*

$$\check{T} = 0.4\ (e^{1}\quad 1) / (20 \text{ Hz} \times 1) = 34 \text{ ms}$$

(b) *To determine slow or fast Rayleigh fading corresponding to $\rho = 0.707$*

Step 5. *Data rate = 50 bps (given)*

Bit period = 1/50 = 20 ms

Step 6. *The average fade duration for $\rho = 0.707, \check{T} = 18$ ms (as computed in Step 3 above)*

It is observed that the given bit period of 20 ms exceeds the average fade duration of 18 ms.
Hence, the signal with a binary digital modulation @ 50 bps data rate undergoes fast Rayleigh fading.

(c) *To determine the average number of bit errors per second*

Step 7. *Since it is given that a bit error occurs whenever any portion of a bit encounters a fade for which $\rho < 0.1$, the average fade duration corresponding to $\rho = 0.1$ is computed to be 2 ms as in Step 2 above. This is less than the duration of one bit period, that is, 20 ms for a given data rate of 50 bps, as calculated in Step 5 above.*

Therefore, only one bit will be lost during a fade.

Step 8. *The average number of level crossings per second for $\rho = 0.1$ can be computed using the relationship*

$$N_L = 2.5\, f_{dm}\, \rho\, e^{\ \rho 2}$$

Or, $N_L = 2.5 \times 20 \text{ Hz} \times 0.1 \times e^{-\,0.01}$

Hence, $N_L = 4.95$ crossings per second

Step 9. *A bit error is assumed to occur whenever a portion of a bit encounters a fade. Since the average fade duration of 2 ms spans only a fraction of a bit duration of 20 ms, the total number of bits in error is same as average number of level crossings per second or 4.95, that is, approximately 5 bits per second.*

Step 10. *Bit-Error-rate (BER) is defined as the number of bits in error in one second divided by the total number of bits transmitted in one second (that is, a data rate of 50 bps).*

Therefore, average bit-error-rate = 5 bps /50 bps = 0.1

2.5.6 Coherence Bandwidth

The coherence bandwidth is a statistical measure of the range of frequencies over which the channel can be considered flat. A flat channel is one which passes all spectral components with approximately equal gain and linear phase and without any distortion. The coherence bandwidth B_c represents the correlation between two fading signal envelopes at frequencies f_1 and f_2 and is a function of the delay spread τ_d.

When the correlation coefficient between two fading signal envelopes at frequencies f_1 and f_2 is equal to 0.5, the coherence bandwidth B_c is approximated by:

$$B_c \approx 1 / (2\, \pi\, \tau_d) \tag{2.15}$$

where τ_d is the delay spread.

Two frequencies that are larger than the coherence bandwidth fade independently. This concept is also useful for diversity reception, wherein multiple copies of the same signal are received.

The coherence bandwidth for two fading amplitudes of two received signals is given as

$$\Delta f = |\, f_1 \quad f_2\, | > B_c = 1 / (2\, \pi\, \tau_d) \tag{2.16}$$

The coherence bandwidth for two random phases of two received signals is given as

$$\Delta f = |\, f_1 \quad f_2\, | > E\ B_c\ = 1 / (4\, \pi\, \tau_d) \tag{2.17}$$

where $E\ B_c$ is the average value of the coherence bandwidth B_c.

If the bandwidth of the transmitted signal is larger than the channel coherence bandwidth, a part of the transmitted signal is truncated, which means nonlinearity is present and the signal could be severely influenced by frequency-selective fading. If the bandwidth of the transmitted signal is smaller than the channel coherence bandwidth, only the gain and phase of the signal are changed, which means nonlinear transformation could not occur.

It is possible to support data rates that are less than the coherence bandwidth of the channel that is approximately $1 / (5\ T_{d(rms)})$.

2.5.7 Coherence Time

Coherence time is the time duration over which two received signals have a strong potential for amplitude correlation. In other words, coherence time T_c is inversely proportional to the Doppler spread. It is used to characterise the time-varying nature of the frequency dispersiveness of the channel in the time domain. If the reciprocal bandwidth of the baseband signal is greater than the coherence time of the channel then the channel will change during the transmission of the baseband signal, thus causing distortion at the receiver.

If the coherence time is defined as the time over which the time correlation function is above 0.5, then the coherence time is approximately given by

$$T_c \approx 0.423 / f_{dm} \tag{2.18}$$

where f_{dm} is the maximum Doppler shift given by V_m / λ_c.

It implies that two signals arriving with a time separation greater than T_c are affected differently by the channel. It is recommended that the symbol rate must exceed $1/ T_c$ in order to avoid distortion in a digital transmission system.

EXAMPLE 2.10 | Symbol rate determination

Consider that a mobile subscriber traveling at a uniform velocity of 96 kmph receive digital data from a wireless communication system operating at 900 MHz carrier frequency. What should be the symbol rate so as to receive distortionless transmission?

Solution

Velocity of the mobile, $V_m = 96$ km/h or 26.67 m/s (given)
Frequency of operation, $f_c = 900$ MHz (given)

Step 1. *To determine wavelength of the signal,* λ_c

Wavelength of signal, $\lambda_c = c/f_c$
Therefore, $\lambda_c = (3 \times 10^8 \text{ m/s}) / (900 \times 10^6 \text{ Hz}) = 0.33$ m

Step 2. *To determine maximum Doppler frequency,* f_{dm}

Maximum Doppler frequency, $f_{dm} = V_m / \lambda_c$
Therefore, $f_{dm} = 26.67 \text{ m/s} / 0.33 \text{ m} = 80.79$ Hz

Step 3. *To determine coherence time,* T_c

Coherence time, $T_c \approx 0.423 / f_{dm}$
$T_c \approx 0.423 / 80.79 \text{ Hz} = 5.23$ ms

Step 4. *To determine the symbol rate*

The symbol rate to receive distortionless transmission is given by $= 1/ T_c$
Hence, the symbol rate = 1 / 5.23 ms = 191 bps

Hence, the symbol rate must exceed 191 bps in order to receive distortionless transmission due to occurrence of frequency dispersion. The channel will not cause distortion due to vehicle mobility. However, distortion still could result from multipath time-delay spread, depending on the channel impulse response.

EXAMPLE 2.11 Correlated fading

What does a small delay spread indicate about the characteristics of a fading channel? If the delay spread is 1 microsecond, will the two different frequencies that are 1 MHz apart, experience correlated fading?

Solution

The delay spread determines to what extent the channel fading at two different frequencies f_1 and f_2 are correlated. A small delay spread indicates that smearing or spreading out effect is less. A small delay spread indicates larger coherence bandwidth and hence correlated fading.

Delay spread, τ_d $= 1\ \mu s$ (given)

Difference in frequency, Δf $= | f_1 \quad f_2 | = 1$ MHz (given)

Step 1. *To determine the coherence bandwidth, B_c*

Coherence bandwidth, $B_c = 1 / (2\ \pi\ \tau_d)$

$$B_c = 1 / (2\ \pi \times 1 \times 10^{\ 6}) = 159.15 \text{ kHz}$$

Step 2. *To determine the relationship between Δf and B_c*

Since $\Delta f = 1$ MHz (given), and $B_c = 159.15$ kHz (as calculated in Step 1)

Therefore, $\Delta f \quad B_c$

Hence, correlated fading will not be experienced.

2.6 SIMULATION OF WIRELESS FADING CHANNELS

From the technical point of view, one of the major distinctions between the wireline and wireless communication lies in the physical properties of wireless channels such as signal propagation losses, multipath, fading, ambient noise, interference, and antenna characteristics. Simulation is an important tool used by engineers to design and implement advanced wireless communication systems that deliver optimum performance. Simulating a wireless communication system involves modeling a mobile radio channel based on mathematical descriptions of the channel. Different transmission media have different properties and are modeled differently. In the performance analysis of a wireless communication system, the ideal Additive White Gaussian Noise (AWGN) channel, with statistically independent Gaussian noise samples corrupting data samples, is the usual starting point for developing basic performance result.

Mobile radio channels introduce noise, fading, interference, and other distortions into the signals that they transmit. Fading describes the rapid fluctuations of the amplitudes, phases, or multipath delays of a radio signal over a short period of time or travel distance so that large-scale path loss effects may be ignored. Fading is caused by interference between two or more versions of the transmitted signal that arrive at the receiver at slightly different times. These multipath waves combine at the receiver antenna vectorially to produce a resultant signal, which can vary widely in amplitude and phase, depending on the distribution of the signal strength and relative propagation time of the waves and the bandwidth of the transmitted signal.

In designing a mobile communication system, it is required to estimate the effects of multipath fading on the wireless channel. The simplest channel model is the Additive White

Facts to Know!

Even when a mobile receiver is stationary, the received signal may fade due to movement of surrounding objects in the radio channel.

Facts to Know!

Multipath fading is described by its envelope fading (non frequency-selective amplitude distribution), Doppler's spread, and time-delay spread. These signals cause frequency selective fades.

Facts to Know!

In practical systems, it is not possible to produce a perfect impulse to serve as input for testing. Therefore, an extremely small pulse is sometimes used as an approximation of an impulse. So far as the pulse is short enough compared to the impulse response, the result will be close to the theoretical impulse response.

Gaussian Noise (AWGN) channel. In this channel, the desired signal is degraded by thermal noise associated with the physical channel itself as well as the electronic hardware involved in the communication link. AWGN implies that it is additive to other signals in the receiver, has a flat spectral density across the frequency range $\infty < f < \infty$, and induces a Gaussian probability distribution at the output of a linear filter. This model, however, is not accurate in the mobile communication systems. Rayleigh and Rician fading channels are useful models of real-world phenomena in wireless communications.

The wireless channels can be characterised by a parameter U, defined as ratio of the power in the dominant path to the power in the scattered path. When $U = 0$ (that is, power in the dominant path is zero), the channel is Rayleigh channel, and when U is equal to infinity (that is, power in the scattered path is zero), the channel is AWGN.

The small-scale variations of a mobile radio signal can be directly related to the impulse response of the mobile radio channel. The impulse response is a wideband channel characterisation and contains all information necessary to simulate any type of radio transmission through the channel.

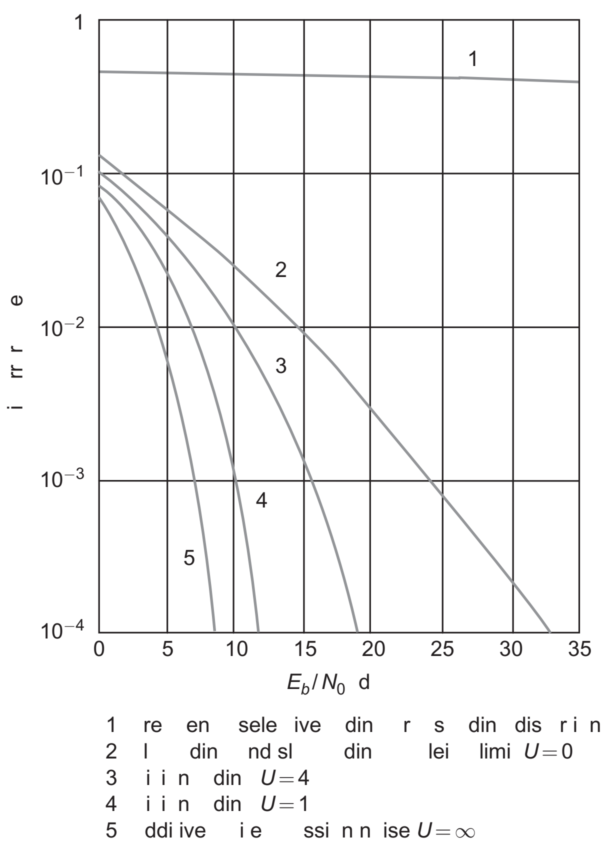

Fig. 2.9 *Theoretical BER for various fading conditions*

The BERTool of Communications ToolBox of technical computing simulation software MATLAB implements a baseband channel model for multipath propagation scenarios that includes the following:

- Local scattering from all angles, with uniform power distribution, around the mobile. This scenario corresponds to the Doppler spectrum. The communications toolbox lets to specify the maximum Doppler shift of the Doppler spectrum.
- N discrete fading paths, each with its own delay and average power gains. A channel having $N = 1$ is called a frequency-flat fading channel. A channel for which N 1 is experienced as a frequency-selective fading channel by a signal of sufficiently wide bandwidth.
- A Rayleigh or Rician model for the first major path. Any subsequent paths use a Rayleigh model.

A mobile radio channel may be modeled as a linear filter with a time-varying impulse response, where the time-variation is due to receiver motion in space. The filtering nature of the channel is caused by the summation of amplitudes and delays of the multiple receiving waves at any instant of time. The Communications Toolbox models a fading channel as a linear Finite Impulse Response (FIR) filter.

Figure 2.9 depicts the theoretical bit error rate-for various fading conditions. These are the simulation results obtained using BERTool. Here, bit-error rate is plotted as a function of the ratio E_b/N_o. Of course, as this ratio increases, the bit-error rate drops. The results show that with a reasonably strong signal level relative to the noise level, an AWGN wireless channel exhibit provides fairly good performance. A Rician fading channel model exhibit provides fairly good performance with larger values of U, which corresponds to an open-area environment or microcells for a digitised voice application.

The Rayleigh fading channel model provides relatively poor performance. This is likely to be seen for slow fading as well as flat fading. Some operating environments produce fading effects worse than the so-called worst case of Rayleigh fading. Examples are fast fading in an urban environment and the fading within the affected frequency band of a selected fading channel. In these cases, no level of E_b/N_o will help in achieving the desired performance, and error-compensation mechanisms are a must. Some examples of error-compensation mechanisms are antenna diversity techniques, forward error-correction methods, and adaptive equalisation schemes.

Rayleigh fading channel model characteristics are often applicable for outdoor urban applications since it takes into consideration Rayleigh fading. It occurs when there are multiple indirect paths between the transmitter and receiver and no distinct line-of-sight path. This represents a worst-case scenario. The Rician fading channel model is applicable in smaller cells or in more open outdoor environments since it takes into consideration Rician fading that occurs when there is a direct line-of-sight path in addition to a number of indirect multiple received signals.

There are special provisions such as path clearance and link budgets which may mitigate the effects of fading. Path clearance ensures that there are no obstructions between the transmitter and receiving antennas. The path clearance calculation is based on the Fresnel zone violations. Propagation delay is the amount of time it takes for a radio signal to travel from its source to the target. Latency is also referred to as delay. Delay spread occurs when multiple signals arrive at different times to the receiver. There are numerous latency mitigation techniques which include reduction in radio range between the transmitter and receiver, reduction in effective radiated power of the interfering cell-site transmitter, redesigning of cell-site antenna systems, adaptive equalisers used in TDMA based cellular systems, and rake receivers used in CDMA based cellular systems.

Key Terms

- Antenna
- Attenuation
- Beamwidth
- Bit error rate (BER)
- Coherence bandwidth
- Coherence time
- Delay spread
- Diffraction
- Direct wave path
- Diversity
- Doppler shift
- Doppler spread
- Fade duration
- Fade rate
- Fading
- Fast fading
- Flat fading
- Free space
- Frequency selective fading
- Line-of-sight (LOS)
- Multipath interference
- Noise
- Obstructive path
- Radio path
- Rayleigh fading
- Reflection
- Refraction
- Rician fading
- Scattering
- Slow fading
- Wireless

Summary

In this chapter, the properties of the wireless media are discussed and the propagation mechanisms, which strongly influence the received signals in a mobile radio environment, are presented. The mechanisms, problems, and related examples discussed here explain the reason as to why wireless communication is fundamentally different from wired communication. Various impairments to propagation such as reflection, diffraction, and scattering have been presented. As the electromagnetic waves travel

from transmitter to mobile receiver, these impairments cause multipath propagation and mobile signal fading which pose serious bottlenecks in the reliable performance of mobile communication systems. The mobile radio channel puts fundamental limitations on the performance of wireless communication systems. Multipath propagation results in delay spread, which causes intersymbol interference limiting the bandwidth of the channel, and irreducible error rates. If fast fading is unchecked, significant degradation in received signal quality due to large-scale bit-error rates occur. If shadow fading is not controlled, it will result in poor signal strength at the receiving end. Therefore, special design techniques such as equalisation and direct-sequence spread-spectrum technology or orthogonal frequency division multiplexing to reduce multipath delay spread, diversity techniques and frequency hopping to control fast fading, and error-control coding are implemented in mobile wireless communication systems to ensure acceptable system performance. The path loss propagation models with an objective of estimating the received signal strength and radio coverage area are discussed in the next chapter.

Important Equations

- ❑ (i) $f_d = (1/\lambda_c)\ V_m \cos\theta$ (2.8)
- ❑ (ii) Average fade duration, $\check{T} = 0.4\ (e^{-\rho^2}\quad 1) / (f_{dm}\ \rho)$ (2.14)
- ❑ (iii) $B_c \approx 1 / (2\ \pi\ Ŧ_d)$ (2.15)

Short-Answer Type Questions with Answers

A2.1 Why is system capacity limited with the wireless medium

The system capacity is limited due to allocation of a limited frequency spectrum for operation in specific service area for a particular application until some special techniques such as frequency-reuse, cell splitting, etc, are deployed in the wireless communication systems.

A2.2 Which is the most critical design parameter for the design analysis and installation of mobile communication systems

The critical design parameters are the achievable signal coverage, the maximum data rate that can be supported by the wireless channel, and the rate of signal fluctuations in the wireless channel. However, the most critical design parameter in modern mobile communication systems is the achievable data rate anytime anywhere using wireless medium.

A2.3 How is propagation path loss related to the received signal power in mobile communications

The propagation path loss and the received signal power are reciprocal to each other. This implies that if the propagation path loss is more in certain operating conditions, the received signal power is less or vice versa.

A2.4 The propagation path loss increases at 20 dB/decade in free-space condition whereas it increases at 40 dB/decade rate in mobile radio operating environment. Why is it so

We know that the received signal power is inversely proportional to R^{γ}, where R is the distance measured from the transmitter to the receiver, and γ is the path-loss exponent, having typical values of 2 and 4 for free-space and mobile radio operating environment respectively. And the received signal power and propagation path loss are reciprocal to each other, assuming all other factors constant. Thus, propagation path loss is directly proportional to R^{γ}. Hence, ΔL_f (in dB) = 20 log ($10\ R_1 / R_1$) = 20 dB per decade in free-space condition. And ΔL_m (in dB) = 40 log ($10\ R_1 / R_1$) = 40 dB per decade in mobile radio environment.

A2.5 What is the value of propagation path loss in dB at a distance of 50 km from the transmitter if it is given that the path loss is 110 dB at a distance of 25 km Assume the mobile radio environment conditions.
Since the propagation path loss increases at the rate of 12 dB/octave in the mobile radio propagation environment condition, the propagation path loss at a distance of 50 km is 12 dB greater than that of at a distance of 25 km. Hence, the value of propagation path loss at a 50-km distance is 110 + 12 = 122 dB, assuming all other factors remaining constant.

A2.6 State the reasons for attenuations of the signal strength of electromagnetic waves upon reflection.
The signal strength of the radio waves gets attenuated that depend on many factors like the frequency of the radio waves, the angle of incidence, and the nature of the medium including its material properties, thickness, homogeneity, etc. Generally, higher frequencies reflect more than lower frequencies.

A2.7 In what ways does the received signal get affected by multipath propagation in mobile communications
The received signal gets affected in many ways by multipath propagation such as

- reception of multiple copies of a signal at different phases; if these phases add destructively, the signal level relative to noise declines, making signal detection at the receiver much more difficult and unreliable
- increase in received data errors due to intersymbol interference in digital transmission
- blocking or shadowing of radio signals, which is caused by obstacles much larger in size than the wavelengths of the operating signals such as a small wall, trees, or a large vehicle.

A2.8 List the factors that influence small-scale fading.
The various factors that influence small-scale fading include multipath propagation, speed of the mobile, speed of the surrounding objects, multipath fading, Doppler effect, and the transmission bandwidth of the signal.

A2.9 Classify the fading effects due to Doppler spread.
Due to Doppler spread, fading effects can also be classified as fast fading or slow fading. In fast fading conditions, rapid fluctuations in received signal strength occur over distances of about one-half a wavelength as the mobile subscriber moves in an urban environment. In slow fading conditions, there is a change in the average received power level about which the rapid fluctuations occur as the mobile subscriber covers distances well in excess of a wavelength due to changes in the urban environment over these longer distances.

A2.10 What are different techniques used to minimise the effects of fading
Error control coding with interleaving, diversity schemes, using directional antennas at the cell-site, and frequency hopping are some of techniques widely used in mobile communication systems to minimise the effects of fading.

A2.11 Define the terms fade rate average fade duration depth of fading.
Fade rate is defined as the number of times that the received signal envelope crosses the threshold value in a positive-going direction per unit time.
Average fade duration is defined as the average period of time for which the received signal is below a specified received signal level.
Depth of fading is defined as the ratio between the mean square value and the minimum value of the fading signal.

A2.12 What is meant by coherence time Mention its significance.
Coherence time is the time duration over which two received signals have a strong potential for amplitude correlation. In other words, coherence time is inversely proportional to Doppler spread. It is used to characterise the time-varying nature of the frequency dispersiveness of the channel in the time domain.

A2.13 List some-error compensation mechanisms deployed in wireless communication systems.
Antenna diversity techniques, forward error-correction methods, and adaptive equalisation schemes are some

of the error-compensation mechanisms deployed in wireless communication systems.

A2.14 What are the distinct applications of Rayleigh and Rician fading channel model characteristics in wireless communications
Rayleigh fading channel model characteristics is often applicable for outdoor urban applications when there are multiple indirect paths between the transmitter and receiver and no distinct line-of-sight path. Rician fading channel model characteristics is applicable in smaller cells or in more open outdoor environments when there is a direct line-of-sight path in addition to a number of indirect multiple received signals.

A2.15 What is latency Mention some latency mitigation techniques widely used in different cellular systems.
Latency is referred to as delay. Delay spread occurs when multiple signals arrive at different times to the receiver. There are numerous latency mitigation techniques which include reduction in radio range between the transmitter and receiver, reduction in effective radiated power of the interfering cell-site transmitter, redesigning of cell-site antenna systems, adaptive equalisers used in TDMA-based cellular systems, and rake receivers used in CDMA based cellular systems.

Self-Test Quiz

S2.1 The wireless medium
(a) is quite reliable for voice and data communication
(b) is not quite reliable for voice and data communication
(c) offers very large bandwidth
(d) does not support mobility

S2.2 It is difficult to achieve accurate timing, synchronisation and phase recovery at the mobile receiver. It is attributed mainly to the effect of
(a) Doppler spread of the received signal
(b) multipath propagation reception
(c) scattering of the transmitted signal
(d) non line-of-sight propagation

S2.3 The propagation path loss
(a) increases with frequency of transmission but decreases with the distance
(b) decreases with frequency of transmission as well as the distance
(c) increases with frequency of transmission as well as the distance
(d) is always constant, independent of frequency of transmission and distance

S2.4 The difference in free-space propagation path loss between two locations at 2 km and 8 km from the transmitter is
(a) 6 dB
(b) 12 dB
(c) 20 dB
(d) 40 dB

S2.5 In mobile radio propagation environment, the typical value of the path loss-exponent, γ is
(a) 2
(b) 3
(c) 4
(d) 5

S2.6 occurs when the radio path between a transmitter and receiver is obstructed by a surface with sharp irregular edges.
(a) Scattering
(b) Refraction
(c) Reflection
(d) Diffraction

S2.7 Two main reasons that contribute to the rapid fluctuations of the signal amplitude in mobile communications are
(a) multipath fading and Doppler effect
(b) reflection and refraction
(c) diffraction and scattering
(d) blocking and shadowing

S2.8 In a digital communication system, the delay spread along with fading causes , thereby limiting the maximum symbol data rate.
(a) intersymbol interference
(b) multipath fading

(c) Doppler effect
(d) high bit-error rates

S2.9 The average delay spread is typically about ______ in an urban area.
(a) $< 0.1\ \mu s$
(b) $0.5\ \mu s$
(c) $3\ \mu s$
(d) $10\ \mu s$

S2.10 Doppler frequency or Doppler shift is given by
(a) $\lambda_c V_m \cos\theta$
(b) $(1/\lambda_c) V_m \cos\theta$
(c) $1/(\lambda_c V_m) \cos\theta$
(d) $(1/V_m) \lambda_c \cos\theta$
where λ_c is the wavelength of the carrier signal, V_m is the relative velocity of the mobile, the angle θ is between the motion of the mobile and direction of arrival of the scattered waves.

S2.11 A base-station transmitter operates at a 900 MHz carrier frequency. For a mobile moving at a speed of 72 km/h in a direction perpendicular to the direction of arrival of the transmitted signal, the received carrier frequency is
(a) 899.99994 MHz
(b) 900.00006 MHz
(c) 900.00003 MHz
(d) 900 MHz

S2.12 A ______ is the one which passes all spectral components with approximately equal gain and linear phase and without any distortion.
(a) Rayleigh fading channel
(b) Rician fading channel
(c) frequency-selective channel
(d) flat channel

S2.13 If the bandwidth of the transmitted signal is larger than the channel coherence bandwidth, then the signal could be severely influenced by
(a) frequency-selective fading
(b) flat fading
(c) fast fading
(d) slow fading

S2.14 ______ channels are useful models of real-world phenomena in wireless communications.
(a) Rayleigh and Rician fading
(b) Frequency-selective fading
(c) AWGN
(d) Fast fading

S2.15 As the E_b/N_o ratio increases, the bit-error rate ______.
(a) increases
(b) decreases
(c) remains same
(d) approaches infinity

S2.16 ______ refers to the phenomenon by which multiple copies of a transmitted signal are received at the receiver, due to the presence of multiple radio paths.
(a) Rayleigh fading
(b) Rician fading
(c) Multipath
(d) Reflection

S2.17 ______ results from the presence of objects between the transmitter and the receiver.
(a) Scattering
(b) Refraction
(c) Shadow fading
(d) Doppler effect

Answers to Self-Test Quiz

S2.1 (b); S2.2 (a); S2.3 (c); S2.4 (b); S2.5 (c); S2.6 (d); S2.7 (a); S2.8 (a); S2.9 (c); S2.10 (b); S2.11 (d); S2.12 (d); S2.13 (a); S2.14 (a); S2.15 (b); S2.16 (c); S2.17 (c)

Review Questions

Q2.1 Why is the mobile radio environment unique as compared to other forms of wireless communication and what are the major difficulties experienced in providing mobile radio service?

Q2.2 Identify at least three factors that can affect signal propagation in a wireless medium. Explain how each factor can occur and/or how it impacts propagation.

Q2.3 Describe at least two problems that can occur as a result of multipath interference.

Q2.4 List three types of fading. Explain how each one typically occurs.

Q2.5 What does a small delay spread indicate about the characteristics of the fading channel?

Q2.6 Identify at least three effects of fading that could be noticed by the listener.

Q2.7 What are the causes of fast and slow fading? Distinguish between them.

Q2.8 State two undesirable effects that can be caused by reflections in line-of-sight communication and explain how they arise.

Q2.9 What is meant by coherence bandwidth? Give its expression.

Q2.10 Distinguish between AWGN channel, Rayleigh fading channel and Rician fading channel in a wireless environment.

Analytical Problems

P2.1 Find the average fade duration for threshold levels of $\rho = 1$, 0.1, and 0.01, when the maximum Doppler frequency is 200 Hz.

P2.2 Determine the maximum and minimum spectral frequencies received from a stationary GSM cell-site transmitter that has a centre frequency of exactly 900 MHz, assuming that the receiver is traveling at a uniform velocity of

(a) 1 km/h (b) 5 km/h (c) 100 km/h

P2.3 A mobile receives a 900-MHz transmission while traveling at a constant velocity for 10 s. The average fade duration for a signal level 10 dB below the rms level is 1 ms.

(a) How far does the vehicle travel during the 10-s interval?
(b) How many fades does the signal undergo at the rms threshold level during a 10-s interval? Assume that the local mean remains constant during travel.

P2.4 For a mobile receiver operating at a carrier frequency of 850 MHz and moving at a constant velocity of 100 km/h, compute the level-crossing rate and average fade duration if $\rho =$ 20 dB.

P2.5 The Doppler spectrum of the wireless system deployed for indoor radio applications operating at 1.9 GHz is often assumed to have a uniform distribution with a maximum Doppler shift of 10 Hz. The threshold for fading is chosen to be 10 dB below the average rms value of the signal. Determine

(a) the average number of fades per second
(b) the average fade duration

P2.6 A flat Rayleigh fading signal at 1 GHz is received by a mobile traveling at a velocity of 80 km/h.

(a) Determine the number of positive-going zero crossings about the rms value that occurs over a time interval of 5 seconds.
(b) Determine the average fade duration just below the rms value.
(c) Determine the average fade duration at a level of 20 dB below the rms value.

P2.7 How rapidly would the signal fade if a mobile operating at 800 MHz is in a vehicle moving at 100 km/h? If the vehicle is moving directly across a cell of 3-km radius, how long will it remain in that cell before it has to be handed off to the next cell?

P2.8 A PCS signal at 1990 MHz arrives at a mobile receiver antenna via two reflected waves differing in path lengths by 19 m.

(a) Calculate the difference in arrival time for the two waves.
(b) Compute the phase difference between the two reflected signals.

P2.9 Compute the level crossing rate with respect to the rms level for a vertical monopole antenna assuming Rayleigh fading. The speed of the mobile is 20 km/h, and the transmission occurs at 800 MHz.

P2.10 Compute the Doppler frequency for the Rayleigh channel if the average fade duration is 400 μs and the threshold level $\rho = 0.01$.

P2.11 A mobile receiver is moving at a uniform speed of 80 km/h, and is receiving the signals at 1 GHz frequency. It encounters Doppler frequencies ranging from 10 Hz to 50 Hz. What is the beam width of the mobile antenna?

P2.12 Determine the coherence bandwidth corresponding to a delay spread of 1 microsecond.

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3

The Propagation Models

Wireless transmission is characterised by the generation of an electromagnetic signal representing the desired information by the transmitter and antenna, the propagation of radio waves through space, and an antenna and receiver that estimates the information from the recovered electrical signal. The core of the signal coverage estimation for any operating environment is a propagation path-loss model that defines the relationship between the loss of transmitted signal strength with the distance between the transmitter and receiver. Propagation models are used extensively for conducting feasibility studies during initial deployment, performing interference studies as the deployment proceeds, and network planning of wireless communication systems. This chapter gives an overview of the propagation criteria for a signal in wireless medium, propagation models including free space, mobile point-to-point, outdoor and indoor propagation models.

3.1 PROPAGATION CRITERIA

In wireless communication, the radio signal propagates in space. The detailed study and analysis of electromagnetic wave propagation characteristics is an important aspect before design and implementation of wireless mobile communication systems. A wired line medium such as twisted-pair copper wires or RF coaxial cable or even an optical fiber provides reliable communication as the signal is well confined within it. In a wired medium, the behaviour of a signal in terms of the received signal power depending on the distance traveling along the wire can be precisely determined since it typically exhibits the linear characteristics. A wired medium limits the maximum frequency component it can transmit, thus limiting the bandwidth of operation, and can be considered as a filter. Moreover, the cost and problem of laying a wired medium and then maintaining it increases. On the other hand, the communication through wireless medium is quite unreliable and unsecured. Also, a limited operating bandwidth is available for communication application because there are many other existing applications of radio wave propagation in space.

Every mobile radio communication system is designed with an objective of providing continuous communication to mobile subscribers that occupy arbitrary locations in the service area. The design can be optimised only if the constraints under which the system has to operate are fully understood, the most important being the accurate prediction of path loss and estimation of received signal strength in the radio coverage area. The proper analysis of radio propagation is quite useful in locating the cell-site and determining its radio-signal coverage as well as achievable data rate. Radio-signal coverage is the service area supported by each cell-site within which the service quality requirements such as the minimum acceptable received signal power for a given transmitted power or the required carrier signal-to-interference (C/I) ratio are acceptable. It also depends on the propagation environments, that is, the environments in which the service area falls, for example, natural terrains (flat, hilly, water, or foliage) and human-made structures (open, suburban, or urban areas). Determination of signal coverage area is essential for design and deployment of both narrowband and wideband mobile communication systems. Signal coverage is influenced by a number of factors that includes the most prominent being the radio frequency of operation and the nature of terrain.

Facts to Know!

It is the wireless medium alone which is capable of providing mobility among the potential users in mobile radio communication systems.

Facts to Know!

Physical propagation models take into account the exact physics of the propagation environment. Statistical propagation models take in empirical approach, measuring propagation characteristics in a variety of environment and then developing a model based on the measured data for a particular type of environment. Empirical models are easier to describe and use than the physical models, but do not provide the same accuracy.

The most important design parameter in mobile communication systems is the achievable signal coverage as the primary objective of the systems is to serve the mobile subscribers in a specified service area. For a given transmitter power, the achievable signal coverage determines the range of operation of a cell-site transmitter and the size of a cell in a cellular topology. This is usually obtained using empirical path-loss models based on measuring the received signal strength as a function of distance while keeping other system parameters constant. Propagation path loss is the attenuation of the field intensity of a radio signal due to all factors influencing it along its radio path. The path loss is caused by many factors such as multi-path propagation, reflection, refraction, diffraction, absorption, and so on. In practice, the path loss, that is, the signal loss from transmitted power to received signal level through the wireless medium depends on the propagation environment including the transmitter and receiver antenna heights. A *path-loss exponent* or *distance-power* or *path-loss gradient*, and a random component that signifies the fluctuations around the average path loss due to shadow fading effects and other similar reasons characterise most of the path-loss models.

Radio-coverage prediction models which are basically point-to-point signal propagation models can predict signal coverage. The service area computed from the prediction models may vary depending on the type of the operating geographical area used such as open area, suburban area, or urban area; terrain used such as flat terrain, hilly terrain, foliage areas or over water. For example, for free space transmission with an omnidirectional antenna installed at the cell-site, the cell coverage is a circle centered at the cell-site transmitter antenna with a radius depending on the propagation loss.

Thus, accurate characterisation of the radio channel through key system parameters along with a mathematical propagation model is important for predicting signal coverage, analysis of interference from different sources, and determining the optimum location for installing cell-site antennas in cellular systems. Radio propagation is quite different in open areas from radio propagation in urban, suburban and indoor areas. In the mobile radio environment, the received signal strength often attenuates at a much higher rate as a function of distance. Propagation path-loss models have traditionally focused on predicting the average received signal strength at a given distance from the transmitter (termed as large-scale propagation models), as well as the rapid fluctuations of the received signal strength in close proximity to a particular receiver location (termed as small-scale multipath or fading propagation models). Figure 3.1 depicts the simplest form of large-scale radio propagation in a mobile communication system.

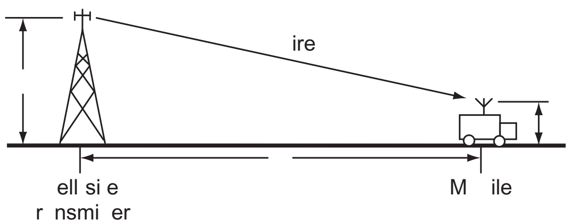

Fig. 3.1 *Radio propagation in a mobile communication system*

The simplest method of relating the received signal power to the distance is to state that the received signal power P_r is proportional to the distance between the transmitter and receiver 'r', raised to an exponent γ, referred to as path-loss exponent or distance-power gradient, that is,

$$P_r = P_0\, r^{-\gamma} \tag{3.1}$$

where P_r is the received signal strength at 'r' metres from the cell site transmitter, and P_0 is the received signal strength at a reference distance (usually one metre in indoor environments and 100 m or 1 km in outdoor environments depending on the cell size in practical systems using low-gain antennas). The path-loss exponent, γ is valuable since it shows the rate of increase of path loss with respect to distance.
Converting it into decibel form, the expression can be written as

$$10 \log (P_r) = 10 \log (P_0) - 10\, \gamma \log (r) \tag{3.2}$$

The term $10\, \gamma \log (r)$ represents the power loss in dB with respect to the received power at reference distance, say one metre.
The path loss at a distance of one metre is defined as

$$L_0\, (\text{dB}) = 10 \log (P_t) - 10 \log (P_0) \tag{3.3}$$

Then the total path loss L_p at a given distance 'r' is given by

$$L_p\, (\text{dB}) = L_0\, (\text{dB}) + 10\, \gamma \log (r) \tag{3.4}$$

This presents the total path-loss in the first metre plus the path-loss relative to the power received at one metre. In the simplest form, the received power in dB is the transmitted power in dB minus the total path loss.

Thus, path loss can be defined as the ratio of the transmitted to received power usually expressed in decibels.

A detailed understanding of radio propagation models is essential for appropriate design, deployment, and optimised performance in any wireless communication system. Signal propagation through a wireless media can vary significantly depending on the operating terrain, frequency of transmission, sources of interference, speed of the mobile unit, and many other dynamic factors, and so it is heavily location-specific. As the path loss encountered along any radio link serves as the dominant factor for characterisation of signal propagation, radio-propagation models typically focus on realisation of the path loss with the objective of predicting the radio coverage for a base station transmitter or modeling the distribution of signals over different types of regions.

Facts to Know!

The propagation models rely on computing the median path loss for a wireless link under a certain probability that the considered conditions will occur.

Each individual wireless communication link generally encounters different paths, terrain, obstructions, atmospheric conditions and other phenomena. So it is intractable to formulate the exact propagation path loss for all types of wireless communication systems in a single mathematical equation. As a result, different models exist for different types of radio links under different conditions.

3.2 FREE-SPACE PROPAGATION MODEL

Free-space propagation model is the fundamental for all propagation path-loss models for any wireless communication application. A free-space propagation model is used to predict the received signal strength when the transmitter and receiver has a clear unobstructed line-of-sight signal path between them. The free space environment could be categorised as one having less clutter and foliage. The propagation path-loss models are extensively used for deployment of cellular communication networks. The coverage of a cell-site depends on the power of the transmitted signal and the path loss. Each mobile unit has a specified receiver sensitivity, for example, it can only detect and decode signals with signal strength larger than its sensitivity level. Because the signal strength decays with distance, the radio coverage can be estimated using the transmitter power, the path-loss model and the receiver sensitivity.

In most operating environments, it is observed that the radio signal strength decays as some power γ of the distance, called the *power-distance gradient* or *path-loss exponent*. That is, if the transmitted power is P_t, the signal strength at a distance 'r' in metres will be proportional to $P_t\, r^{-\gamma}$. In free space propagation, there is no loss of energy as a radio wave propagates in free space, but there is attenuation due to the spreading of the electromagnetic waves.

Facts to Know!

Sensitivity is a receiver parameter that indicates the minimum signal level required at the input of receiver unit (at the output of receiver antenna terminal) in order to provide reliable communications. It is often expressed in dBm, that is, the power in mW expressed in decibels.

Since an isotropic radiator radiates equally in all directions, the power density is simply the transmitted power divided by the surface area of the sphere. Expressing it mathematically,

$$P_D = P_t / (4\,\pi\, r^2) \tag{3.5}$$

where P_D is the power density in watts/m^2, P_t is the transmitter power in watts, r is the distance from the transmitting antenna in metres, and $4\,\pi\, r^2$ is surface area of the sphere.

Facts to Know!

If a sphere were drawn at any distance from the transmitting source and concentric with it, all the signal energy from the source would pass through the surface of the sphere.

It is emphasised here that the attenuation in the signal is not due to any loss of energy in the free space, but only due to the spreading out of the energy as it moves away from the radiating source.

Practical antennas do not radiate equally in all directions as that of in case of isotropic antennas and, therefore, the transmitting antenna has gain, G_t. In fact, antennas are passive devices and do not have actual power gain. They achieve a greater power density in certain directions at the expense of reduced power density in other directions.

So Eq. (3.5) can be modified to include the transmitting antenna gain, G_t

That is,
$$P_D = (P_t\, G_t) / (4\,\pi\, r^2) \tag{3.6}$$

Usually, antenna gain is specified in dBi, where the 'i' indicates gain with respect to an isotropic radiator. The transmitting antenna gain G_t must be converted to a power ratio to be used in Eq. (3.6).

The effective isotropic radiated power (EIRP) of a transmitting system in a given direction is defined as the transmitter power that would be needed with an isotropic radiator, to produce the same power density in the given direction. That means,

$$\text{EIRP} = P_t\, G_t \tag{3.7}$$

Equation (3.6) can be modified for use with EIRP as

$$P_D = (\text{EIRP}) / (4\,\pi\, r^2) \tag{3.8}$$

EXAMPLE 3.1 | EIRP and power-density calculations

A wireless communication transmitter having RF power of 113 W is used with an antenna of 5 dBi gain. Calculate the EIRP and the power density at a distance of 11 km.

Solution

Step 1. *To convert antenna gain of 5 dBi to a power ratio*

$G_t = 10^{(5/10)} = 3.16$

Step 2. *To calculate the EIRP*

$\text{EIRP} = P_t G_t$

$\text{EIRP} = 113 \times 3.16 = 357.1 \text{ W}$

Step 3. *To calculate the power density*

$P_D = \text{EIRP} / (4 \pi r^2)$

$P_D = 357.1\text{W} / \; 4\pi (11 \times 10^3 \text{m})^2 \; = 235 \text{ nW/m}^2$

A receiving antenna absorbs some of the signal energy from radio waves that pass through it. Since the signal energy in the radio wave is directly proportional to the area through which it passes, a receiving antenna having large area will intercept more energy than a smaller one. Receiving antennas are also more efficient at absorbing signal power from some directions than from other directions. That simply means receiver antennas too have gain. In fact, the antenna gain is same whether the antenna is used for transmitting or receiving RF signals.

The power extracted from the radio wave by a receiving antenna depends on its physical size as well as its gain. The effective area of a receiving antenna can be defined as

$$A_{eff} = P_r / P_D \tag{3.9}$$

where A_{eff} = effective area of the receiving antenna in m^2

P_r = power delivered by a receiving antenna to the receiver in watts

P_D = power density of the radio wave in watts/m^2

It implies that the effective area of a receiving antenna is the area from which all the power in the incident radio wave is extracted and delivered to the receiver unit.

Rewriting Eq. (3.9),

$$P_r = P_D A_{eff}$$

Substituting P_D from Eq. (3.6), we get

$$P_r = \; (P_t G_t) / (4 \pi r^2) \;\; A_{eff} \tag{3.10}$$

The effective area of a receiving antenna depends on its gain, G_r as well as the wavelength of the incident radio wave λ_c, and is given as

$$A_{eff} = G_r \lambda_c^{\,2} / (4 \pi) \tag{3.11}$$

Substituting A_{eff} into Eq. (3.10), the received power P_r in free space is given by

$$P_r = \; (P_t G_t) / (4 \pi r^2) \;\; G_r \lambda_c^{\,2} / (4 \pi)$$

Or,

$${}_r = \;{}_t\;\;{}_t\;\;{}_r\; \boldsymbol{\lambda_c} / (4 \boldsymbol{\pi} \boldsymbol{r})^2 \tag{3.12}$$

where 'r' is the distance between the transmitter and receiver and is of same units as λ_c.

This equation is known as *riis' free-space equation* and is used to estimate the signal power received by a receiver antenna separated from a transmitting antenna by a distance 'r' in the free space, neglecting the system losses.

Free-space propagation path loss, L_{pf}, is defined as

$$L_{pf} = P_t / P_r = \; 1 / (G_t G_r) \;\; (4 \pi r / \lambda_c)^2$$

When $G_t = G_r = 1$ (unity gain), then free-space path loss is given by

$$_{f} = (4\,\pi\,\boldsymbol{r} / \boldsymbol{\lambda_c})^2 \tag{3.13}$$

Or, $$L_{pf} = (4\,\pi\,r\,f_c / 3 \times 10^8)^2 \quad \text{(where } r \text{ is in metres)}$$

Or, $$L_{pf}\,(\text{dB}) = 20 \log (4\,\pi / 3 \times 10^8) + 20 \log r\,(\text{m}) + 20 \log f_c\,(\text{Hz})$$

Or, $$L_{pf}\,(\text{dB}) = -147.56 + 20 \log r\,(\text{m}) + 20 \log f_c\,(\text{Hz})$$

Or, $$_{f}\,(\text{dB}) = 32.44 + 20 \log \boldsymbol{r}\,(\text{km}) + 20 \log \boldsymbol{f_c}\,(\text{MHz}) \tag{3.14}$$

This is called free-space path loss equation.

Figure 3.2 illustrates the plot between $_{f}$(dB) and distance $\boldsymbol{r}$' at specified frequencies $\boldsymbol{f_c}$, based on free space path loss equation.

As evident from Eq. (3.14) also, the plot of free space propagation path loss versus distance indicates that as the frequency increases, the free space path loss also increases at the same distance from the transmitter. Free-space path loss increases with frequency of transmission because the effective area of a receiving antenna with a given gain decreases with frequency. However, for an ideal isotropic radiator and a receiving antenna of constant effective area, there is no dependence of free-space attenuation on frequency. In the free-space path-loss expression, the square-law attenuation in the signal due to distance is represented by its logarithmic equivalent, that is, 20 log r (km). There is no term for antenna heights of cell-site or mobile unit, since this is irrelevant in free space.

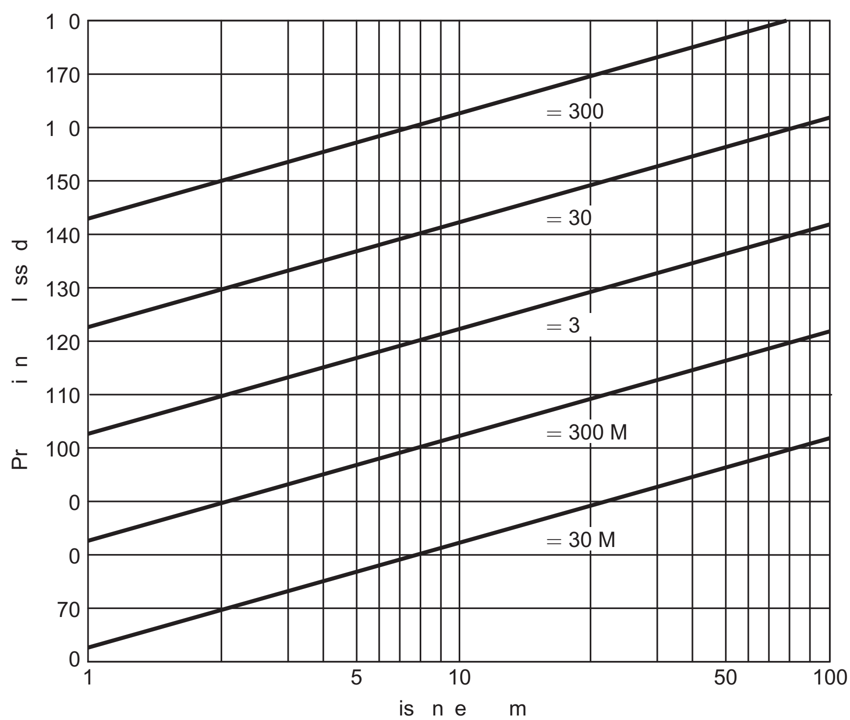

Fig. 3.2 *Plot of free-space propagation path-loss vs distance*

EXAMPLE 3.2 Free-space path loss

Calculate the free-space path loss for a signal transmitted at a frequency of 900 MHz, the distance between the transmitter and receiver being 1 km.

Solution

The frequency of transmission, $f_c = 900$ MHz or 900×10^6 Hz (given)

The distance between *T*x and *R*x, $r = 1$ km or 1000 m (given)

Step 1. *To find the wavelength,* λ_c

We know that $\lambda_c = c / f_c$

Therefore, $\lambda_c = (3 \times 10^8 \text{ m/s}) / (900 \times 10^6 \text{ Hz}) = 0.33$ m

Step 2. *To calculate the free-space path loss,* L_{pf}

We know that free-space path loss, $L_{pf} = (4\,\pi\, r / \lambda_c)^2$

Therefore, $L_{pf} = (4\,\pi \times 1000 / 0.33)^2$

Or, $L_{pf}\,(\text{dB}) = 10 \log (4\,\pi \times 1000 / 0.33)^2 = 20 \log (4\,\pi \times 1000 / 0.33)$

Hence, the free-space path loss $_f\,(\text{dB}) = 91.6$ dB

Using Eq. (3.13) in Eq. (3.12), we get

$$_r = \;_t\;_t\;_r / \;_f \tag{3.15}$$

Or, $$\text{r (dBm)} = \;_t\,(\text{dBm}) + \;_t\,(\text{dBi}) + \;_r\,(\text{dBi}) - \;_f\,(\text{dB}) \tag{3.16}$$

Attenuation due to transmission-line losses or mismatch at the transmitter and receiver, if any, should be included in this equation to obtain the actual received power.

At the first metre ($r = 1$ m), let $P_r = P_0$; then from Eq. (3.12)

$$P_0 = P_t\, G_t\, G_r\, (\lambda_c / 4\,\pi)^2 \tag{3.17}$$

Then from Eq. (3.12) and Eq. (3.17),

$$P_r = P_0 / r^2 \tag{3.18}$$

In dB, this equation takes the form

$$10 \log (P_r) = 10 \log (P_0) - 20 \log (r) \tag{3.19}$$

This means that there is a 20-dB per decade or 6-dB per octave loss in signal strength as a function of distance in free space.

EXAMPLE 3.3 Range of a base station transmitter

Determine the radio coverage range of a base station that transmits a RF signal at 100 W, given that the receiver threshold level is –100 dBm. Assume that the path loss at the first metre is 30 dB in a mobile radio propagation condition ($\gamma = 4$).

Solution

Step 1. *To convert transmitter power in dBm units.*

Transmitter power, $P_t = 100$ W or 100,000 mW (given)

We know that $P_t\,(\text{dBm}) = 10 \log\; P_t\,(\text{mW})$

Therefore, $P_t\,(\text{dBm}) = 10 \log\; P_t\,(100{,}000) = +50$ dBm

Step 2. *To find path loss,* L_p *(dB)*

Receiver threshold level, $P_r\,(\text{dBm}) = -100$ dBm (given)

We know that path loss, $L_p\,(\text{dB}) = P_t\,(\text{dBm}) - P_r\,(\text{dBm})$

Or, path loss, $L_p\,(\text{dB}) = +50 \text{ dBm} - (-100 \text{ dBm}) = 150$ dB

Step 3. *To determine the radio coverage range, r*

Path loss at first metre, $L_0 = 30$ dB (given)

Propagation path constant, $\gamma = 4$ (given)

We know that path loss, L_p (dB) $= L_0$ (dB) $+ 10 \log (r)^\gamma$

where r is the radio coverage range of the base station transmitter in metres.

Therefore, L_p (dB) $= L_0$ (dB) $+ 10 \log (r)^4$

Or, L_p (dB) $= L_0$ (dB) $+ 40 \log (r)$

Or, $40 \log (r) = L_p$ (dB) $- L_0$ (dB)

Or, $40 \log (r) = 150 - 30 = 120$ dB

Hence, $r = 10^{(120/40)} = 10^3 = 1000$ metres or 1 km

Hence, the radio coverage range $\boldsymbol{r} = 1$ km

Facts to Know!

Converting dBm to mW does not change the relative measurement characteristics of decibel. It simply means that there is a reference point for the values used in the calculation; any dB values can be added or subtracted as if they are represented in the same units.

EXAMPLE 3.4 **Received power in free-space propagation**

A wireless communication transmitter has an output power of 165 watts at a carrier frequency of 325 MHz. It is connected to an antenna with a gain of 12 dBi. The receiving antenna is 15 km away and has a gain of 6 dBi. Calculate the power delivered to the receiver, considering free-space propagation. Assume that there are no other losses or mismatches in the system.

Solution: Refer Fig. 3.3.

Transmitter output power,

$P_t = 165$W or 165,000 mW (given)

Transmitter antenna gain, $G_t = 12$ dBi (given)

Receiver antenna gain, $G_r = 6$ dBi (given)

Carrier frequency, $f_c = 325$ MHz (given)

Distance between transmitter and receiver,

$r = 15$ km (given)

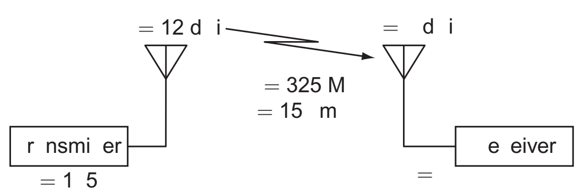

Fig. 3.3 *Illustration of communication link for Example 3.4*

Step 1. *To convert P_t(W) in P_t(dBm)*

We know that P_t(dBm) $= 10 \log P_t$(mW)

Therefore, P_t(dBm) $= 10 \log 165{,}000 = +52.17$ dBm

Step 2. *To find free-space path loss, L_{pf} (dB)*

L_{pf} (dB) $= 32.44 + 20 \log r$ (km) $+ 20 \log f_c$ (MHz)

L_{pf} (dB) $= 32.44 + 20 \log 15$ (km) $+ 20 \log 325$ (MHz)

L_{pf} (dB) $= 32.44 + 23.52 + 50.24 = 106.20$ dB

Step 3. *To calculate power delivered to the receiver, P_r (dBm)*

P_r(dBm) $= P_t$ (dBm) $+ G_t$ (dBi) $+ G_r$ (dBi) $- L_{pf}$(dB)

P_r(dBm) $= 52.17 + 12 + 6 - 106.20 =$ 36.03 dBm

Step 4. *To express P_r(dBm) in P_r(watts)*

We know P_r (dBm) $= 10 \log P_r$ (mW)

Therefore, $-36.03 = 10 \log P_r$ (mW)

Or, P_r (mW) $= 10^{(-36.03/10)} = 249.46 \times 10^{-6}$ mW

Hence, P_r (W) $= 249.46 \times 10^{-9}$ watts

The transmission delay or time taken by a wave from transmitter to a point 'r' metre away from it, Ί can be given by

Ί $= r / c$ where c is the speed of electromagnetic waves in space

Or, Ί $= r(\text{m}) / (3 \times 10^8 \text{ m/s})$

Ί $\approx 3.33\, r$ ns or 3.33 ns per metre of distance (3.20)

EXAMPLE 3.5 **Path loss, received power and transmission delay**

A wireless communication base station transmits 10 watts of power at a carrier frequency of 1 GHz. If the receiver station is at a distance of 1.6 km from the base station, then determine
(a) the propagation path loss (in dB) in a free space environment
(b) the received signal power (in dBm)
(c) the transmission delay in ns
Assume that the transmitter and receiver antenna gains are 1.6 each.

Solution: Refer Fig. 3.4.

Transmitter output power, $P_t = 10\text{W or } 10{,}000 \text{ mW}$ (given)
Transmitter antenna gain, $G_t = 1.6$ (given)
Receiver antenna gain, $G_r = 1.6$ (given)
Carrier frequency, $f_c = 1 \text{ GHz or } 1000 \text{ MHz}$ (given)
Distance between transmitter and receiver, $r = 1.6$ km (given)

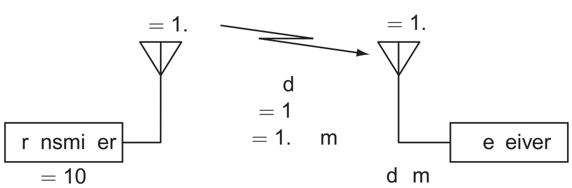

Fig. 3.4 *Illustration of communic ation link for Example 3.5*

Step 1. *To convert P_t(W) in P_t(dBm)*

We know that $P_t(\text{dBm}) = 10 \log P_t(\text{mW})$
Therefore, $P_t(\text{dBm}) = 10 \log 10{,}000$
$= +40 \text{ dBm}$

Step 2. *To convert G_t(ratio) in G_t(dB)*

We know that $G_t(\text{dB}) = 10 \log G_t(\text{ratio})$
Therefore, $G_t(\text{dB}) = 10 \log 1.6 = 2 \text{ dB}$

Step 3. *To convert G_r(ratio) in G_r(dB)*

We know that $G_r(\text{dB}) = 10 \log G_r(\text{ratio})$
Therefore, $G_r(\text{dB}) = 10 \log 1.6 = 2 \text{ dB}$

(a) *To determine free-space path loss, L_{pf} (dB)*

$L_{pf}(\text{dB}) = 32.44 + 20 \log r\,(\text{km}) + 20 \log f_c\,(\text{MHz})$
$L_{pf}(\text{dB}) = 32.44 + 20 \log 1.6\,(\text{km}) + 20 \log 1000\,(\text{MHz})$
$L_{pf}(\text{dB}) = 32.44 + 4.08 + 60 = 96.5 \text{ dB}$

(b) *To determine the received signal power, Pr (dBm)*

$P_r(\text{dBm}) = P_t(\text{dBm}) + G_t(\text{dB}) + G_r(\text{dB}) - L_{pf}(\text{dB})$
$P_r(\text{dBm}) = 40 + 2 + 2 - 96.5 = -52.5 \text{ dBm}$

(c) *To determine tansmission delay,* Ί *(ns)*

Transmission delay, Ί = 3.33 ns per metre of distance
For 1600 m distance, Ί $= 3.33 \times 1600 = 5.33$ s

EXAMPLE 3.6 **Power delivered to the receiver**

A base-station transmitter has a power output of 10 watts operating at a frequency of 250 MHz. The transmitter is connected by 20 m of an RF coaxial cable, which has a loss of 3-dB/100 m specification, to an antenna that has a gain *of 9 dBi. The receiving antenna is 25 km away and has a gain of 4 dBi. There is negligible loss in the receiver feeder line, but the receiver is mismatched; the receiving antenna and feeder cable are designed for a 50-Ω impedance, but the receiver input has 75-Ω impedance, resulting into a mismatch loss of about 0.2 dB̃. Calculate the power delivered to the receiver, assuming free-space propagation.*

Solution: Refer Fig. 3.5.

Transmitter output power, $P_t = 10$ W or 10,000 mW (given)
Transmitter antenna gain, $G_t = 9$ dBi (given)
Receiver antenna gain, $G_r = 4$ dBi (given)
Carrier frequency, $f_c = 250$ MHz (given)
Distance between transmitter and receiver, $r = 25$ km (given)

Step 1. *To convert P_t (W) in P_t (dBm)*

We know that $P_t(\text{dBm}) = 10 \log P_t(\text{mW})$
Therefore, $P_t(\text{dBm}) = 10 \log 10{,}000 = +40$ dBm

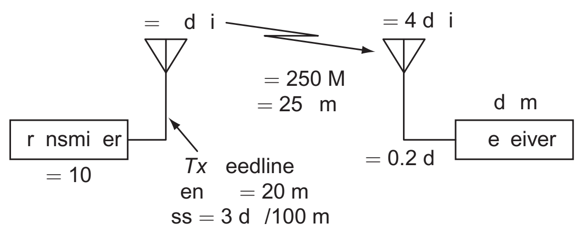

Fig. 3.5 *Illustration of wireless link for Example 3.6*

Step 2. *To determine free-space path loss, L_{pf} (dB)*

$$L_{pf}(\text{dB}) = 32.44 + 20 \log r\,(\text{km}) + 20 \log f_c\,(\text{MHz})$$
$$= 32.44 + 20 \log 25\,(\text{km}) + 20 \log 250\,(\text{MHz})$$
$$= 108.3 \text{ dB}$$

Step 3. *To find the Tx antenna RF cable loss, L_t (dB)*

Cable length = 20 m (given)
Cable attenuation = 3 dB/100m (given)
Therefore, *Tx* antenna RF cable loss, $L_t = 20 \text{ m} \times 3 \text{ dB/100m}$
Tx antenna *R* cable loss, $L_t = 0.6$ dB

Step 4. *To calculate the power delivered to the receiver, Pr (dBm)*

Rx antenna *R* cable loss, $L_r = 0.2$ dB (given)

$$P_r(\text{dBm}) = P_t(\text{dBm}) \quad L_t(\text{dB}) + G_t(\text{dB}) \quad L_{pf}(\text{dB}) + G_r(\text{dB}) - L_r(\text{dB})$$
$$P_r(\text{dBm}) = 40 \quad 0.6 + 9 - 108.3 + 4 - 0.2 = \quad 56.1 \text{ dBm}$$

Hence power delivered to the receiver is 56.1 dBm.

3.3 MOBILE POINT-TO-POINT PROPAGATION MODEL

The free-space propagation model does not apply in a mobile radio environment. Propagation path loss depends on distance of the mobile from its serving cell-site, carrier frequency of transmission f_c (or wavelength λ_c), the antenna heights of cell-site and mobile unit, and the local terrain characteristics such as buildings and hills. Mobile radio propagation path loss is uniquely different from the kind of path losses experienced in other communication media. The signal received by a mobile unit remains constant only over a small operating area and varies as the mobile unit moves. This is mainly due to the variations in the terrain conditions and presence of man-made structures surrounding the mobile unit.

Area-to-area signal prediction models are not useful for cellular communication systems because of the large uncertainty of the prediction. The area-to-area prediction model usually provides an accuracy of prediction with a measured standard deviation of about 8 dB, which means that 68 per cent of the actual path-loss data are within 8 dB of the predicted path-loss value.

Facts to Know!

Radio propagation in mobile environment is mainly characterised by three partially related effects known as multipath fading, shadowing, and path loss.

This implies that the uncertainty range for predicting the path loss is too large. When plotting received signal strengths at a given radio-path distance, the deviation from predicted values is approximately 8 dB. This is applicable in various operational geographical areas, including near the cell-site mainly due to the close-in buildings around it, as well as at a distant location due to the large variation along different radio paths.

Moreover, at a distance from the cell site, some radio paths are line-of-sight, some are partial line-of-sight, and some are out-of-sight. Thus, the received signals could be strong, normal, and weak, accordingly. Therefore, the standard deviation of 8 dB is always found along the predicted path-loss curve at any distance.

> **Facts to Know!**
>
> The standard deviation obtained from the measured data remains the same along the different path-loss curves regardless of the operating environment.

The point-to-point propagation prediction model reduces the uncertainty range by including the terrain contour profiles in the path loss predictions. In a non-obstructive condition, the direct radio path from the cell-site to the mobile unit is not obstructed by the terrain contour but the radio path may be obstructed by man-made structures. In the mobile radio environment, line-of-sight condition is generally not available. Under these conditions, the antenna height gain is calculated for every location of the mobile unit that it covers. Taking into account the antenna-height gain at various mobile locations, the path-loss slope will have a standard deviation of less than 2 to 3 dB only instead of 8 dB as observed in area-to-area prediction model.

In obstructive condition (that is, the direct path from the cell-site to the mobile unit is obstructed by the terrain contour), first the area-to-area prediction model is applied. This is then followed by diffraction loss. When the heavy foliage is close in at the mobile unit, the loss due to the foliage can be obtained from the diffraction loss. In either case, according to the theoretical model, the 40-dB/decade path loss slope applies. The mobile point-to-point prediction model provides a standard deviation of less than 3 dB from the predicted value.

The point-to-point propagation model is very useful for designing a mobile cellular system with a radius of 15 km or less for each cell. Since the path loss or received signal strength data follows the log-normal distribution, 68 per cent of predicted values obtained from a point-to-point propagation model are within 2 to 3 dB. In irregular terrain conditions, the signal received by a mobile varies due to the change in terrain conditions and presence of natural or man-made obstructions.

3.3.1 Two-ray Point-to-point Propagation Model

The point-to-point propagation model is a basic system-design tool that is used to generate a signal coverage map, an interference area map, or a handoff occurrence map. In many mobile communication systems, the maximum distance between the cell-site and the mobile is at the most only a few tens of kilometres, and the earth surface may be assumed to be flat. A simple two-ray radio path model can be used to estimate the propagation path loss and the received signal strength at the mobile, which also takes into consideration the effect of cell-site transmitter and mobile receiver antenna heights.

Figure 3.6 shows a simple model based on a direct path and a ground-reflected path.

It is obvious here that the distance between the cell-site and the mobile is much larger than the heights of the transmitting or receiving antenna heights or even the product of heights of the transmitter antenna and receiver antenna. Consider the signal transmission over a smooth, reflecting, and flat plane such as the earth.

The stronger the reflected wave is, the larger the path loss is. The stronger reflected wave occurs at a very small incident angle θ. This means that a small incident angle corresponds to a large reflection coefficient because of the nature of the reflection mechanism, and the wave reflected from the ground is a 180 phase shift. Therefore, the amount of reflected energy always becomes negative. The addition of a strong reflected wave to a direct wave tends to weaken the direct wave. The larger the incident angle of the reflected wave, the weaker the reflected wave, and the signal reception becomes the free-space condition.

From Fig. 3.6, the propagation distance r_1 of the direct path TR can be derived from the right-angled triangle TAR, that is,

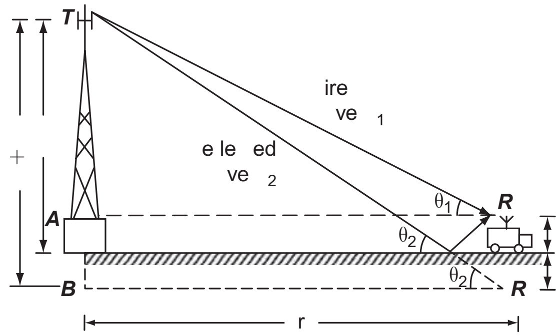

Fig. 3.6 *A mobile point-to-point propagation path-loss model*

$$TR^2 = AR^2 + TA^2$$

Or, $$r_1^{\,2} = r^2 + (h_t - h_r)^2$$

Or, $$r_1 = \sqrt{(r^2 + (h_t - h_r)^2)} \tag{3.21}$$

Similarly, the propagation distance r_2 of the reflected path TR' can be derived from the right-angled triangle TBR', that is,

$$TR'^2 = BR'^2 + TB^2$$

Or, $$r_2^{\,2} = r^2 + (h_t + h_r)^2$$

Or, $$r_2 = \sqrt{(r^2 + (h_t + h_r)^2)} \tag{3.22}$$

Since λ_c is very small as compared with 'r', a slight change in r can cause a significant change in the phase of the received signal. Depending on the difference between the phases of the direct path and reflected path, the received signal components from these two paths may enhance each other or cancel each other.

Taking into account the phase difference, the received signal power is given by

$$P_r = P_t\, G_t\, G_r\ \lambda_c/(4\,\pi\, r\,)^2 \quad 1 + a_v\, e^{\,j\,\Delta\theta} \quad ^2 \tag{3.23}$$

where a_v is the reflection coefficient, which is   1 in a mobile radio environment because of the small incident angle of the ground wave caused by a relatively low cell-site antenna height.

And $\Delta\theta$ is the phase difference between the carrier signals of the two received rays, and is related to the difference between the two propagation distances r_2 and r_1

$$P_r = P_t\, G_t\, G_r\ \lambda_c/(4\,\pi\, r\,)^2 \quad 1 \quad e^{\,j\,\Delta\theta} \quad ^2 \tag{3.24}$$

Or, $$P_r = P_t\, G_t\, G_r\ \lambda_c/(4\,\pi\, r\,)^2 \quad 1 - (\cos\Delta\theta + j\sin\Delta\theta\,) \quad ^2$$

Or, $$P_r = P_t\, G_t\, G_r\ \lambda_c/(4\,\pi\, r\,)^2 \quad 1 - \cos\Delta\theta - j\sin\Delta\theta \quad ^2$$

Or, $$P_r = P_t\, G_t\, G_r\ \lambda_c/(4\,\pi\, r\,)^2\ (1 - \cos\Delta\theta\,)^2 \qquad \text{ignoring } j \text{ terms}$$

Using binomial expansion and ignoring higher-order terms,

$$P_r = P_t\, G_t\, G_r\ \lambda_c/(4\,\pi\, r\,)^2\ \ 2\,(1 - \cos\Delta\theta\,)$$

Or, $$P_r = 2\,P_t\, G_t\, G_r\ \lambda_c/(4\,\pi\, r\,)^2\ \ 2\sin^2(\Delta\theta/2\,) \qquad \text{using trigonometric identity}$$

Or, $$P_r = 4\,P_t\, G_t\, G_r\ \lambda_c/(4\,\pi\, r\,)^2 \quad \sin(\Delta\theta/2\,) \quad ^2$$

For $\Delta\theta < 0.6$ radians, $\sin(\Delta\theta/2\,) \quad \Delta\theta/2$

Therefore, $$P_r = 4\,P_t\, G_t\, G_r\ \lambda_c\,/\,(4\,\pi\, r\,)^2\ \ (\Delta\theta/2\,)^2 \tag{3.25}$$

Now, the phase difference $\Delta\theta$ between the carrier signals of the two received rays is related to the difference between the two propagation distances r_2 and r_1 by

$$\Delta\theta = \theta_2 \quad \theta_1 = 2\pi / \lambda_c \,(r_2 - r_1)$$

Using Eqs (3.21) and (3.22),

$$\Delta\theta = 2\pi/\lambda_c \quad \sqrt{(r^2 + (h_t + h_r)^2)} - \sqrt{(r^2 + (h_t - h_r)^2)}$$

Rationalising and simplifying it using $r \quad (h_t \times h_r)$ and $r_1 \quad r$, and $r_2 \quad r$, we get

$$\Delta\theta = 2\pi / \lambda_c \,((2\, h_t\, h_r)/ r)$$

Or,

$$= 2 \times 2\pi\, (h_t\, h_r) / (\lambda_c\, r)$$

Or,

$$\Delta\theta/2 = 2\pi\, (h_t\, h_r) / (\lambda_c\, r)$$

Squaring on both sides, we get

$$(\Delta\theta/2)^2 = 4\pi^2\, (h_t\, h_r)^2 / (\lambda_c\, r)^2 \tag{3.26}$$

Using Eqs. (3.26) in Eq. (3.25), we get

$$P_r = 4\, P_t\, G_t\, G_r \quad \lambda_c / (4\,\pi\, r)^2 \quad 4\pi^2\, (h_t\, h_r)^2 / (\lambda_c\, r)^2$$

Or,

$$_r = \; _t \; _t \; _r \, \boldsymbol{h_t^2\, h_r}^2 / \boldsymbol{r}^4 \tag{3.27}$$

$$P_r\,(\text{dBm}) = P_t(\text{dBm}) + G_t\,(\text{dB}) + G_r\,(\text{dB}) + 20 \log h_t + 20 \log h_r - 40 \log r \tag{3.28}$$

Note that h_t, h_r and r should be expressed in the same units.

As seen from Eq. (3.28), at large distance ($r \quad h_t h_r$), the received power P_r falls off with distance raised to the fourth power, or at a rate of 40 dB/decade. This is a much more rapid path loss than is experienced in free space (20 dB/decade). Also, at large values of r, the received power as well as the propagation path loss in a mobile environment become independent of the frequency of transmission.

Equation (3.27) can be written as

$$P_r = P_t\, G_t\, G_r / L_{pm} \tag{3.29}$$

where L_{pm} is the propagation path loss in mobile point-to-point Lee model and is given as

$$= \boldsymbol{r}^4 / (\boldsymbol{h_t^2\, h_r}^2) \tag{3.30}$$

$$(\text{dB}) = 40 \log \boldsymbol{r} - 20 \log \boldsymbol{h_t} - 20 \log \boldsymbol{h_r} \tag{3.31}$$

The parameters h_t, h_r and r should be expressed in the same units.

Thus, it is implied that the propagation path loss in a mobile environment L_{pm} increases by 40 dB for every increase in distance by 10 times (that is, it follows 40 dB/decade or 12 dB/octave path loss). If the cell-site antenna height is doubled, there will be a reduction in path loss by 6 dB.

Expressing Eq. (3.29) in dB form, we get

$$_r(\text{dBm}) = \; _t(\text{dBm}) + \; _t(\text{dB}) + \; _r(\text{dB}) - \quad (\text{dB}) \tag{3.32}$$

The *two-ray propagation model* is found to be reasonably accurate for predicting the large-scale received signal strength over distances of several kilometres for mobile radio communication systems that use tall cell-site towers (heights which exceed 30 m), as well as line-of-sight microcell application in urban environments.

EXAMPLE 3.7 | Propagation path loss using two-ray propagation model

Determine the propagation path loss for a radio signal at 800 MHz, with a transmitting antenna height of 30 m and a receiving antenna height of 2 m, over a distance of 10 km, using two-ray mobile point-to-point propagation model. How is it comparable with that of free-space propagation path loss model?

Solution

Frequency of transmission, $f_c = 800$ MHz (given)
Transmitting antenna height, $h_t = 30$ m (given)
Receiving antenna height, $h_r = 2$ m (given)
Distance between Tx and Rx, $r = 10$ km (given)

Step 1 *To find path loss using two-ray propagation model, L_{pm} (dB)*

L_{pm} (dB) $= 40 \log r$ (m) $\quad 20 \log h_t$ (m) $\quad 20 \log h_r$ (m)
Or, L_{pm} (dB) $= 40 \log 10000 \quad 20 \log 30 \quad 20 \log 2$
Or, L_{pm} (dB) $= 160 \quad 29.54 \quad 6.02$
Hence, (dB) $= 124.44$ dB

Step 2. *To find path-loss using free-space propagation model, L_{pf} (dB)*

L_{pf} (dB) $= 32.44 + 20 \log r$ (km) $+ 20 \log f_c$ (MHz)
Or, L_{pf} (dB) $= 32.44 + 20 \log 10 + 20 \log 800$
Hence, $_f$ (dB) $= 110.5$ dB

Step 3. *To compare two-ray and free-space propagation models*

The path loss in a mobile operating environment given by a point-to-point two-ray propagation model is more than that of given by free-space propagation model. The additional path loss in a mobile environment is attributed to impairment such as reflection of a propagated signal from obstructions closer to the receiver.

The point-to-point prediction model can also be used to provide overall radio coverage of all cell-sites and to avoid co-channel interference. Moreover, the occurrence of handoff in the cellular system can be predicted more accurately.

It must be noted that under any situation, the received signal power P_r cannot be higher than that estimated from the free-space path loss model. Up to 3 km of the cell-site in a man-made environment, the buildings and road orientations affect the received signal. The received signal power at the mobile unit traveling along an in-line road can be 10 dB higher than that along a perpendicular road to the direction of the cell-site transmitting antenna. The correction factor for change in effective antenna heights should be applied. The detailed terrain contour information should be included in the path-loss predictions to reduce the uncertainty range to remain within 0.8 dB. The selection of the cell-site should be avoided in the forest area. The antenna height at the cell-site should be kept higher than the top of the tallest trees in near vicinity.

The amount of reflection and diffraction of a radio wave from an object is a function of the dimensions of the objects in terms of wavelengths. At the low-frequency end of the specified VHF range, only very large objects such as the ground and buildings cause reflections. As the frequency increases, the wavelengths become smaller and relatively small objects begin to reflect the radio waves. Due to the increase in multipath interference and a considerable attenuation of signals as they penetrate buildings, the signal loss varies with frequency of transmission, type of building construction, size and location of windows, and so on, but a rough estimate would be 20 dB more attenuation of signals at 800 MHz in a typical steel-reinforced concrete office building, reduced to about 6 dB if the user is near a window facing the cell-site.

3.3.2 Near Distance Propagation

The free-space propagation model is valid to predict the values of received signal strength for values of distance 'r' that are in the far-field of the transmitting antenna. The far-field, or *raunhofer region*, of a transmitting antenna is defined as the region beyond the far-field distance r_f, which is related to the largest linear dimension of the transmitter antenna aperture and the carrier wavelength. The Fraunhofer distance, or far-field distance is given by

$$r_f = 2\,L^2 / \lambda_c \tag{3.33}$$

where L is the largest physical linear dimension of the antenna. Moreover, to be in the far-field region, r_f must satisfy the conditions: $r_f \quad L$, and $r_f \quad \lambda_c$.

EXAMPLE 3.8 | Fraunhofer distance or far-field distance

Determine the Fraunhofer distance for a transmitting antenna with maximum dimension of 1 metre and an operating frequency of 900 MHz.

Solution

Maximum dimension of antenna, $L = 1$ m (given)
Operating frequency, $f_c = 900$ MHz or 900×10^6 Hz

Step 1. *To find the wavelength, λ_c*

We know that $\lambda_c = c / f_c$
Therefore, $\lambda_c = (3 \times 10^8 \text{ m/s}) / (900 \times 10^6 \text{ Hz}) = 0.33$ m

Step 2. *To determine the Fraunhofer distance, r_f*

We know that the Fraunhofer distance, $r_f = 2\,L^2 / \lambda_c$
Therefore, Fraunhofer distance, $r_f = 2 \times (1)^2 / 0.33 = 6$ m

Facts to Know!

The 3-dB beamwidth is a measure of the size of the antenna beam. As a general rule of thumb, the 3-dB beamwidth is directly proportional to the wavelength and inversely proportional to the antenna diameter.

In mobile radio communication systems operating in outdoor applications, within a 1 km radius (near-in distance) of the cell-site, the signal reception at the mobile unit reduces due to narrow antenna beamwidth in the vertical plane of a high-gain omnidirectional antenna used at the cell-site. Thus, the signal reception at a mobile unit less than 1 km away from the cell-site will be drastically reduced because of the large elevation angle which causes the mobile unit to be in the shadow region (outside the main lobe of the radiation pattern of high-gain omnidirectional antenna). The larger the elevation signal, the weaker the reception level due to the antenna's vertical pattern.

The nearby surroundings of the cell-site can influence the reception level either up or down when the mobile unit is within the 1 km radius. When the mobile unit is 1 km away from the cell site, the effect due to the nearby surroundings of the cell-site becomes insignificant. For land-to-mobile propagation, the effective antenna height at the cell-site strongly affects the mobile. Similarly, due to fewer roads within the 1-km radius around the cell-site, the road orientation, in-line and perpendicular, close to the cell-site can cause 10-20 dB variation in signal-reception levels.

The antenna radiation pattern is not isotropic in the vertical plane. Table 3.1 shows the typical levels of attenuation and path loss slope at increasing values of incident angles of waves (by virtue of increase in cell-site antenna heights) from a 6-dB gain omnidirectional antenna used at the cell-site and 3-metres antenna height used at the mobile which is located 100 metres away.

As the incident angle becomes larger, the 40 dB/decade path loss slope is no longer valid. If the antenna beam is directed at the mobile unit, the path-loss slope of 24 dB/decade is observed for an antenna height of 30 m, 22 dB/decade slope for an antenna height of 60 m, and 20 dB/decade for an antenna height of 90 m or above at the cell-site, which is the path-loss slope for free-space propagation condition.

To calculate near-field distance d_f, $\Delta\theta$ is set as π in Eq. (3.26), and $r = d_f$.

Table 3.1 | *Path-loss slope at different cell-site antenna heights*

Cell-site antenna height, h_t (m)	*Incident angle, θ (degrees)*	*Elevation angle, φ (degrees)*	*Attenuation, A (dB)*	*Path-loss slope, dB/decade*
30	11.77	10,72	6	24
60	21.61	20.75	16	22
90	30.4	29.6	21	20

That is, $$(\pi /2)^2 = 4\pi^2 \, (h_t \, h_r)^2 \, / \, (\lambda_c \, d_f)^2 \tag{3.34}$$

Or, $$\boldsymbol{d_f} = 4 \, (\boldsymbol{h_t} \, \boldsymbol{h_r}) \, / \, (\lambda_c) \tag{3.35}$$

It may be noted that for the signal received within the near field ($r < d_f$), the free-space path loss formula is used, and outside the near field ($r > d_f$), the mobile radio path loss formula is used for the best approximation.

3.3.3 Propagation over Flat Open Area

The propagation model for land-to-mobile transmission over a flat open area or water is depicted in Fig. 3.7.

The cell-site antenna as well as the mobile antenna is well above the open flat ground level or water level. In signal transmission over flat open area or water, there is one direct wave from cell-site transmitter to the mobile receiver. In addition to direct wave, there are always two equal-strength reflected waves, first reflected wave from the ground close to mobile unit and second reflected wave from ground or water far away from the mobile unit. The second reflected wave far away from the mobile is also strong because there are no surrounding objects near this point, and therefore considered.

The total received power at the mobile unit would be obtained by summing three components, that is, direct wave, first reflected wave, and second reflected wave. Taking into account the phase difference of these three components, the received signal power is given as

$$P_r = P_t \, G_t \, G_r \;\; \lambda_c \, / \, (4 \, \pi \, r)^2 \quad 1 + a_{v_1} e^{j\Delta\theta_1} + a_{v_2} e^{j\Delta\theta_2} \quad {}^2 \tag{3.36}$$

where a_{v1} and a_{v2} are the reflection coefficient, which is 1 in a mobile radio environment because of the small incident angles of the ground wave caused by a relatively low cell-site antenna height, and $\Delta\theta_1$ and $\Delta\theta_2$ are the path-length differences between the direct wave and two reflected waves, respectively.

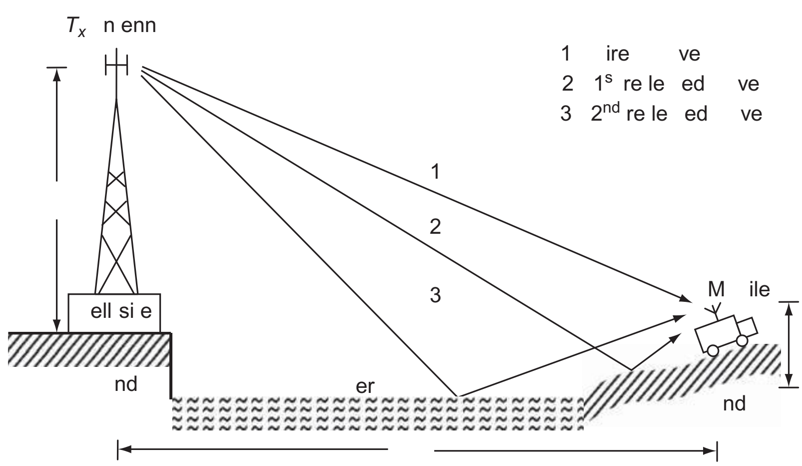

Fig. 3.7 | *A mobile point-to-point propagation path-loss model over water*

Or, $\quad P_r = P_t\, G_t\, G_r \;\; \lambda_c / (4\,\pi\, r)^2 \quad 1 - e^{j\Delta\theta_1} - e^{j\Delta\theta_2} \quad ^2$

Or, $\quad P_r = P_t\, G_t\, G_r \;\; \lambda_c / (4\,\pi\, r)^2 \quad 1 - (\cos\Delta\theta_1 + j\sin\Delta\theta_1) - (\cos\Delta\theta_2 + j\sin\Delta\theta_2) \quad ^2$

Or, $\quad P_r = P_t\, G_t\, G_r \;\; \lambda_c / (4\,\pi\, r)^2 \; (1 - \cos\Delta\theta_1 - \cos\Delta\theta_2)^2$ Ignoring j terms

Since $\Delta\theta_1$ and $\Delta\theta_2$ are very small for land-to-mobile path, then

$$\cos\Delta\theta_1 \quad 1, \cos\Delta\theta_2 \quad 1.$$

Therefore, $\quad P_r = P_t\, G_t\, G_r \;\; \lambda_c / (4\,\pi\, r)^2 \; (1 - 1 - 1)^2$

Or, $$P_r = P_t\, G_t\, G_r \;\; \lambda_c / (4\,\pi\, r)^2 \tag{3.37}$$

Or, $$P_r = P_t\, G_t\, G_r / L_{pw} \tag{3.38}$$

where L_{pw} is the propagation path loss for land-to-mobile propagation over water, and is given as

$$L_{pw} = (4\,\pi\, r / \lambda_c)^2 \tag{3.39}$$

Or, $\quad L_{pw} = (4\,\pi\, r f_c / 3 \times 10^5)^2$ (where r is in km)

Or, $\quad L_{pw}\,(\text{dB}) = 20 \log (4\,\pi / 3 \times 10^5) + 20 \log r\,(\text{km}) + 20 \log f_c\,(\text{Hz})$

Hence, $\quad {}_{w}\,(\text{dB}) = 32.44 + 20 \log \boldsymbol{r}\,(\text{km}) + 20 \log \boldsymbol{f_c}\,(\text{MHz})$

This is the path loss equation over flat open area or water. It is seen that L_{pw} varies as 20 dB/decade with distance, same as that of free-space path loss. Thus, it is implied that the propagation path loss for land-to-mobile propagation over water varies as 20 dB/decade or 6 dB/octave with distance between the transmitter and the receiver, which happens to be exactly same as that of free-space path loss.

It is concluded here that the path loss for land-to-mobile propagation over land in normal terrain, 40 dB/decade as per the two-ray propagation model, is different for land-to-mobile propagation over flat open area or water, 20 dB/decade (which is same as the free-space path loss).

3.4 OUTDOOR PROPAGATION PATH-LOSS MODELS

Radio propagation models are empirical in nature, that is, these are developed based on large collections of measured data for the specific wireless communication environment. For any model, the collection of data has to be sufficiently large to provide enough likeliness to all kinds of situations that can happen in that specific environment. Like all empirical models, radio propagation models do not point out the exact behaviour of a link, rather, they predict the most likely behaviour of the link that they may exhibit under the specified conditions.

Facts to Know!

Weather phenomena such as rain, fog, dust or snowstorms, air disturbances, and significant differences in temperature at different altitudes can affect the performance and reliability of wireless links.

Different propagation models have been developed to predict path loss over irregular terrain. Some of the most popular outdoor propagation models are Okumara, Hata, COST 231 Hata, Longley-Rice, and Durkin model.

3.4.1 Okumara Propagation Model

The Okumura propagation model is best suited for large cell coverage (distances up to 100 km) in urban areas, and it can extrapolate predictions up to 3 GHz. It can be used for effective base-station antenna heights from 30 m to 1000 m. The effective mobile receiver antenna height is taken as 3 m. This model has been proven to be accurate and is used by computer simulation tools. A simplified version of a path loss model, proposed by Okumura, for propagation in an urban mobile environment, is expressed as

$$L_{pO}(\text{dB}) = L_{pf}(\text{dB}) + \alpha_m (f_c, r) - \alpha_t - \alpha_r - \Sigma\, \alpha_c \tag{3.40}$$

where L_{pf} is the free-space propagation path loss in dB

α_m is the median attenuation relative to free space, and is a function of f_c and r

α_t is the effective base station antenna height (h_t) gain factor, and varies at a rate of 20 dB/decade, and is given as

$$\alpha_t = 20 \log (0.005 \times h_t) \qquad 1000 \text{ m} > h_t > 30 \text{ m} \tag{3.41}$$

α_r is the effective mobile receiver antenna height (h_r) gain factor, and varies at a rate of 10 dB/decade for heights less than 3 m, and is given as

$$\alpha_r = 20 \log (0.33 \times h_r) \qquad 10 \text{ m} > h_r > 3 \text{ m} \tag{3.42}$$

$$\alpha_r = 10 \log (0.33 \times h_r) \qquad h_r \leq 3 \text{ m} \tag{3.43}$$

α_c is the correction factor gain such as type of environment (suburban area, open area), water surfaces, isolated obstacle, etc.

The Okumara model is based on measured data in different terrains with specified system parameters. The standard deviation between the measured and predicted path loss data is about 10 –14 dB. This model is ideal for using in cities with many urban structures but not many tall building structures. The Okumara model serves as a base for the Hata Propagation Model.

3.4.2 Hata Propagation Model

The Hata propagation path loss model is an empirical model based on Okumura's model, and is extensively used in macrocellular wireless mobile communication systems for large cells exceeding 1 km radio coverage. Morphology, and obstructions (natural or man-made) are causes of propagation impairments that result in signal loss and are added to the path loss. The base-station antenna is generally installed on towers or rooftops of high-rise buildings. A simplified version of a path-loss model, proposed by Hata, for propagation in a dense urban mobile environment is expressed as

$$(\text{dB}) = 68.75 + 26.16 \log \boldsymbol{f_c} - 13.82 \log \boldsymbol{h_t} + (44.9 - 6.55 \log \boldsymbol{h_t}) \log \boldsymbol{r} \tag{3.44}$$

where L_{pH} is the median value of the propagation path loss in dB

f_c is the frequency of transmission in MHz

h_t is the effective cell-site antenna height in metres ranging from 30 m to 200 m

r is the distance of the mobile from cell-site in km

The Hata path loss propagation model is applicable in the mobile communication systems operating in the frequency range between 150 MHz and 1000 MHz, having an effective base station transmitter antenna height between 30 m to 200 m. Usually, a mobile antenna height of 2 m is taken for all practical purpose. It is seen that path loss increases with frequency (26.16 dB per decade) as well as distance at a much greater rate than in the free-space (20 dB per decade), as expected. However, path loss is reduced due to increased cell-site antenna height.

If the mobile receiver antenna height is different than 2 m then the mobile antenna correction factor is required to be applied to Hata propagation path-loss model. An effective mobile receiver antenna height may range from 1 m to 10 m.

The general expression for median path loss in urban area is given by

$$L_{pH\,(urban)}(\text{dB}) = 69.55 + 26.16 \log f_c - 13.82 \log h_t - \alpha_r + (44.9 - 6.55 \log h_t) \log r \tag{3.45}$$

where α_r is the correction factor for effective mobile antenna height, which depends on the size of the coverage area. For example,

For a large city, α_r (dB) $= 8.29 (\log 1.54\, h_r)^2 - 1.1$ for $f_c \leq 300$ MHz

α_r (dB) $= 3.2 (\log 11.75\, h_r)^2 - 4.97$ for $f_c > 300$ MHz

For small to medium city, α_r (dB) $= (1.1 \log f_c - 0.7)\, h_r - (1.56 \log f_c - 0.8)$

The median path loss in suburban area is given by

$$L_{pH\,(suburban)}\,(\text{dB}) = L_{pH\,(urban)}\,(\text{dB}) - 2\ \ \log\,(f_c/28)^{\,2} - 5.4 \tag{3.46}$$

The median path loss in open rural area is given by

$$L_{pH\,(open)}\,(\text{dB}) = L_{pH\,(urban)}\,(\text{dB}) - 4.78\,(\log f_c)^2 + 18.33 \log f_c - 40.94 \tag{3.47}$$

EXAMPLE 3.9 Propagation path loss in dense urban city

Determine the propagation path loss for a radio signal at 800 MHz, with a transmitting antenna height of 30 m and a receiving antenna height of 2 m, over a distance of 10 km in a dense urban mobile environment, using Hata propagation path-loss model. If the free-space propagation path-loss is 110.5 dB for the given system parameters, how is Hata propagation path loss comparable with that of free-space propagation path loss?

Solution

Frequency of transmission, f_c = 800 MHz (given)
Transmitting antenna height, h_t = 30 m (given)
Receiving antenna height, h_r = 2 m (given)
Distance between Tx and Rx, r = 10 km (given)

Step 1. *To find propagation path loss, L_{pH}(dB) using Hata model*

The propagation path loss using Hata propagation path loss model is given by

$$L_{pH}(\text{dB}) = 68.75 + 26.16 \log f_c - 13.82 \log h_t + (44.9 - 6.55 \log h_t) \log r$$

$$L_{pH}(\text{dB}) = 68.75 + 26.16 \log 800 - 13.82 \log 30 + (44.9 - 6.55 \log 30) \log 10$$

Hence, $L_{pH}(\text{dB}) = 159.5$ dB

Step 2. *To comparison L_{pH}(dB) with free-space propagation path loss*

Free-space propagation path loss, L_f (dB) = 110.5 dB (given)
Difference between two path loss values = 159.5 − 110.5 = 49 dB
There is 49 dB more attenuation in mobile environment using Hata propagation path loss model as compared to that in free space for the given system parameters.

EXAMPLE 3.10 Propagation path loss in a large city

Determine the propagation path loss for a radio signal at 900 MHz cellular system operating in a large urban city, with a base station Tx antenna height of 100 m and mobile R_x antenna height of 2 m. The mobile unit is located at a distance of 4 km. Use the Hata propagation path loss model.

Solution

Frequency of transmission, f_c = 900 MHz (given)
Transmitting antenna height, h_t = 100 m (given)
Receiving antenna height, h_r = 2 m (given)
Distance between Tx and Rx, r = 4 km (given)

Step 1. *To find $L_{pH\,(urban)}$ (dB)*

The general expression for median path loss in urban area is given by

$L_{pH\,(urban)}$ (dB) $= 69.55+26.16 \log f_c - 13.82 \log h_t - \alpha_r + (44.9 - 6.55 \log h_t) \log r$

where α_r is the correction factor for effective mobile antenna height, which depends on the size of the coverage area.

Step 2. *To determine α_r for a large city*

α_r (dB) $= 3.2 (\log 11.75\, h_r)^2 - 4.97$ for given $f_c = 900$ MHz

α_r (dB) $= 3.2 (\log 11.75 \times 2)^2 - 4.97$

α_r (dB) $= 1.045 \approx 1$ dB

Step 3. *To determine $L_{pH\,(urban)}$ (dB) for specified cellular system parameters*

$$L_{pH\,(urban)} \text{ (dB)} = 69.55 + 26.16 \log 900 - 13.82 \log 100 - 1 + (44.9 - 6.55 \log 100) \log 4$$

Hence, $_{(ur\ \ an)}$ (dB) $= 137.3$ dB

3.4.3 COST-231 Hata Propagation Model

A model that is widely used for predicting path loss in mobile wireless systems operating from 500 MHz to 2000 MHz is the COST-231 Hata model. It also contains corrections for urban, suburban and open rural environments. The general expression for median path loss in the urban area as given by COST-231 Hata model is

$$L_{pCH\,(urban)} \text{ (dB)} = 46.3 + 33.9 \log f_c - 13.82 \log h_t - \alpha_r + (44.9 - 6.55 \log h_t) \log r + C_m \qquad (3.48)$$

The parameter C_m is defined as 3 dB for urban environments, and 0 dB for suburban or open environments.

3.4.4 Longley–Rice Propagation Model

The Longley–Rice propagation model, also known as the Irregular Terrain Model (ITM), calculates large-scale median propagation loss relative to free-space propagation loss over irregular terrain. It is applicable for point-to-point wireless communication systems operating over different terrain conditions in the frequency range from 40 MHz up to 100 GHz. It has been used extensively for frequency planning in television broadcasting and for preparing the tables of channel allocations for VHF/UHF broadcasting. The modified model can be applied to radio propagation in urban areas for mobile radio application by adding the urban factor which considers urban clutter near the mobile receiving antenna.

The Longley–Rice propagation model has two parts—a model for predictions over an area and a model for point-to-point link predictions.

- The area-to-area prediction model is used when the terrain path profile is not available, and provides a method to estimate the path-specific parameters such as terrain irregularity, horizon distance and angular trans-horizon distance between transmitting and receiving antennas, horizon elevation angle, etc.
- The Point-to-point wireless link prediction model is used when a detailed path profile is available, and path-specific parameters can be easily determined.

The main drawback of the Longley–Rice propagation model is that it does not consider the effect of multipath, buildings, foliage, and other environmental factors.

3.4.5 Durkin's Propagation Model

In 1969 and 1975, Durkin proposed a computer simulator for predicting field strength contours over irregular terrain. Durkin's path loss simulator consists of two parts:

- Access a topographic database of a proposed service area.
- Using quantised maps of service area heights, reconstruct the ground profile information along the radial line-of-sight path joining the transmitter with the receiver, including diffraction from obstacles along this radial path.

Facts to Know!

A Fresnel zone is an elliptical area between two directional antennas. When setting up a wireless link between them, it is recommended that no more than 40% of the Fresnel zone be blocked by obstruction in order to maintain a reliable connection.

The model assumes that the receiving antenna receives all of its energy along that radial path joining the transmitter to the receiver. It implies that the model does not consider multipath propagation due to reflections from other surrounding objects and local scatterers. The model identifies isolated weak signal spots on the radial path. The simulation algorithm calculates the estimated path loss between the fixed locations of the transmitter and receiver. Thereafter, the simulated location of the receiver is iteratively moved to different locations around the fixed transmitter location in the given service area. The combined results of all these iterations are used to deduce the contour of the received signal strength and hence the propagation path loss.

Firstly, the algorithm decides whether a line-of-sight path exists between the transmitter and the receiver. This is done by computing the difference for each point along the radial path between the height of this path from the heights of the ground profile reconstructed earlier. If the difference is positive at any point along the profile, it means that a line-of-sight path does not exist. If the difference at all points along the profile is negative, it implies that a clear line-of-sight path exists between the transmitter and receiver antennas and there is no obstruction due to terrain profile.

Let the path have a clear line-of-sight between the transmitter and receiver antennas. The next step is to check the clearance of the first Fresnel zone. There may be two outcomes:

- If the first Fresnel zone of the radio path is cleared without any obstruction from the ground profile then free space path loss formula can be applied.
- If the first Fresnel zone of the radio path is not cleared, and there is an obstruction that just meets the radial path between the transmitter and the receiver, then the received signal strength is 6 dB less than the corresponding value computed from free-space condition. This is due to the diffraction loss as a result of some part of incident energy at the edge of the obstruction is lost. This is the case of line-of-sight but with inadequate first Fresnel-zone clearance. The simulator program calculates the free-space path loss as well as the mobile point-to-point path loss. The least value out of these two is taken for determining the received signal strength for the given terrain profile. The loss due to inadequate first Fresnel-zone clearance is calculated and added to it. In case of non line-of-sight condition, the simulator program calculates the loss due to single diffraction edge, two diffraction edges, three diffraction edges, and so on, which is then summed up together to add to the free-space path loss.

Facts to Know!

As a general rule of thumb, the first Fresnel zone must be kept free of obstructions in order to receive acceptable transmission under free space propagation conditions.

Durkin's propagation model has been adopted by the Joint Radio Committee in the UK for estimation of effective mobile radio coverage areas.

3.5 INDOOR PROPAGATION PATH-LOSS MODELS

An indoor propagation environment is more hostile than a typical outdoor propagation environment. The indoor propagation model estimates the path loss inside a room or a closed area inside a building delimited by walls of any form. Phenomena like lack of a line-of-sight condition, multipath propagation, reflection, diffraction, shadow fading, heavy signal attenuation, close proximity of interference sources, and rapid fluctuations in the wireless channel characteristics have a significant influence on the received power in indoor propagation. Moreover, the ranges involved need to be of the order of 100 metres or less. Typically, multipath propagation is very important in indoor environments. Simple empirical propagation models are therefore not sufficient.

The indoor propagation models are suitable for wireless devices designed for indoor application to approximate the total path loss an indoor wireless link may experience. Typically, such wireless devices use the lower microwave bands of around 2.4 GHz. However, the model applies to a much wider frequency range. The indoor propagation models can be used for picocells in cellular network planning.

3.5.1 Partition and Building Penetration Losses

Generally, an indoor environment comprises of buildings having a variety of partitions and obstacles which cause additional propagation path loss. Different types of partitions are used such as those for forming part of the building structures, called *hard* or *fixed partitions*, and obstacles inside the building, called *soft* or *movable partitions*. In such a building in which wireless devices are used, signals may pass through fixed/movable furniture, computers, moving people within a room as well as through walls, doors, and/or windows if the piconet users are located in different rooms on the same floor or in different floors.
To determine partition losses, the following assumptions may be made.

- Free space path loss occurs between partitions.
- Signal strength may drop suddenly as it passes through partitions.
- The amount of partition loss depends on the type of partitions.
- Signal losses between different floors of the same building exhibit special behaviour and are required to be modeled separately.

Total partition loss is the sum of partition attenuation factor and floor partition factor.

$$L_{PL}\,(\text{dB}) = \sum_{i=1}^{N_p} PA_i(dB) + \sum_{j=1}^{N_f} A_j(dB) \tag{3.49}$$

where L_{PL} is the total partition loss;

PA_i is the partition attenuation factor due to the ith partition and there are a total of N_p partitions on the same floor of the building through which the signal travels;

A_j is the floor attenuation factor due to the jth floor and there are a total of N_f floors in that building through which the signal travels.

Table 3.2 gives typical partition losses for different types of fixed partitions used in a building.

Table 3.2 *Typical partition losses in a building*

S. No.	*Type of partition*	*Partition loss (dB)*
1.	Metallic	26
2.	Aluminium wall	20.4
3.	Concrete wall	13
4.	Foil insulation within wall	3.9
5.	Double plasterboard wall	3.4
6.	Cloth partition	1.4

Another important factor for indoor propagation when the transmitter is located outside the building is *building penetration loss*. Signal penetration within the building is a function of frequency of transmissions, height, and the building materials. Signal strength received inside the building increases with height, whereas penetration loss decreases with increasing frequency. Building penetration loss decreases at a rate of about 1.9 dB per floor from the ground floor up to the 15^{th} floor and then starts increasing due to shadow effects of adjacent tall buildings. Building penetration loss on the ground floor of the building typically varies from 8 dB to 20 dB for a frequency range of 900 MHz to 2 GHz respectively. The number and size of windows as well as the type of material used for construction of windows in the building have a significant contribution towards total building penetration loss. Building penetration loss behind windows is typically 6 dB less than that behind exterior walls. Moreover, plate glass has a penetration loss of about 6 dB as compared to 3–30 dB from metallic lead-lined glass.

3.5.2 Log-distance Propagation Model

The log-distance path loss model is a radio propagation model that predicts the path loss which is encountered by a signal inside a building or densely populated areas over distance. The model is applicable to indoor propagation modeling. Log-distance path loss model is based on distance-power law, and is expressed as

$$L_{pLog}\,(\mathrm{dB}) = L_p\,(\mathrm{r}_0)\,(\mathrm{dB}) + 10\,\gamma\,\log\left(\frac{r}{r_0}\right) + {}_{g}\,(\mathrm{dB}) \tag{3.50}$$

where $L_p\,(r_0)$ is the path loss at the reference distance r_0 (usually taken as 1 m), and is a function of frequency of transmission, f_c and distance between transmitter and receiver, r.

γ is the path-loss propagation constant,

r is the distance between the transmitter and receiver in metres,

${}_{g}$ is a normal random variable with zero mean, reflecting the attenuation caused by flat fading. In case of no fading, ${}_{g}$ is taken as 0. In case of only shadow fading or slow fading, this random variable may have Gaussian distribution with σ standard deviation in dB, resulting in log-normal distribution of the received power in watts. In case of only fast fading caused by multipath propagation, the corresponding gain may be modeled as a random variable with Rayleigh distribution or Ricean distribution.

Empirical measurements of path loss exponent (γ) and standard deviation (σ) corresponding to different indoor propagation conditions are shown in Table 3.3.

Table 3.3 *Empirical measured values of γ and σ*

Type of indoor structure	*requency, f_c*	*Path-loss exponent, γ*	*Standard deviation, σ*
Vacuum or infinite space	-------	2.0	0
Grocery store	914 MHz	1.8	5.2
Retail store	914 MHz	2.2	8.7
Office with soft partition	900 MHz; 1.9 GHz	2.4; 2.6	9.6; 14.1
Office with hard partition	1.5 GHz	3.0	7.0
Suburban home or street	900 MHz	3.0	7.0
Metalworking factory	1.3 GHz	1.6 for LOS; 3.3 for obstructive path	5.8 for LOS; 6.8 for obstructive path

Thus, the log-distance model is a combination of a modified power-distance law and a log-normal fading model that also uses empirical data. It is highly recommended to ultimately take field measurements to verify that the model is accurately characterising the operating environment. If found necessary, the model can be corrected using the measured data,

3.5.3 Attenuation Factor Path Loss Model

The attenuation factor path loss model is a radio propagation model that predicts the path loss which includes the effect of type of the building as well as the signal variations caused by partitions and obstacles present inside the building. The attenuation factor model is expressed as

$$L_{pA}\ (\text{dB}) = L_p(r_0)\ (\text{dB}) + 10\,\gamma \log\left(\frac{r}{r_0}\right) + L_{PL}\,(\text{dB}) \tag{3.51}$$

where L_{PL} is the total partition loss and is given by

$$L_{PL}\,(\text{dB}) = \sum_{i=1}^{N_p} PA_{\ i}(\text{dB}) + \sum_{j=1}^{N_f} A_{\ j}(\text{dB}) \tag{3.52}$$

Substituting it in the expression for L_{pA}, we get

$$L_{pA}\ (\text{dB}) = L_p(r_0)\ (\text{dB}) + 10\,\gamma \log\left(\frac{r}{r_0}\right) + \sum_{i=1}^{N_p} PA_{\ i}(\text{dB}) + \sum_{j=1}^{N_f} A_{\ j}(\text{dB}) \tag{3.53}$$

As mentioned previously, PA signifies partition attenuation factor for a specific obstruction experienced by a ray drawn between the transmitter and receiver and there could be N_p number of rays in 3D. A represents a floor attenuation factor for N_f number of floors of the building.

If the path loss is required to be determined for the indoor propagation in the same floor of the building, then the path-loss exponent value for that floor should be known. In that case, the path loss using attenuation factor model can be rewritten as

$$L_{pA}\ (\text{dB}) = L_p(r_0)\ (\text{dB}) + 10\,\gamma_S\ \log\left(\frac{r}{r_0}\right) + \sum_{i=1}^{N_p} PA_{\ i}(\text{dB})\ +\ A\ \ (\text{dB}) \tag{3.54}$$

where γ_S is the path-loss exponent value for the same floor, and varies from 1.6 to 3.3 in an indoor environment. The attenuation factor path loss model provides 4 dB standard deviation between the measured and predicted path-loss as compared to 13 dB given by log-distance model. Thus, this model provides flexibility and excellent accuracy.

If the path loss exponent value for multiple floors is known based on extensive measurements, and denoted by γ_M, then the expression for L_{pA} can be simplified as

$$L_{pA}\ (\text{dB}) = L_p(r_0)\ (\text{dB}) + 10\,\gamma_M\ \log\left(\frac{r}{r_0}\right) + \sum_{i=1}^{N_p} PA_{\ i}(\text{dB}) \tag{3.55}$$

The indoor propagation path loss in a multi-floor building is given by the sum of free-space path loss ($\gamma = 2$) and additional loss factor which increases exponentially with distance, and is expressed as

$$L_{pA}\ (\text{dB}) = L_p(r_0)\ (\text{dB}) + 20 \log\left(\frac{r}{r_0}\right) + \beta \times \text{r} + \sum_{i=1}^{N_p} PA_{\ i}(\text{dB})\ +\ \text{FAF (dB)} \tag{3.56}$$

where β is the attenuation constant for the wireless channel, expressed in dB/m, and depends on frequency and number of floors in the building. The value of β decreases with increase in frequency and increases as the number of floors in the building increases. The typical measured value of β at 850 MHz is 0.62 and 0.48 for a 4-floor and 2-floor building respectively. Similarly, typical measured value of β for a 4-floor building is 0.57 and 0.47 at 1.7 GHz and 4.0 GHz respectively. This model is also called *ree Space plus Linear Path Attenuation Model.*

3.6 SIGNAL ATTENUATION DUE TO FOLIAGE

Foliage loss is the loss in signal power during propagation because of the presence of foliage environment on its way. Foliage mainly comprises of dense trees or vegetation of any type. Foliage loss is the signal loss due to the sizes of trunks of trees, branches and leaves (including their texture and thickness) of trees; density of trees (that is, spacing between adjacent trunks, branches, and leaves); distribution of sizes of trunks, branches, and leaves of trees in a thick forest area; height of the trees relative to the antenna heights of the cell-site and mobile unit.

When the foliage (vegetation) is uniformly heavy and the lengths of the radio path are quite short, the signal loss due to foliage can be treated equivalent to wire-line loss (dB/m) on a linear scale. When the foliage is non-uniform and the radio path length is comparatively long, the signal loss due to foliage is used in terms of dB/octave or dB/decade on a logarithmic scale. In general, foliage loss occurs approximately proportional to f_c^{-4}, where f_c is the frequency of transmission. For example, at 800 MHz the signal loss due to foliage along the radio path is 40 dB/decade, which is 20 dB more than the free-space path loss, with the same amount of additional loss in a mobile environment.

Therefore, in mobile communications involving foliage loss, the total signal loss would be approximately 60 dB/decade (= 20 dB/decade free-space loss + 20 dB signal loss due to foliage loss + 20 dB signal loss due to mobile environment). Close proximity of foliage at the base station transmitter site always heavily attenuates the received signal strength and thus degrades the received signal quality. Therefore, the cell-site must be placed away from dense trees. If the heavy foliage is close to the mobile unit, the additional signal loss due to foliage must be considered. However, a rough estimate should be sufficient for the purpose of overall system design. If the sizes of tree leaves are very large and thick as in the case of tropic zones, the signal can hardly penetrate, thereby resulting in total blockage of signals.

3.7 LONG-DISTANCE PROPAGATION

At microwave frequencies in the range of 1000 MHz, the electromagnetic waves are neither propagated along the surface of the earth nor reflected by the ionosphere. But the long-distance (up to as large as 1000 km) propagation occurs due to the phenomenon of super refraction or duct propagation. The distance of tropospheric wave propagation is much greater than the line-of-sight propagation. In reality, the standard atmospheric conditions hardly exists where the dielectric constant is assumed to decrease uniformly with height above the earth's surface and attains the value of unity at zero air density. The dielectric constant of air essentially depends on the weather conditions—for example, it is slightly greater than unity in dry air and increases with the presence of moisture in the air, there are often many layers of air, one above the other, having different moisture contents and temperatures. These types of atmospheric conditions cause duct formation in addition to reflection, refraction and scattering of electromagnetic waves at microwave frequencies.

Facts to Know!

Troposphere is the region at about 10 km above the surface of the earth and the temperature decays at the rate of 6.5 degree C per km till it reaches at about-50 degree C at the upper boundary of the troposphere.

At microwave frequencies, the region where the variation of dielectric constant is usually high or refractive index decreases rapidly with height, actually traps the electromagnetic energy and causes it to travel along the surface of the earth. This usually happens near the surface of the earth within 50 metres

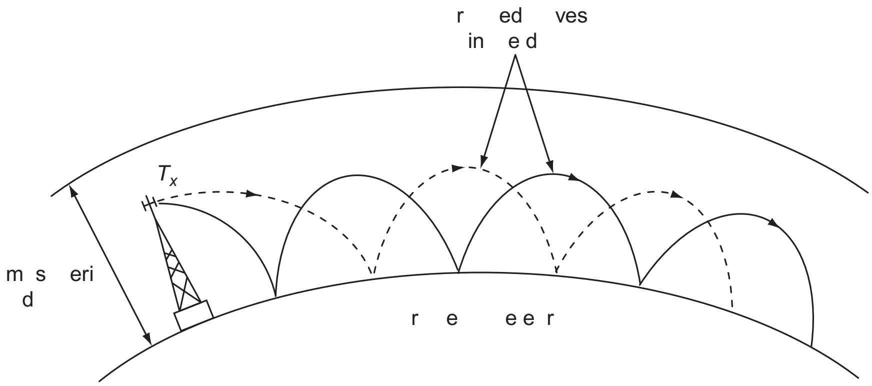

Fig. 3.8 *Long-distance propagation in the atmospheric duct*

of the troposphere. Within the troposphere region, the atmosphere has a dielectric constant slightly greater than unity due to high air density and decreases to unity at greater heights where the air density approaches to zero. Therefore, the electromagnetic waves are continuously refracted in the duct by the troposphere and reflected by the earth's surface. This results in the propagation of waves around the curvature of the earth far beyond the line-of-sight range.

This phenomenon of long-distance propagation due to duct formation is depicted in Fig. 3.8.

Thus, the main requirement for duct formation is super refraction which is a result of temperature inversion in the atmosphere with height. This means in the troposphere region, the temperature increases with height rather than usual decrease of temperature at the rate of 6.5 degree C per km in the standard atmosphere. It may be noted that only those electromagnetic waves are trapped in the troposphere which enter with small angles with respect to horizon. The duct propagation condition also depends on the location of the transmitter with respect to the duct formation. The most favourable condition is when the transmitter happens to be inside the duct. Ground-based ducts of about 1.5 metres thickness are mostly formed over the sea. However, less frequently and temporarily ground-based ducts are formed over land areas due to the cooling air of the earth. The elevated ducts of about up to 300 metres thickness are formed at an elevation of about 300–1500 metres.

The long-distance wave propagation due to tropospheric ducts may cause interference in those wireless communication systems where frequencies are repeatedly used in different areas, which is the case in cellular mobile communication systems employing the frequency-reuse concept. The interference can be reduced by proper design of antenna system such as by either using low-power transmitters, or directional antennas, or umbrella antenna beam patterns. Another disadvantage of long-distance propagation is occurrence of stronger signal levels at a particular location at one time but weaker signal levels at the same location at another time. This is caused by propagation of electromagnetic waves in a non line-of-sight manner in the varying atmospheric conditions at different heights.

Key Terms

- Attenuation
- Beamwidth
- Coverage area
- dB
- dBi
- dBm
- EIRP
- Flat fading
- Foliage
- Free space
- Free-space loss
- Gain
- Line-of-sight
- Multipath
- Omnidirectional
- Path loss
- Propagation
- Reflection
- Service area
- Shadow fading
- Slow fading
- Transmission medium
- Troposphere

Summary

In this chapter, the methods for estimation of propagation path loss and the received signal power levels under different operating conditions are presented. In a mobile-to-mobile land communication, both the transmitter and the receiver are in motion. Buildings and obstacles between the transmitter and the receiver usually obstruct the propagation path. The parameters of the path loss models must be evaluated taking into consideration the obstructed path, the diffraction around building corners and the rooftops. Unlike radio propagation in micro-cells (microcellular systems adapt cell size of a few kilometres to tens of kilometres in mostly urban areas), special case studies need to be carried out and necessary correction in parameter values corresponding to a specific operating environment must be done. Similarly, a detailed near-field propagation study within the building needs be conducted before deciding a path loss model in picocells covering a range of 20 m to 100 metres or indoor areas.

Important Equations

- ❑ $P_r = P_t G_t G_r \lambda_c / (4 \pi r)^2$ (3.12)
- ❑ $L_{pf} = (4 \pi r / \lambda_c)^2$ (3.13)
- ❑ L_{pf} (dB) = 32.44 + 20 log r (km) + 20 log f_c (MHz) (3.14)
- ❑ $P_r = P_t G_t G_r / L_{pf}$ (3.15)
- ❑ P_r (dBm) = P_t (dBm) + G_t (dBi) + G_r (dBi) − L_{pf} (dB) (3.16)
- ❑ $P_r = P_t G_t G_r h_t^2 h_r^2 / r^4$ (3.27)
- ❑ $L_{pm} = r^4 / (h_t^2 h_r^2)$ (3.30)
- ❑ L_{pm} (dB) = 40 log r −20 log h_t − 20 log h_r (3.31)
- ❑ P_r (dBm) = P_t(dBm) + G_t (dB) + G_r (dB) − L_{pm} (dB) (3.32)
- ❑ $d_f = 4 (h_t h_r) / (\lambda_c)$ (3.35)
- ❑ L_{pH}(dB) = 68.75 + 26.16log f_c −13.82log h_t + (44.9 − 6.55log h_t) log r (3.44)

Short-Answer Type Questions with Answers

A3.1 State the major factors causing propagation path loss.

The propagation path loss is the attenuation in the signal power as the signal propagates from transmitter to receiver through the wireless medium. There are numerous factors which influence the signal propagation. Some of the factors causing propagation path loss include multipath propagation, reflection, refraction, diffraction, scattering, and absorption in mobile communications.

A3.2 Distinguish between large-scale propagation path loss models and small-scale propagation path-loss models.

Propagation path-loss models have traditionally focused on predicting the average received signal strength at a given distance from the transmitter. This is termed as large-scale propagation path-loss models. The propagation models that are based on the rapid fluctuations of the received signal strength in close proximity to a particular receiver location are termed small-scale propagation path loss models or fading propagation models.

A3.3 Why does signal propagation through wireless media vary significantly

The significant variation in received signal level depends on the operating terrain conditions, frequency of transmission, sources of interference, speed of the mobile vehicle, and many other dynamic factors. Thus, the signal propagation through wireless media is heavily location-specific.

A3.4 Define the term EIRP. How is it related to transmitter power and Tx antenna gain

The effective isotropic radiated power (EIRP) of a transmitting system in a given direction is defined as the transmitter power that would be needed with an isotropic radiator, to produce the same power density in that direction. The relationship between EIRP, transmitter power (P_t) and *T*x antenna gain (G_t) is given by EIRP $= P_t\, G_t$

A3.5 Comment on the gain of receiver antennas.

A receiving antenna absorbs some of the signal energy from electromagnetic waves that pass through it. Since the signal energy in the radio wave is directly proportional to the area through which it passes, a receiving antenna having large area will intercept more signal energy than a smaller one. Receiving antennas are also more efficient at absorbing signal power from some directions than from other directions, depending upon its characteristics. That simply means receiver antennas too have gain.

A3.6 Why can free space propagation model not be applied in a mobile radio environment

Propagation path loss depends on distance of the mobile subscriber from its serving cell-site, carrier frequency of transmission, the antenna heights of the cell-site and mobile unit, and the local terrain characteristics such as buildings and hills. Since the free-space propagation model depends only on the distance between the cell-site and the mobile subscriber as well as the carrier frequency of transmission, it is not suitable in a mobile radio environment.

A3.7 Are area-to-area signal-prediction models applicable for cellular communication systems If not why

Area-to-area signal-prediction models are generally not used to estimate the radio coverage area for cellular communication systems because of the large uncertainty of the prediction. The area-to-area prediction model usually provides an accuracy of prediction with a measured standard deviation of about 8 dB, which means the actual path-loss data are within 8 dB of the predicted path-loss values. This implies that the uncertainty range for predicting the path loss is too large. When plotting received signal strengths at a given radio-path distance, the deviation from predicted values is approximately 8 dB.

A3.8 List the typical application areas of using the point-to-point propagation model in mobile communication systems.

The point-to-point propagation model is a basic system design tool in mobile communication systems that is used to generate a signal coverage map, an interference area map, or a handoff occurrence map.

A3.9 How does the propagation path loss for land-to-mobile propagation over water vary?

The propagation path loss for land-to-mobile propagation over water varies as 20 dB/decade or 6 dB/octave with distance between the transmitter and the receiver. This happens to be exactly the same as that of free-space path loss.

A3.10 What is meant by foliage Define foliage loss.

Foliage means dense trees or vegetation of any type. Foliage loss is the loss in signal power during propagation because of presence of foliage environment in its path. Foliage loss may occur due to the sizes of trunks of trees, branches and leaves of trees including their texture and thickness; density of trees; distribution of sizes of trunks, branches, and leaves of trees in a thick forest area; height of the trees relative to the antenna heights of the cell-site and mobile unit.

A3.11 How is location of cell-site and mobile unit influenced by foliage loss

Close proximity of foliage at the cell-site transmitter always heavily attenuates the received signal strength and thus degrades the received signal quality. Therefore, the cell site must be placed away from dense trees. If the heavy foliage is close to the mobile unit, there may be significant degradation in the received signal quality, and additional signal loss due to foliage must be considered in the system design.

A3.12 What is Fraunhofer region of a transmitting antenna How is the far-field distance related with the largest physical linear dimension of the antenna

The Fraunhofer region or far-field region of a transmitting antenna is defined as the region beyond the far-field distance which is related to the largest linear dimension of the transmitter antenna aperture and the carrier wavelength. The far-field distance is directly proportional to the square of the largest physical linear dimension of the antenna.

A3.13 Why does signal level at the mobile unit reduce within a 1-km radius (near-in distance) of the cell-site in mobile radio communication systems
Within a 1-km radius (near-in distance) of the cell site in mobile radio communication systems, the signal reception at the mobile unit reduces due to narrow antenna beamwidth in the vertical plane of a high-gain omnidirectional antenna used at the cell-site. The large elevation angle caused by narrow beamwidth of the antenna causes the mobile unit to be in the shadow region (outside the main lobe of the radiation pattern of high-gain omnidirectional antenna). The larger the elevation signal, the weaker the reception level at the mobile unit located nearer to the cell-site due to the antenna's vertical radiation pattern.

A3.14 How does duct formation take place
In the troposphere region, the temperature increases with height rather than usual decrease of temperature at the rate of 6.5 degree C per km in the standard atmosphere. Within the troposphere region, the atmosphere has a dielectric constant slightly greater than unity due to high air density and decreases to unity at greater heights where the air density approaches to zero. Therefore, the electromagnetic waves are continuously refracted in the duct by the troposphere and reflected by the earth's surface. This results in the propagation of waves around the curvature of the earth far beyond the line-of-sight range. This phenomenon of long-distance propagation is termed duct propagation.

A3.15 What are the disadvantages of long-distance propagation How can it be minimised
The long-distance wave propagation due to tropospheric ducts may cause interference in frequency-reuse based cellular mobile communication systems, and occurrence of stronger signal levels at a particular location at one time but weaker signal levels at the same location. The interference can be minimised by using low-power transmitters and directional antennas at the cell-site.

A3.16 How can the value of path-loss exponents be determined experimentally
To determine the value of path-loss exponent in a given area, the receiver location is fixed, and the transmitter is placed at a number of locations varying in distances from the receiver. The received signal power or the path loss in dB is plotted against the distance on a logarithmic scale. The slope of the best-fit line through the measured values is taken as the path-loss exponent.

Self-Test Quiz

S3.1 The received signal power P_r is proportional to the distance between transmitter and receiver 'r', raised to an exponent γ, referred to as path loss exponent or distance-power gradient, as per the following expression:

(a) $P_r = P_0\, r^{-\gamma}$ (c) $P_r = P_0\, (1/r^{-\gamma})$
(b) $P_r = P_0\, r^{\gamma}$ (d) $P_r = P_0\, r^{2\gamma}$

where P_0 is the received signal strength at a reference distance, usually taken as one metre.

S3.2 In free-space propagation, (choose the most appropriate correct statement)
(a) there is no loss of energy as a radio wave propagates
(b) there is significant loss of energy as a radio wave propagates
(c) there is attenuation due to the spreading of the electromagnetic waves
(d) there is no attenuation due to the spreading of the electromagnetic waves

S3.3 A wireless communication transmitter has an RF power of 10 W and Tx antenna gain of 3 dB. The EIRP is
(a) 30 W (c) 10 W
(b) 3.33 W (d) 20 W

S3.4 Free-space propagation path loss is
(a) inversely proportional to frequency of transmission
(b) directly proportional to frequency of transmission

(c) independent of frequency of transmission
(d) directly proportional to square of the frequency of transmission

S3.5 There is a ______ dB per decade attenuation in signal strength as a function of distance in free-space propagation.
(a) 6 (c) 20
(b) 12 (d) 40

S3.6 In a wireless communication system, the transmitter and receiver stations are located at a distance of 10 km. The transmission delay of the radio signal is typically
(a) 33.3 microseconds (c) 3.33 nanoseconds
(b) 3.33 milliseconds (d) 33.3 milliseconds

S3.7 Using the point-to-point propagation prediction model, the path-loss slope will have a standard deviation of less than ______ only.
(a) 1 dB (c) 6−8 dB
(b) 2–3 dB (d) 10−12 dB

S3.8 In most of the mobile communication systems, the maximum distance between the cell-site and the mobile is at the most ______ (assuming the earth surface to be fairly flat).
(a) few metres
(b) few kilometres
(c) few tens of kilometres
(d) few hundreds of kilometres

S3.9 Assuming system parameters constant, the three parameters on which the propagation path loss in mobile point-to-point Lee model depend are
(a) frequency of transmission, *Tx* antenna height, and *Rx* antenna height
(b) frequency of transmission, distance between transmitter and receiver, and *Tx* antenna height
(c) frequency of transmission, distance between transmitter and receiver, and *Rx* antenna height
(d) distance between transmitter and receiver, *Tx* antenna height, and *Rx* antenna height

S3.10 The propagation path loss in a mobile radio environment increases by 40 dB for every increase in distance between the cell-site and mobile subscriber by ______ times.
(a) 2 (c) 8
(b) 4 (d) 10

S3.11 If the cell-site antenna height is doubled, there will be
(a) an increase in propagation path loss by 6 dB
(b) reduction in path loss by 6 dB
(c) reduction in path loss by 12 dB
(d) no change in path loss

S3.12 When an electromagnetic wave travels in free space, it suffers from
(a) absorption (c) refraction
(b) attenuation (d) super-refraction

S3.13 Radio waves in the UHF range normally propagate by means of
(a) space waves (c) ground waves
(b) sky waves (d) surface waves

S3.14 Electromagnetic waves are refracted when they
(a) encounter a perfectly conducting surface
(b) pass through a small slot in a conducting plane
(c) pass into a medium of different dielectric constant
(d) are polarised at right angles to the direction of propagation

S3.15 When microwave signals follow the curvature of the earth, this phenomenon is known as
(a) troposcatter (c) ionospheric reflection
(b) ducting (d) Faraday effect

S3.16 The main requirement for duct formation is ______ which is a result of temperature inversion in the atmosphere with height.
(a) multipath propagation (c) super refraction
(b) ionospheric reflection (d) troposcatter

Answers to Self-Test Quiz

S3.1 (a); S3.2 (c); S3.3 (d); S3.4 (d); S3.5 (c); S3.6 (a); S3.7 (b); S3.8 (c); S3.9 (d); S3.10 (d); S3.11 (b); S3.12 (b); S3.13 (a); S3.14 (c); S3.15 (b); S3.16 (c)

Review Questions

Q3.1 Why is propagation path loss one of the key parameters used in the analysis of radio wave propagation for mobile communication?

Q3.2 How are the service areas classified, depending on the following two criteria?
(a) Natural terrain along the propagation path
(b) Human-made structures along the propagation path

Q3.3 Explain the free-space propagation model and derive an expression for the received signal power. Make suitable assumptions as necessary.

Q3.4 Signal attenuation is greater for mobile communication than for free-space communication. State the various reasons.

Q3.5 In what ways is radio propagation on land different from that in free space?

Q3.6 Derive an expression for mobile point-to-point propagation model (two-ray model) to determine the received signal power. Explain the use of two-ray model to justify mobile radio path loss and antenna height effects.

Q3.7 Derive an expression for the received power and the phase difference between two fixed stations over water. Assume that the height of the transmitting and receiving antenna are h_t and h_r respectively and are located at heights H_t and H_r above the ground level.

Q3.8 Describe the parameters responsible for signal loss due to foliage. How does foliage loss vary with foliage density, path lengths and frequency of transmission?

Q3.9 Describe the effect on the received signal quality due to change of
(a) cell-site antenna height
(b) location of cell-site antenna
(c) effective antenna height with change of location of mobile unit

Assume that the mobile is driven up a positive slope (up to a high spot).

Q3.10 Differentiate between area-to-area prediction model and point-to-point propagation model used for estimating radio coverage in a mobile radio communication.

Q3.11 Explain how area-to-area prediction curves can be obtained. What role does 1-km intercept point and the path-loss slope play in obtaining the area-to-area prediction curves?

Q3.12 What are the different ways in which a signal can propagate over long distances? Explain what role does duct propagation play in reception of the mobile signal over very large distances. What are its causes?

Analytical Problems

P3.1 A wireless transmitter has an output power of 50 W. It is connected to its antenna by a coaxial cable that is 25 metres long and is properly matched. The signal attenuation in the coaxial cable is specified as 5 dB/100 m. The transmitting antenna has a gain of 8.5 dBi.
(a) How much power is available to the transmitting antenna?
(b) Compute the EIRP in the direction of maximum antenna gain.
(c) Calculate the power density at 1 km away from the transmitting antenna in the direction of maximum antenna gain, assuming free-space propagation.

P3.2 The transmission power of a cellular communication system operating at 900 MHz is 40 W. Assume free-space propagation conditions and 0 dB omnidirectional antennas at both cell-site transmitter and mobile receiver.
(a) Express the transmitter power in units of dBm.
(b) Compute the free-space propagation path loss in dB at a distance of 1 km.
(c) How much power is received by a mobile unit at a distance of 1 km?

P3.3 A wireless communication transmitter radiates 50 W of RF signal power.

(a) Express the transmitter power in units of dBm and dBW.
(b) If the transmitter power is applied to a unity gain antenna with a 900-MHz carrier frequency, what is the received power in dBm at a free-space distance of 100 m from the transmitter antenna?
(c) Compute the received power in dBm at a free-space distance of 10 km, taking received power calculated in (b) as reference.

P3.4 If cell-site transmitter power is 10 W, frequency of transmission is 900 MHz, determine the received signal power in watts at a distance of 1 km in free space. Assume 0 dB omnidirectional antennas at the cell-site as well at the mobile receiver.

P3.5 Determine the received signal power by a mobile at a distance of 10 km from a 50-W cell-site transmitter operating at a carrier frequency of 1900 MHz. The *Tx* antenna gain is unity and *R*x antenna gain is 2. Assume the free-space propagation condition.

P3.6 What is the coverage of a cell-site that transmits 100 W of power, given that the receiver sensitivity is -100 dBm, the path loss at the first metre is 32 dB, and the path loss gradient is 4?

P3.7 Consider a wireless communication transmitter antenna radiating a power of 5 W at 900 MHz. Calculate the received signal power at a distance of 2 km if propagation is taking place in free space.

P3.8 A cellular mobile communication system operates at 900 MHz. Compute the propagation path loss at a distance 5 km away from the cell-site. Assume the height of the cell-site transmitting antenna is 50 m and the mobile receiving antenna is 1.5 m above ground.

P3.9 If the mobile received power at a reference distance $r_o = 1$ km is equal to -30 dBm, find and tabulate the received powers at distances of 2 km, 5 km, 10 km, and 20 km from the same base-station transmitter for the following path-loss models:
(a) Free space model
(b) Propagation model with path-loss exponent, $\gamma = 3$
(c) Propagation model with path loss exponent, $\gamma = 4$
(d) Two-ray ground reflection point-to-point Model
(e) Hata propagation model
Assume $f_c = 900$ MHz, $P_t = +40$ dBm, $h_t = 40$ m, $h_r = 2$ m, $G_t = G_r = 1$. Comment on the difference between the results obtained from these propagation models.

P3.10 Determine the propagation path loss of a 900-MHz cellular system operating in a large city from a base station with the cell-site antenna height of 100 m and mobile unit installed in a vehicle with an antenna height of 2 m, when the distance between the mobile unit and the base station is 4 km.

P3.11 Use the Hata propagation model to determine the maximum radii of cells at 900 MHz and 1900 MHz respectively, having a maximum acceptable path loss of 130 dB. Assume a cell-site antenna height of 50 m and a mobile antenna height of 2 m.

P3.12 The measured received signal power at 10 metres away from a transmitter using an isotropic antenna is 2 mW. Assume the receiver sensitivity of -100 dBm and path loss exponent of 2.9. Determine the coverage range if an additional 10 dB margin is needed to compensate for shadow loss over and above the median path loss.

P3.13 Consider a 100 mW transmitter and free-space propagation between transmitting and receiving isotropic antennas. A commercial mobile receiver is used for data transmission with a specified receiver sensitivity of -90 dBm. Calculate the radius of the service area at a transmission frequency of 800 MHz.

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Cellular communication is designed to enhance the spectrum efficiency as well as the system capacity while maintaining the desired signal quality. The main principle of cellular communication is to divide a large geographical area into a number of contiguous smaller geographical coverage areas called cells, each one of which is served by its own cell-site or low-power base station located at its centre. Cells constitute the design of the heart of cellular systems. The focus in this chapter is to understand the essential principles of cellular communication, and the formation of regular hexagonal cellular structures with multiple clusters. The most serious concern due to frequency reuse is cochannel interference which may degrade the performance of a cellular system operation. Finally, a brief overview of various methods employed to reduce cochannel interference is also given.

4

Principles of Cellular Communication

4.1 CELLULAR TERMINOLOGY

A cell is the basic geographic unit of a cellular system. A cell is the radio area covered by a cell-site that is located at its centre. In other words, the radio coverage by one base station or a cell-site is referred to as a cell, which is also called a *footprint*. In a cellular system, the most important factor is the size and shape of a cell. Because of constraints imposed by natural irregular terrain, man-made structures, and non-uniform population densities, the actual shape of the cell may not be either a circle or a regular geometrical shape but may be a little distorted. For proper analysis and evaluation of a cellular system, an appropriate model of a cell shape is needed. Figure 4.1 depicts ideal cell, actual cell and possible cell models such as equilateral triangle, square, and hexagon that represent a cell boundary with a radius R from the centre of the cell.

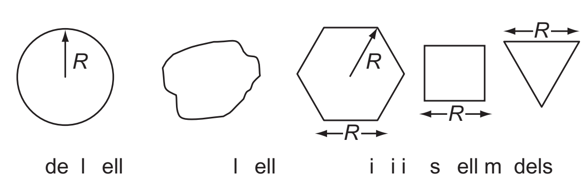

Fig. 4.1 | *Ideal, actual and fictitious cell models*

The actual shape of the cell is determined by the desired received signal level by the mobile subscribers from its base-station transmitter in its operating area. The received signal is affected by many factors including reflections, refractions, and contour of the terrain as well as multipath propagation due to presence of natural and man-made structures.

A cell is not a perfect polygon. So real footprints are vague in nature. On the other hand, cellular layouts using irregular structures limit growth and are also inefficient. For this reason, cellular layouts and performance studies are based on regular topologies as they allow the systematic growth though they may be just conceptual.

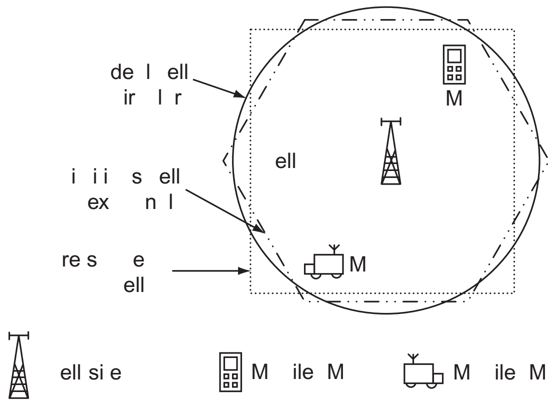

Fig. 4.2 *Illustration of a cell with a CS and Mobile*

The base station, also called Cell-Site (CS), located approximately at its centre, serves all mobile users in the cell. Figure 4.2 illustrates an ideal cell area (circular), a hexagonal cell area (used in most models), and a square cell area (an alternative shape) with a cell-site at its centre and a number of mobile units (M) within the cell area.

The shape of the cell can be circular around the cell-site transmitting tower under ideal radio environment. The periphery of the circle is equal to the acceptable received signal level from the transmitting signal. It means that if the cell-site is located at the centre of the cell, the cell area and periphery are determined by the signal strength within the region. This depends on many factors, such as the height of the cell-site transmitting antenna; contour of the terrain; presence of tall buildings, hills, valleys, vegetation; and atmospheric conditions. Therefore, the actual shape of the cell may be a zigzag shape which indicates a true radio coverage area. However, for all practical purposes, a regular hexagonal geometry shape approximates the cell boundary, which is a good approximation of a circular region. However, the square is another alternative shape that can be used to represent the cell area.

4.2 CELL STRUCTURE AND CLUSTER

In practice, cells are of arbitrary shape which is quite close to a circle, is the ideal radiation pattern of an omnidirectional antenna. Because of the randomness inherent nature of the mobile radio propagation and irregular geographical terrain, it is easier to obtain insight and plan the cellular network by visualising all the cells as having the same shape. By approximating a uniform cell size for all cells, it is easier to analyse and design a cellular topology mathematically. It is highly desirable to construct the cellular system such that the cells do not overlap, and are tightly packed without any dead signal spots. The cellular topology formed by using ideal circular shape results into overlaps or gaps between them which is not desirable in cellular communications which has to be essentially continuous. This form of layout requires the use of regular topologies (say, a hexagonal topology) instead of a circular shape, as depicted in Fig. 4.3.

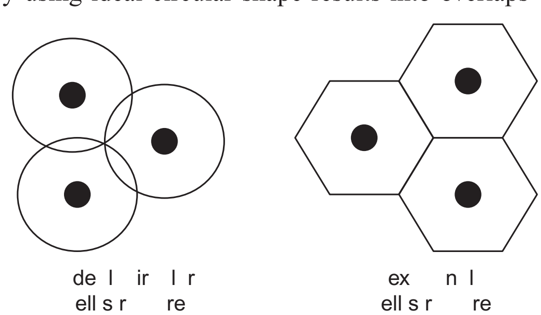

Fig. 4.3 *The ideal and regular hexagonal cell structure*

In Fig. 4.3, the middle dark circles represent cell-sites. This is where the base-station radio equipment and their antennas mounted on tall towers are located. A cell-site gives radio signal coverage to a cell. In other words, the cell-site is a location or a point at the centre of the cell, whereas the cell is a wide geographical service area.

The design and performance of cellular systems using regular geometrical topologies may not correspond to real mobile environments, but these topologies do provide valuable information and guidelines for structuring practical cellular configuration layouts. Cells of the same shape form a tessellation so that there are no ambiguous areas that belong to multiple cells or to no cell. The cell shape can be of only three types of regular polygons: equilateral triangle, square, or regular hexagon as shown in Fig. 4.4.

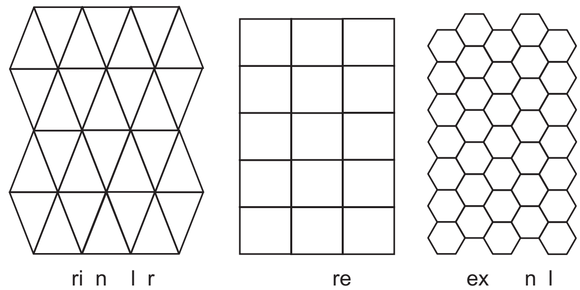

Fig. 4.4 *Illustration of possible geometrical cellular structures*

A cellular structure based on a regular hexagonal topology, though fictitious, offers best possible non-overlapped cell radio coverage. Traditionally, a regular hexagonal-shaped cell is the closest approximation to a circle out of these three geometrical shapes and has been used for cellular system design. In other words, for a given radius (largest possible distance between the polygon centre and its edge), the hexagon has the largest area. Moreover, it allows a larger region to be divided into nonoverlapping hexagonal subregions of equal size, with each one representing a cell area. Octagons and decagons geometrical patterns do represent shapes closer to a circular area as compared to hexagons, but they are not used to model a cell as it is not possible to divide a larger area into non-overlapping subareas of the same size.

A mobile radio communication system is generally required to operate over areas too large to be economically covered by a single cell-site. Therefore, several or many widely spaced transmitter sites are required to provide total area coverage. The spacing between the base stations need not be regular and the cell or the area served by a base station need not have any particular shape. However, the absence of an orderly geometrical structure makes the system design more difficult and results in inefficient use of spectrum and uneconomical deployment of equipment. The propagation considerations recommend the circle as a cell shape for defining the area covered by a particular base station. This is impracticable for design purpose, since there could be areas which are contained either in no cell or in multiple cells. On the other hand, any regular polygon can cover the service area with no gaps or overlaps. The regular hexagonal shape results in the most economical system layout design.

In most modeling, simulation, measurements, and analysis of interference in cellular systems, hexagons are used to represent the cell structure. A hexagon is closer to a circular area and multiple hexagons can be arranged next to each other, without having an overlapping area or uncovered space in between. In other words, the hexagonal-shaped cells fit the planned area nicely, with no gap and no overlap among the adjacent hexagonal cells. Thus, it simplifies the planning and design of a cellular system.

Facts to Know!

Cells are always drawn as hexagons because it makes it simpler and easier to show adjacent cells without any overlap. In reality, the cell shape is closer to a circle but it may be affected by surrounding buildings and other geographic features.

EXAMPLE 4.1 | Significance of cellular topology

Consider a single high-power transmitter that can support 40 *voice channels over an area of* 140 *km² with the available spectrum. If this area is equally divided into seven smaller areas (cells), each supported by lower power transmitters so that each cell supports* 30% *of the channels, then determine*

(a) coverage area of each cell

(b) total number of voice channels available in cellular system

Comment on the results obtained.

Solution

Total service area to be covered = 140 km (given)
Total number of channels available = 40 (given)
Number of cells = 7 (given)

***(a)** To determine coverage area of each cell*

***Step 1.** Coverage area of a cell = Total service area / Number of cells*

Hence, coverage area of a cell = 140 km / 7 = 20 km

***(b)** To determine total number of voice channels available in the cellular system*

***Step 2**. Number of voice channels per cell = 30% of original channels (given)*

Number of voice channels per cell = $0.3 \times 40 = 12$ channels/cell

Total number of voice channels available in cellular system is given by the number of channels per cell multiplied by the number of cells in the service area.

Hence, total number of voice channels = $12 \times 7 = 84$ channels

Comment on the results

- Thus, there is a significant increase in the number of available channels (84 channels as calculated above) in a given cellular system as compared to a non-cellular system (40 channels as given).
- This means the system capacity is increased.
- However, care has to be taken in allocation of channels to various cells in such a way so as to prevent interference between the channels of one cell and that of another cell.
- Adjacent cells should not be allocated the same channels, whereas cells located far apart can be allocated the same channels using frequency reuse scheme.

A Cellular Cluster A group of cells that use a different set of frequencies in each cell is called a cellular cluster. Thus, a cluster is a group of cells with no reuse of channels within it. It is worth mentioning here that only a selected number of cells can form a cluster. It follows certain rules before any cell can be repeated at a different location.

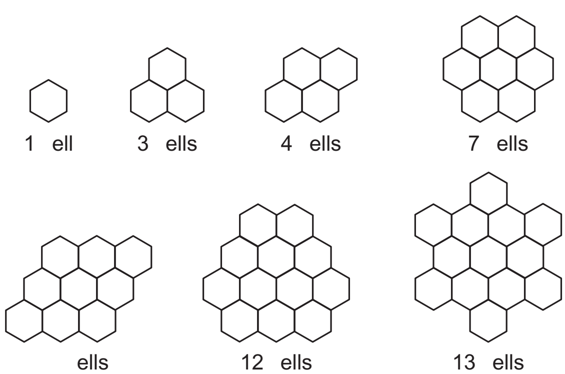

Fig. 4.5 *Common reuse patterns of hexagonal cell clusters*

Some common reuse cluster patterns are given in Fig. 4.5.

Two or more different cells can use the same set of frequencies or channels if these cells are separated in space such that the interference between cells at any given frequency is at an acceptable level. That means, the cluster can be repeated any number of times in a systematic manner in order to cover the designated large geographical service area. Let there be K number of cells having a different set of frequencies in a cluster. Then K is termed as the cluster size in terms of the number of cells within it.

EXAMPLE 4.2 | Number of clusters

Calculate the number of times the cluster of size 4 have to be replicated in order to approximately cover the entire service area of 1765 km^2 with the adequate number of uniform-sized cells of 7 km^2 each.

Solution

Size of the cluster, $K = 4$ (given)
Area of a cell, $A_{cell} = 7$ km^2 (given)

Step 1. *To determine area of the cluster*

Area of a cluster, $A_{cluster} = K \times A_{cell}$

Therefore, $A_{cluster} = 4 \times 7 \text{ km}^2 = 28 \text{ km}^2$

Step 2. *To determine number of clusters in the service area*

Total service area, $A_{system} = 1765 \text{ km}^2$ (given)

Number of clusters in service area $= A_{system} / A_{cluster}$

Number of clusters in service area $= 1765 \text{ km}^2 / 28 \text{ km}^2$

Number of clusters in service area $= 63$

Hence, the number of times the cluster of size 4 has to be replicated is 63.

Each cell size varies depending on the landscape. Typical size of a cell may vary from a few 100 metres in cities (or even less at higher frequencies) to several kilometres on the countryside. Smaller cells are used when there is a requirement to support a large number of mobile users, in a small geographic region, or when a low transmission power may be required to reduce the effects of interference. So typical uses of small cells are in urban areas, low transmission power required, or higher number of mobile users.

It is clear that if the cell area is increased, the number of channels per unit area is reduced for the same number of channels and is good for less populated areas, with fewer mobile users. Generally, large cells are employed in remote areas, coastal regions, and areas with few mobile users, large areas that need to be covered with minimum number of cell-sites. It may also be noted that the cell area and the boundary length are important parameters that affect the handoff from a cell to an adjacent cell. A practical solution for optimum cell size is to keep the number of channels per unit area comparable to the number of mobile subscribers to be served within that cell.

> **Facts to Know!**
>
> The cell radius is mainly determined by the cell-site transmitter power and cell-site antenna height; these two parameters are decided by the system design engineer. Therefore, it is his responsibility to estimate how many radio channels (and hence system capacity) would be created through frequency reuse.

EXAMPLE 4.3 Cell size and system capacity

(a) Assume a cellular system of 32 *cells with a cell radius of 1.6 km, a total spectrum allocation that supports* 336 *traffic channels, and a reuse pattern of 7. Calculate the total service area covered with this configuration, the number of channels per cell, and a total system capacity. Assume regular hexagonal cellular topology.*

(b) Let the cell size be reduced to the extent that the same area as covered in Part (a) with 128 *cells. Find the radius of the new cell, and new system capacity.*

Comment on the results obtained.

Solution

(a) *To calculate total service area, number of channels per cell, and system capacity*

Total number of cells in service area $= 32$ (given)

Radius of a cell, $R = 1.6$ km (given)

Step 1. *To calculate area of a regular hexagonal cell*

Area of a regular hexagonal cell, $A_{cell} = 3\sqrt{3}/2 \times R^2$

Therefore, $A_{cell} = 3\sqrt{3}/2 \times (1.6 \text{ km})^2 = 6.65 \text{ km}^2$

Step 2. *To calculate total service area*

Total service area covered $=$ no. of cells in total area $\times$ Area of a cell

Hence, total service area covered $= 32 \times 6.65 = 213 \text{ km}^2$

Step 3. *To calculate number of channels per cell*

Total number of available traffic channels = 336 (given)

Frequency reuse pattern (cluster size) = 7 (given)

Hence, number of channels per cell = 336/7 = 48

Step 4. *To calculate total system capacity*

Total system capacity = number of channels per cell × number of cells

Hence, total system capacity = 48 × 32 = 1536 channels

(b) *Total number of available cells = 128 (given)*

Total service area = 213 km^2 (as calculated in Step 2)

Step 5. *To determine area of new regular hexagonal cell*

Area of a regular hexagonal cell = total service area / number of cells

$$= 213 \text{ km}^2 / 128 = 1.66 \text{ km}^2$$

Step 6. *To find radius of new smaller cell, R*

Area of a regular hexagonal cell = $3\sqrt{3}/2 \times R^2$

But, $3\sqrt{3}/2 \times R^2 = 1.66 \text{ km}^2$ (as calculated in Step 5)

Or, $R = 0.8$ km

Hence, radius of new smaller cell $R = 0.8$ km

Step 7. *To find new system capacity*

New system capacity = number of channels per cell × number of cells

New system capacity = 48 × 128

Hence, new system capacity = 6144 channels

Comment on the results It is observed that as the number of cells are increased from 32 to 128 to cover the same service area (213 km^2), the size of the cell (in terms of radius R) is decreased from 1.6 km to 0.8 km. Keeping the identical number of channels (48) per cell, total system capacity is significantly increased from 1536 channels to 6144 channels. Hence, cell size is one of the major factors to determine the system capacity for a given number of frequency channels allocated to serve the designated area.

4.3 FREQUENCY REUSE CONCEPT

What is the essence of cellular communication? As outlined previously, if a single base station serves a wireless communication system, a high power transmitter is needed to support a large number of users. Moreover, due to availability of limited RF spectrum, the maximum number of simultaneous users in this system is also limited. If allocated RF spectrum or a given set of frequencies (frequency channels) can be reused in a given large geographical service area without increasing the interference then the service area can be divided into a number of small areas called cells, each allocated a subset of frequencies. With smaller area coverage, lower power transmitters with lower height antennas can be used at a base station.

The conventional radio communication systems are faced with the problems of limited service area capability and inefficient spectrum utilisation. This is because these systems are usually designed for providing service in an autonomous geographic zone and by selecting RF channels from a specified allocated frequency band. Contrary to this, the present mobile radio communication system are designed for wide area coverage and high grade of service. At the same time, the systems are required to provide continuous communication through an effective usage of available spectrum. This dictates that the mobile radio network design must satisfy the objective of providing continuous and wide service area coverage while optimally using the RF spectrum.

The increase in system capacity is achieved with the use of smaller cells, reuse of frequencies, and cell sectoring. Frequency reuse is the core concept of the cellular communications. The design process of selecting and allocating channel groups for all the cellular base stations within a system is called *frequency reuse*. Thus, large coverage area, efficient spectrum utilisation and enhanced system capacity are the major attributes of cellular communication. However, this requires proper system design and complex operation of the cellular mobile system working in a hostile mobile propagation environment and system interference in order to ensure the desired service performance.

> **Facts to Know !**
>
> An n-cell frequency reuse scheme allows only 1/n of the total number of channels to be available in each cell, which greatly increases the probability of blocking for a user trying to access the system.

In a mobile radio network designed on the basis of frequency reuse concept, it must be ensured that the service area is adequately protected from the cochannel and the adjacent-channel interference. The carrier-to-interference ratio (*C/I*) requirements are considerably lower for digital systems as compared to analog systems. It is seen that spectrum efficiency increases if the *C/I* value is lowered. This is due to the fact that lowering the acceptable value of *C/I* reduces the frequency reuse distance and the reuse pattern. The cochannel interference can be controlled by geographical separation whereas adjacent-channel interference depends on the receiver filter characteristics and out-of-band transmission.

A regular geometrical hexagonal pattern results in obtaining optimum area coverage and efficient spectrum utilisation. The minimum value of cluster size provides optimum spectrum occupancy. However, in actual design, due to physical limitations the location of base stations cannot follow the regular geometrical hexagonal pattern. The resultant location errors distort the regular pattern, thereby causing serious interference problems.

> **Facts to Know !**
>
> The plan of dividing the large geographic service area into many small contiguous cells and using a low-power transmitter with low antenna as base station in each cell is referred to as cellular communications.

Mobile users communicate only via the base stations. Each cell is allocated a finite number of Radio Frequency (RF) channels, depending upon the number of simultaneous users required to be served. This enables the cells that are located sufficiently physically apart to reuse the same set of frequencies, without causing cochannel interference. However, each adjacent cell within a cluster operates on different frequencies to avoid interference.

Cells, which use the same set of frequencies, are referred to as *cochannel cells*. The space between adjacent cochannel cells is filled with other cells that use different frequencies to provide frequency isolation.

A typical cluster of seven cells, each repeated seven times with frequency reuse, is illustrated in Fig. 4.6.

If the system is not properly designed, cochannel interference may occur due to the simultaneous use of the same channel. This is the major concern in frequency reuse. Specifically, if the available channels are reused for additional traffic, it is possible to

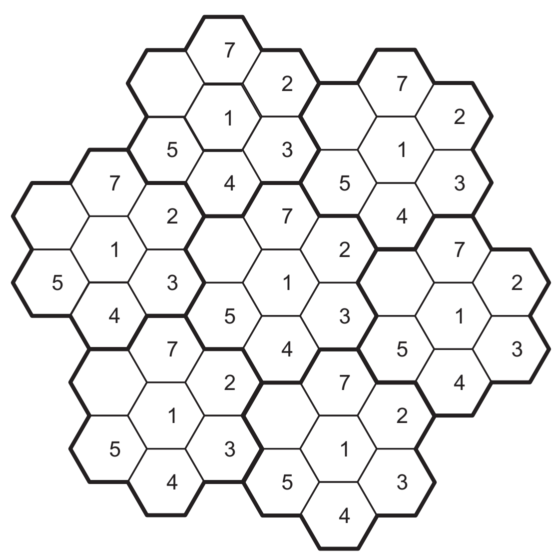

Fig. 4.6 *Illustration of frequency reuse (Fx: A set of frequency channels)*

Facts to Know!

Frequency channels are allocated to each cell by means of an intelligent frequency-planning technique so as to minimise the cochannel and adjacent channel interference while meeting the performance requirements both in terms of received signal quality as well as traffic capacity in these cells.

serve more number of users, thereby increasing the system capacity within allocated RF spectrum, and hence enhancing spectrum efficiency as well.

The total number of channels available in a cellular system is finite because of limited RF spectrum allocation. The capacity of a cellular system is defined by the total number of channels available, which depends on how the available channels are deployed. So, the total number of available channels without frequency reuse, N, is the allocated RF spectrum band divided by the number of RF channels having equal channel bandwidth.

EXAMPLE 4.4 | Frequency reuse and spectrum efficiency

Consider a single high-power transmitter that can support 100 voice channels covering a given service area. Let the service area be divided into seven smaller areas (cells) as shown in Fig. 4.7, each supported by lower power transmitters. The available spectrum of 100 voice channels is divided into 4 groups of 25 channels each. The cells (1, 7), (2, 4), (3, 5), and 6 are assigned distinct channel groups. Show that the total number of channels that can be supported is enhanced to 175 to cover the same service area. Comment on the results obtained.

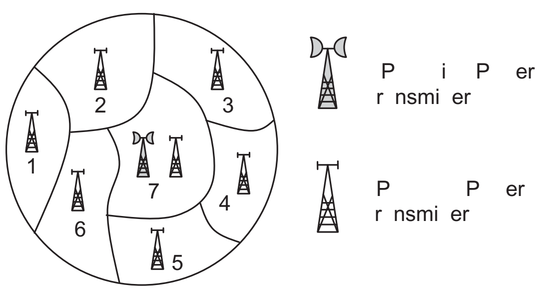

Fig. 4.7 | *Illustration for frequency reuse and spectrum efficiency*

Solution

Total number of channels available, $N = 100$ (given)

Case 1. *When a single high-power transmitter is used to cover the given service area*

This implies that it is a non-cellular system.

Hence, total number of channels in the system are limited to 100 only.

Case 2. *When the service area is divided into seven cells*

Number of distinct cells $= 7$ (given)

Number of channel groups $= 4$ (given)

Number of channels per channel group $= 25$ (given)

Step 1. *Allocation of channel groups to cells*

Let channel group 1 be allocated to cells 1 as well as 7; channel group 2 be allocated to cells 2 as well as 4; channel group 3 be allocated to cells 3 as well as 5; and channel group 4 be allocated to cell 6 (refer given Fig. 4.7).

Step 2. *Total number of channels available in the specified cellular system*

Total number of channels allocated to all cells is equal to the number of channels per channel group multiplied by the number of distinct cells. That is,

Total number of channels allocated to all cells $= 25 \times 7$

Hence, total number of channels available $= 175$ channels

Comment on the results It is seen from the above example that the total number of channels that can be supported by the given cellular system is increased to 175 from 100 in a non-cellular system to cover the same service area. Hence, it can be concluded that

> 'The theoretical coverage range and capacity of a cellular communication system are unlimited, with optimum use of RF spectrum utilisation.'
>
> 'The frequency reuse can drastically increase the spectrum efficiency, thereby increasing the system capacity.'

However, there is a need to address the following technical issues for proper design and planning of a cellular network:

- Selection of a suitable frequency reuse pattern
- Physical deployment and radio-coverage modeling
- Plans to account for the expansion of the cellular network
- Analysis of the relationship between the capacity, cell size, and the cost of the infrastructure

The reason for the complexity of the cellular system is, of course, frequency reuse. Once a mobile moves out of the radio coverage of a cell, the channel pair it occupied for duplex communication link is now available for another communication link in that cell. By making cells smaller, frequencies can be reused at shorter distances. Typically, once the radius drops below about 0.5 km, the hand-offs occur so frequently that it is difficult to cope with a mobile moving at high speed. The flexibility of cell sizes allows for larger cells in less-developed areas and smaller cells in areas of higher traffic.

Facts to Know!

In the first instance, it seems that there is no theoretical limit to this (smaller cells, more frequency reuse and thus higher capacity), but there are practical limits. As cells become smaller, more cell sites are needed and hand-offs occur more frequently, requiring more computing power and faster response both at the system level and in the individual mobile phone.

EXAMPLE 4.5 | Frequency reuse and system capacity

A mobile communication system is allocated RF spectrum of 25 MHz and uses RF channel bandwidth of 25 kHz so that a total number of 1000 voice channels can be supported in the system.

(a) If the service area is divided into 20 cells with a frequency reuse factor of 4, compute the system capacity.

(b) The cell size is reduced to the extent that the service area is now covered with 100 cells. Compute the system capacity while keeping the frequency reuse factor as 4.

(c) Consider the cell size is further reduced so that the same service area is now covered with 700 cells with the frequency reuse factor of 7. Compute the system capacity.

Comment on the results obtained.

Solution

Number of available voice channels, $N = 1000$ (given)

Step 1. *To determine the cluster capacity*

We know that in a cellular system based on frequency-reuse concept, all the given available channels, that is, 1000, are allocated to each cluster uniformly.

Therefore, each cluster can serve 1000 active users simultaneously.

In other words, the capacity of a cluster = 1000

(a) *To compute the system capacity for given K*

Number of cells covering the area = 20 (given)

Frequency reuse factor or cluster size = 4 (given)

Step 2. *To determine number of clusters*

Number of clusters = number of cells/cluster size

Therefore, number of clusters = 20/4 = 5

Step 3. *To determine the system capacity*

The capacity of a cluster = 1000 (as calculated is Step 1)

Number of clusters = 5 (as calculated is Step 2)

Thus, number of channels in all 5 clusters = 1000 × 5 = 5000
Hence, the system capacity = 5000 users

(b) *To compute new system capacity for increased number of cells.*
Number of cells covering the area = 100 (given)
Frequency reuse factor or cluster size = 4 (given)

Step 4. *To determine number of clusters*
Therefore, number of clusters = 100/4 = 25

Step 5. *To determine new system capacity*
Thus, number of channels in all 25 clusters = 1000 × 25 = 25000
Hence, the new system capacity = 25000 users

(c) *To compute new system capacity for increased number of cells and cluster size.*
Number of cells covering the area = 700 (given)
Frequency reuse factor or cluster size = 7 (given)

Step 6. *To determine number of clusters*
Therefore, number of clusters = 700/7 = 100

Step 7. *To determine new system capacity*
Thus, number of channels in 100 clusters = 1000 × 100 = 100,000
Hence, the new system capacity = 100 000 users

Comments on the results It is observed that as the number of cells covering a given service area is increased, the number of clusters having all available number of channels increases. This results into significant increase in the number of active users in the system or the system capacity. Hence, it is concluded that frequency reuse enhances system capacity.

4.4 CLUSTER SIZE AND SYSTEM CAPACITY

The K number of cells in the cluster would utilise all N available channels. In this way, each cell in the cluster contains N/K number of channels only.

Alternately, the total number of channels available in a cluster, N is equal to the number of channels per cell ($J \leq N$) multiplied by the number of cells per cluster (K), that is,

$$N = J \times K \tag{4.1}$$

In a cellular system, the whole geographical area where the cellular services are required to be provided is divided into a number of clusters having a finite number of cells. The K cells in a cluster use the complete set of available frequency channels.

Since N is the total number of available channels, it can be seen that a decrease in the cluster size K is accompanied by an increase in the number of channels J allocated per cell. Thus, by decreasing the cluster size, it is possible to increase the capacity per cell.

The cluster can be replicated many times to cover the desired geographical area by a cellular communication system. The overall system capacity, C, can then be theoretically determined by simply multiplying the number of clusters in a system (say M) with total number of channels allocated to a cluster, N, i.e.,

$$C = M \times N \tag{4.2}$$

Using the relationship $N = J \times K$, we get

$$C = M \times J \times K \tag{4.3}$$

If K is decreased and J is proportionally increased so that $C = M \times J \times K$ is satisfied, it is necessary to replicate the smaller cluster more times in order to cover the same geographical service area.

This means the value of M has to be increased. Since $J \times K$ $(=N)$ remains constant and M is increased, it shows that the system capacity C is increased. That is, when K is minimised, C is maximised. But minimising K will increase cochannel interference.

EXAMPLE 4.6 | Cellular system capacity

Consider that a geographical service area of a cellular system is 4200 km². A total of 1001 radio channels are available for handling traffic. Suppose the area of a cell is 12 km².

(a) How many times would the cluster of size 7 have to be replicated in order to cover the entire service area? Calculate the number of channels per cell and the system capacity.

(b) If the cluster size is decreased from 7 to 4, then does it result into increase in system capacity? Comment on the results obtained.

Solution

Service area of a cellular system, $A_{sys} = 4200 \text{ km}^2$ (given)

Coverage area of a cell, $A_{cell} = 12 \text{ km}^2$ (given)

Total number of available channels, $N = 1001$ (given)

(a) *To calculate number of clusters, cell capacity, and system capacity*

Cluster size, $K = 7$ (given)

Step 1. *To calculate the coverage area of a cluster*

The coverage area of a cluster, $A_{cluster} = K \times A_{cell}$

Therefore, $A_{cluster} = 7 \times 12 \text{ km}^2 = 84 \text{ km}^2$

Step 2. *To calculate the number of clusters*

The number of times that the cluster has to be replicated to cover the entire service area of cellular system $= A_{sys} \quad A_{cluster}$

Or, number of clusters, $M = 4200 \quad 84$

Hence, number of clusters, $M = 50$ clusters

Step 3. *To calculate cell capacity*

Since total number of available channels are allocated to one cluster, therefore, the number of channels per cell, $J = N \quad K$

Or, cell capacity, $J = 1001 \quad 7$

Hence, cell capacity, $J = 143$ channels/cell

Step 4. *To calculate system capacity*

The system capacity, $C = N \times M$

Or, system capacity, $C = 1001 \times 50$

Hence, the system capacity, $C = 50\,050$ channels

(b) *To calculate new system capacity for reduced K*

New cluster size, $K = 4$ (given)

Step 5. *To calculate the coverage area of a new cluster*

The coverage area of a cluster, $A_{cluster} = K \times A_{cell}$

Therefore, $A_{cluster} = 4 \times 12 \text{ km}^2 = 48 \text{ km}^2$

Step 6. *To calculate increased number of clusters*

The number of times that the cluster has to be replicated to cover the entire service area of a cellular system $= A_{sys} \quad A_{cluster}$

Or, number of clusters, $M = 4200 \quad 48$

Hence, number of clusters, $M = 87$ (approx.)

Step 7. *To calculate new system capacity*

The system capacity, $C = N \times M$

Or, system capacity, $C = 1001 \times 87$

Hence, the system capacity, C = 87 000 channels

Comments on the results From (a) and (b) above, it is seen that for decrease in cluster size from 7 to 4 results into an increase in number of clusters from 50 to 87 for a given service area. The system capacity is increased from 50,050 channels to 87,000 channels. Therefore, decreasing the cluster size does increase the system capacity. However, the average signal-to-cochannel interference also increases which has to be kept at an acceptable level in order to achieve desirable signal quality.

Assume that the cell size is kept constant and a fixed spectrum per cluster is allocated. Then more number of cells per cluster (that is, higher value of K) means

- Fewer channels per cell
- Less system capacity
- Less cochannel interference (cochannel cells farther apart)

And less number of cells per cluster (that is, lower value of K) means

- More channels per cell
- More system capacity
- More cochannel interference (cochannel cells closer together)

So it is desirable to choose reuse factor K to maximise capacity per area subject to interference limitations.

EXAMPLE 4.7 | Cluster and system capacity

A cellular communication service area is covered with 12 *clusters having* 7 *cells in each cluster and* 16 *channels assigned in each cell. Show that*

(a) the number of channels per cluster are 112

(b) the system capacity is 1344

Solution

Number of clusters in the service area = 12 (given)

Number of cells in a cluster = 7 (given)

Number of channels in a cell = 16 (given)

(a) *To determine the number of channels per cluster*

Number of channels in a cluster is given by the number of cells in a cluster multiplied by the number of channels in a cell, that is,

Number of channels in a cluster = 7×16

Hence, number of channels per cluster = 112 channels/cluster

(b) *To determine the system capacity*

The system capacity is given by the number of clusters in a given area multiplied by the number of channels in a cluster, that is,

Number of channels in the system = 112×12

Hence, the system capacity = 1344 channels/system

4.5 METHOD OF LOCATING COCHANNEL CELLS

Cells, which use the same set of frequencies or channels, are referred to as cochannel cells. The two shift parameters *i* and *j* can be used to determine the location of the nearest cochannel cell in a hexagonal geometry where *i* and *j* are separated by 60 degrees, as shown in Fig. 4.8. The shift parameters *i* and *j* can have any integer value 0, 1, 2, 3, and so on.

To locate the nearest cochannel cells (neighbouring cells or cells in the first tier), mark the centre of the cell as (0, 0) for which cochannel cells are required to be located. Define the unit distance as the distance of centres of two adjacent cells. Now follow the two steps mentioned below:

Step 1 Move *i* number of cells along any chain of hexagons.
Step 2 Turn 60 degrees counterclockwise and move *j* number of cells.

The method of locating cochannel cells in a cellular system using the preceding procedure is shown in Fig. 4.9 for $i = 3$ and $j = 2$, where cochannel cells are shaded. The parameters *i* and *j* measure the number of nearest neighbouring cells between cochannel cells.

Example 4.8 Relationship between cluster size K and shift parameters i, j

Illustrate and prove that for a regular hexagonal geometry, the cell cluster size is given by the relationship

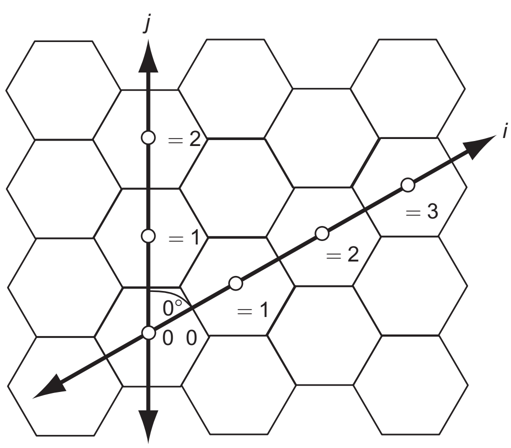

Fig. 4.8 *Shift parameters* i *and* j *in a hexagonal geometry*

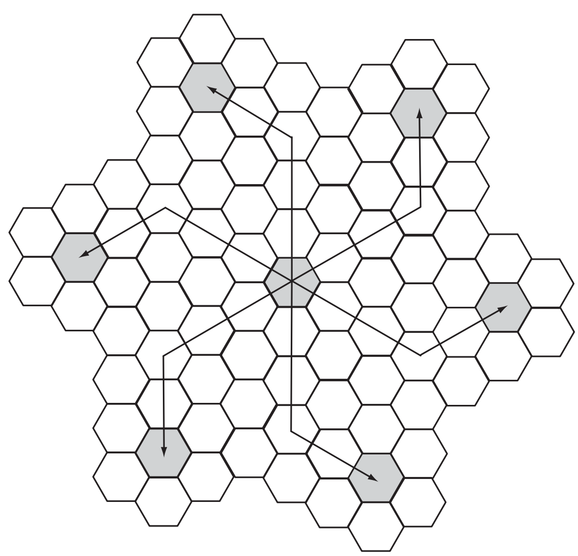

Fig. 4.9 *Locating cochannel cells in a cellular system for* i =3, j = 2. *Shaded cells are cochannel cells*

$$K = i^2 + j^2 + i \times j$$

where K = number of cells per cluster or cluster size
i = number of cells (centre to centre) along any chain of hexagon
j = number of cells (centre to centre) in 60° counterclockwise of i

Solution

Refer Fig. 4.10. Let 'R' be the distance from the centre of a regular hexagon and any of its vertex. A regular hexagon is one whose sides are also equal to 'R'. Let 'd' be the distance between the centres of two adjacent regular hexagons.

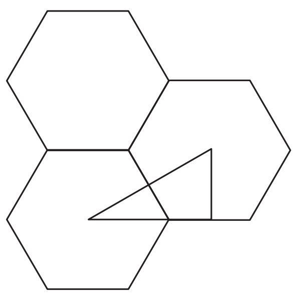

Fig. 4.10 *Distance between two adjacent cells*

Step 1. *To show that* $d = \sqrt{3}R$

From the geometry of the figure, $OA = R$ and $AB = R/2$.

Then, $OB = OA + AB = R + R/2 = 3\ R/2$.

Then, in right-angled $\Delta\ OAP$,

$OP = OA \sin 60^\circ = (\sqrt{3}/2)R$

Let the distance between the centres of two adjacent hexagonal cells, OQ, be denoted by 'd'. Then,

$OQ = OP + PQ$ (where $OP = PQ$)

Therefore, $d = (\sqrt{3}/2)R + (\sqrt{3}/2)R$

Hence, $d = \sqrt{3}R$ (4.4)

Step 2. *Area of a small hexagon,* $A_{small\ hexagon}$ *w*

The area of a cell (the small hexagon) with radius R is given as

$A_{small\ hexagon} = (3\sqrt{3}/2) \times R^2$ (4.5)

Step 3. *Procedure of locating cochannel cells*

A cell has exactly six equidistant neighbouring (nearest or first tier) cells, formed by following the procedure of locating cochannel cells, corresponding to six sides of the hexagon. That is,

- firstly moving i number of cells along i axis,
- secondly, turning 60 degrees counterclockwise, and
- then moving j number of cells along j axis, as shown in Fig. 4.11.

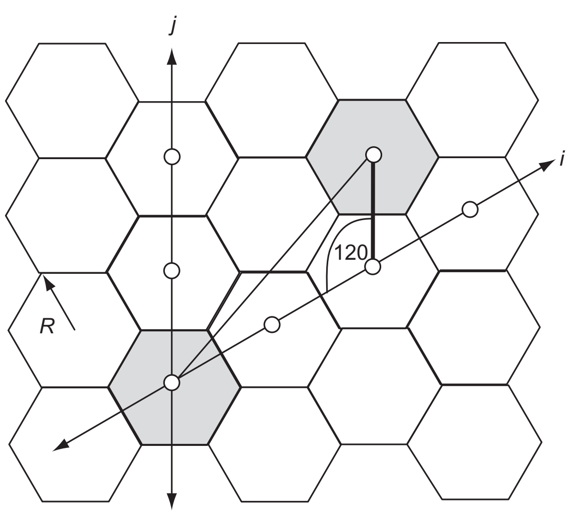

Fig. 4.11 *Relationship between* K *and shift parameters* i, j

Step 4. *To establish relation between D, d and shift parameters*

Let 'd' be the distance between two adjacent cells and 'D' be the distance from the centre of the cell under consideration to the centre of a nearest cochannel cell.

Using cosine formula to $\Delta\ \ Y$ in Fig. 4.11, we have

$$^2 = \ Y^2 + Y\ ^2 - 2 \times \ Y \times Y \ \cos 120$$

Or, $D^2 = (i \times d)^2 + (j \times d)^2 - 2 \times (i \times d) \times (j \times d) \cos 120$

Or, $D^2 = (i \times d)^2 + (j \times d)^2 - 2 \times (i \times d) \times (j \times d) \times (- \ )$

Or, $D^2 = (i \times d)^2 + (j \times d)^2 + (i \times d) \times (j \times \mathrm{d})$

Or, $D^2 = d^2\ (i^2 + j^2 + i \times j)$ (4.6)

Step 5. *To establish relation between D, R and shift parameters*

Using Eqs. (4.4) and (4.6),

$$D^2 = 3 \times R^2 \times (i^2 + j^2 + i \times j) \qquad (4.7)$$

Step 6. *Area of a large hexagon, $A_{large\ hexagon}$*

By joining the centres of the six nearest neighbouring cochannel cells, a large hexagon is formed with radius equal to D, which is also the cochannel cell separation. Refer Fig. 4.12. The area of the large hexagon with radius D can be given as

$$A_{large\ hexagon} = (3\sqrt{3} / 2) \times D^2 \tag{4.8}$$

Using Eq. (4.7),

$$A_{large\ hexagon} = (3\sqrt{3} / 2) \times 3 \times R^2 \times (i^2 + j^2 + i \times j) \tag{4.9}$$

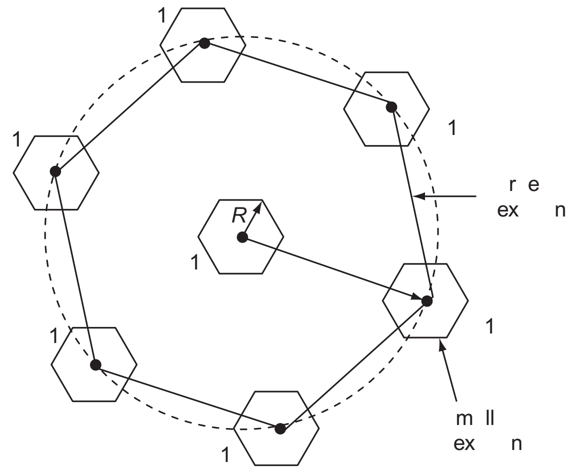

Fig. 4.12 | *A larger hexagon in the first tier*

Step 7. *To determine number of cells in the large hexagon, L*

Number of cells in large hexagon,

$$L = A_{large\ hexagon} \quad A_{small\ hexagon} \tag{4.10}$$

Using Eqs. (4.9) and (4.5) in Eq. (4.10), we get

$$L = 3 \times (i^2 + j^2 + i \times j) \tag{4.11}$$

Step 8. *Relationship between L and cluster size K*

On the other hand, from the geometry as given in Fig. 4.13 for cluster size 7, as an example, it can be easily seen that the larger hexagon formed by joining the centers of cochannel cells in the first tier encloses 7 cells of the middle cluster plus 1/3rd of the number of 7 cells of all surrounding six neighbouring clusters. In general, it can be computed that the larger hexagon encloses the centre cluster of K cells plus 1/3rd the number of the cells associated with six other peripheral clusters in the first tier.

Hence, the total number of cells enclosed by the larger hexagon is

$$L = K + 6 \times \ (1 \ \ 3) \times K$$

Or,
$$L = 3 \times K \tag{4.12}$$

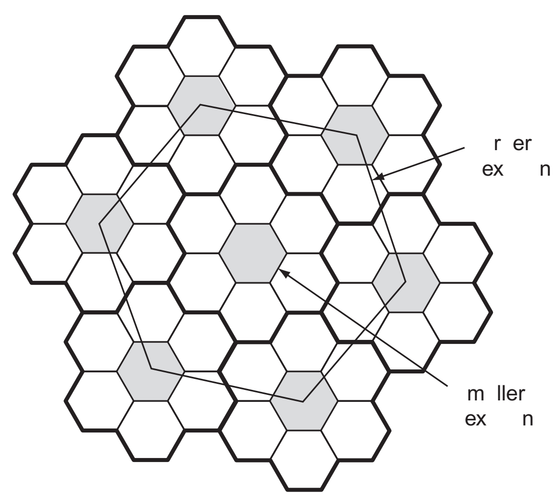

Fig. 4.13 | *Number of clusters in the first tier for K = 7*

Step 9. *To establish relation between K and shift parameters*

From Eq. (4.11) and (4.12), we get

$$3 \times K = 3 \times (i^2 + j^2 + i \times j)$$

Or,
$$K = i^2 + j^2 + i \times j \tag{4.13}$$

Thus, the cluster size K (i.e., the number of cells per cluster) can be determined from the shift parameters i, j using the above mathematical expression, where i and j can have any integer values such as 0, 1, 2, 3, . so on.

Remarks It may be worthwhile to note here that (i, j) cannot have (0, 0) value because $K = 0$ is meaningless. Also, $(i, j) = (0,1)$ or $(1, 0)$ results in $K = 1$ which is applicable only in CDMA cellular systems in which all cells use the same frequency channels, but not FDMA- or TDMA- based cellular systems in which adjacent cells cannot be assigned the same frequency channels.

EXAMPLE 4.9 To determine cluster size *K*

Determine the number of cells in clusters for the following values of the shift parameters i and j in a regular hexagonal geometry pattern:
(a) $i = 2$ and $j = 4$ (b) $i = 3$ and $j = 3$

Solution

(a) *To determine the number of cells in a cluster for $i = 2$ and $j = 4$*
Using the relationship, $K = i^2 + j^2 + i \times j$
The number of cells in a cluster, also called cluster size K, can be determined as

$$K = 2^2 + 4^2 + 2 \times 4 = 4 + 16 + 8 = 28$$

(b) *To determine the number of cells in a cluster for $i = 3$ and $j = 3$*

$$K = 3^2 + 3^2 + 3 \times 3 = 9 + 9 + 9 = 27$$

Therefore, it must be remembered that only the specific values of cluster size are valid and used in frequency planning of the cellular system as determined by the given integers of shift parameters i and j in a regular hexagonal geometry.

Table 4.1 depicts several frequency reuse patterns, together with the cluster sizes for easy reference.

Table 4.1 *Frequency reuse pattern and cluster size*

requency reuse pattern (i, j) *or* (j, i)	*Cluster size* $K = i^2 + j^2 + i \times j$
(1, 1)	3
(2, 0)	4
(2, 1)	7
(3, 0)	9
(2, 2)	12
(3, 1)	13
(4, 0)	16
(2, 3)	19
(4, 1)	21
(5, 0)	25
(3, 3)	27

4.6 FREQUENCY REUSE DISTANCE

Reusing an identical frequency channel in different cells is limited by cochannel interference between cells and the cochannel interference can become a major problem in cellular communication. So it is desirable to find the minimum frequency reuse distance D in order to reduce this cochannel interference.

The minimum distance, which allows the same frequency to be reused in cochannel cells, will depend on many factors such as

- the number of cochannel cells in the vicinity of the centre cell,
- the type of geographic terrain contour,
- the antenna height, and
- the transmitted power at each cell-site.

Assume that the size of all the cells is approximately same; the cell size is usually determined by the coverage area of the signal strength in each cell. As long as the cell size is fixed, cochannel interference is independent of transmitted power of each cell. It means that the received signal threshold level at the mobile unit is adjusted to the size of the cell.

Actually, cochannel interference is a function of a parameter known as frequency reuse ratio, q, and is defined as

$$q = D/R \tag{4.14}$$

where D is the distance between two nearest cochannel cells marked as C_1, and R is the radius of the cells under consideration, as shown in Fig. 4.14. It may be noted here that this ratio is applicable for any value of cluster size K.

The parameter q is also referred to as the cochannel reuse ratio or the cochannel reuse factor or cochannel interference reduction factor or frequency reuse ratio.

Facts to Know!

The real power of the cellular concept is that interference is not related to the absolute distance between cells but to the ratio of the distance between cochannel (same frequency) cells to the cell radius.

Fig. 4.14 | *Frequency reuse ratio $q = D/R$*

EXAMPLE 4.10 | **Frequency reuse distance, *D***

Determine the distance from the nearest cochannel cell for a cell having a radius of 0.64 *km and a cochannel reuse factor of* 12.

Solution

The radius of a cell, $R = 0.64$ km (given)
The cochannel reuse factor, $q = 12$ (given)
To determine the distance from the nearest cochannel cell, D
We know that $q = D/R$,
Or, $D = q \times R$
Therefore, $D = 12 \times 0.64 \text{ km} = 7.68 \text{ km}$
Hence, the distance from the nearest cochannel cell $D = 7.68$ km
Thus, the important parameters of the network designed on cellular approach are

- Reuse pattern, K
- Reuse distance, D
- Frequency reuse factor, q

The frequency reuse factor determines the minimum distance for repeating a set of frequencies and is expressed as $q = D/R$ where R is the cell radius. The spectrum efficiency is most significantly influenced by the frequency reuse factor. The concept of frequency reuse when applied, permits the system to meet the important objective of serving a large area, while using a relatively small frequency spectrum. But if the network is not designed properly, serious interferences may occur. To minimise interference, there must be adequate spatial separation between cells that use the same frequencies and the cells that use adjacent channel frequencies. The frequency assignment depends on the channel bandwidth, modulation scheme adopted, reuse factor and the carrier-to-interference ratio requirements.

EXAMPLE 4.11 | **Frequency reuse ratio, *q***

Determine the frequency reuse ratio for a cell radius of 0.8 *km separated from the nearest cochannel cell by a distance of* 6.4 *km.*

Solution

The radius of a cell, $R = 0.8$ km (given)
The distance between nearest cochannel cells, $D = 6.4$ km (given)
To determine the frequency reuse ratio, q
We know that $q = D \quad R$
Or, $q = 6.4 \quad 0.8 = 8$
Hence, the frequency reuse ratio for given parameters $\boldsymbol{q} = 8$

The frequency reuse ratio q is related to the cluster size (or frequency reuse factor) K by

$$\boldsymbol{q} = \boldsymbol{D} \quad \boldsymbol{R} = \sqrt{3K} \tag{4.15}$$

Theoretically, a large K is desired. However, the total number of allocated channels N is fixed. When K is too large, the number of channels assigned to each of K cells becomes small. It is always true that the total number of allocated channels N in a cluster is divided by K to calculate the system capacity per cell. As K increases, system capacity per cell and hence spectrum efficiency will reduce significantly.

EXAMPLE 4.12 Relationship between frequency reuse ratio *q* and cluster size *K*

Illustrate and prove that for a regular hexagonal geometry, the frequency reuse ratio is given by the relationship

$$q = \sqrt{3K}$$

where $K = i^2 + j^2 + i \times j$; i and j being the shift parameters.

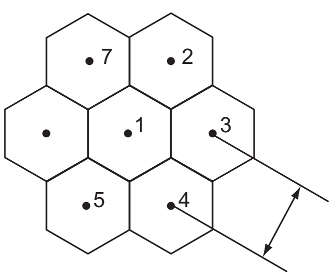

Fig. 4.15 *Distance between two adjacent cells,* d.

Solution

The geometry of an array of regular hexagonal cells is depicted in Fig. 4.15, where R is the radius of the hexagonal cell (from its centre to one of its vertex). A hexagon has exactly six equidistant neighbouring hexagons corresponding to six sides of the hexagon.

Step 1. *Relation between d and R*

Let the distance between the centres of two adjacent hexagonal cells be denoted by d. Then, using the trigonometry, it can be seen that

$$d = \sqrt{3}R \tag{4.16}$$

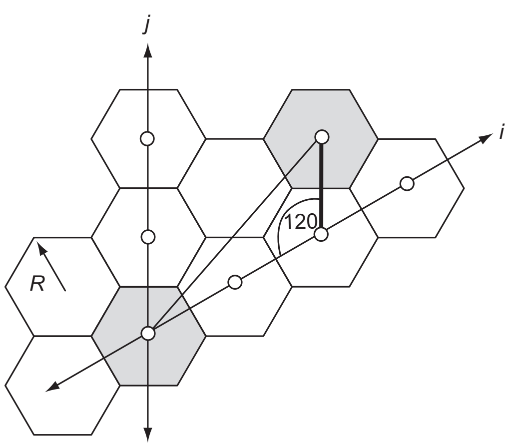

Fig. 4.16 *Cochannel cell in a regular hexagonal geometry*

Step 2. *Procedure of locating a cochannel cell*

The nearest cochannel hexagonal cell to the cell under consideration can be located using shift parameters i, j in a regular hexagonal geometry. Figure 4.16 depicts the regular hexagonal geometry of one colocated cell. The procedure of locating a cochannel cell, corresponding to any one side of the hexagon is as follows:

- Firstly, move i number of cells along the i axis from the centre of the hexagonal cell under consideration (say point to point Y) along one side of hexagon.
- Secondly, turn 60 degrees counterclockwise.
- Then move j number of cells along j axis (point Y to point) to locate the centre of the nearest cochannel cell.

Let D be the distance from the centre of the cell under consideration to the centre of a nearest cochannel cell (that is,).

Step 3. *To derive the relation between D. d and shift parameters*

Applying cosine formula to $\Delta \quad Y$, we have

$$D^2 = (i \times d)^2 + (j \times d)^2 - 2 \times (i \times d) \times (j \times d) \cos 120$$

Or, $D^2 = (i \times d)^2 + (j \times d)^2 - 2 \times (i \times d) \times (j \times d) \times (- \quad)$

Or, $D^2 = (i \times d)^2 + (j \times d)^2 + (i \times d) \times (j \times d)$

Or, $D^2 = d^2 (i^2 + j^2 + i \times j)$ (4.17)

Using Eq. (4.16),

$$D^2 = 3 \times R^2 \times (i^2 + j^2 + i \times j) \quad (4.18)$$

Step 4. *To establish relationship between K and shift parameters*

$$K = i^2 + j^2 + i \times j \quad \text{(given)}$$

Substituting it in Eq. (4.18), we get

$$D^2 = 3 \times R^2 \times K$$

Or, $D^2 \quad R^2 = 3 \times K$

Or, $D \quad R = \sqrt{3K}$

By definition $q = D \quad R$; therefore, we get

$$q = \sqrt{3K}$$

Thus, the frequency reuse ratio q can be determined from the cluster size K (i.e., the number of cells per cluster).

$q = D/R$ ratio is a parameter used to describe the frequency reuse factor for a cellular system. The D/R ratio for any cellular system determines the reuse factor as well as the distance D between the frequency reusing cell-sites and the radius R of the serving cell-sites. Table 4.2 illustrates standard frequency reuse ratios for different cluster size, K.

Table 4.2 *Frequency reuse ratio and cluster size*

Cluster size K	*requency reuse ratio* $q = \sqrt{3K}$
3	3.00
4	3.46
7	4.58
9	5.20
12	6.00
13	6.24
19	7.55
21	7.94
27	9.00

Because the D/R measurement is a ratio, if the radius of the cell is decreased, the distance between frequency reusing cochannel cells sites is also decreased in the same proportion for maintaining same cochannel interference reduction factor. Conversely, if a cell has a large radius, the distance between frequency reusing cells must be proportionally increased to maintain the same D/R ratio.

Since q increases with K and a smaller value of K has the effect of increasing the capacity of the cellular system. But at the same time, this results into increase in cochannel interference. Hence the choice of q (or K)

has to be made such that the signal-to-cochannel interference ratio is at an acceptable level. If all the cell-sites transmit the same power, then as K increases, the frequency reuse distance D increases. This increased D reduces the possibility that cochannel interference may occur.

The frequency reuse method is useful for increasing the efficiency of spectrum usage but results in cochannel interference because the same frequency channel is used repeatedly in different cochannel cells. In most mobile radio environments, use of $K = 7$ is not sufficient to avoid interference. Increasing K greater than 7 would reduce the number of channels per cell, and that would also reduce the spectrum efficiency.

Now the challenge is to obtain the optimum value of K that can still meet the desired system performance requirements in terms of system capacity, spectrum utilisation and signal quality. This involves estimating cochannel interference and selecting the minimum frequency reuse distance D to reduce cochannel interference.

4.7 COCHANNEL INTERFERENCE AND SIGNAL QUALITY

The frequency reuse method is useful for increasing the efficiency of spectrum usage but results in cochannel interference because the same frequency channel is used repeatedly in different cochannel cells in a service area. In this situation, the received signal quality is affected by the amount of radio coverage area as well as the cochannel interference.

Facts to Know!

Cochannel interference occurs equally in all available channels in a given area assuming non-selective fading channel environment.

The cochannel interference is caused due to the reuse of the same carrier frequency at different geographical locations. Because cochannel interfering signals are amplified, processed and detected in the same manner as the desired signal, the receiver is particularly vulnerable to these emissions. Thus, cochannel interference may either desensitise the receiver or override or mask the desired signal. It may also combine with the desired signal to cause serious distortions in the detected output.

The cochannel interference can then be measured by selecting any one channel (as one channel represents all the channels) and transmitting on that channel at all cochannel sites. In a fully equipped hexagonal-shaped cellular system, there are always six cochannel interfering cells in the first tier. Figure 4.17 depicts a typical field measurement test set-up 1 to measure cochannel interference at the mobile unit, in which the mobile unit is moving in its serving cell.

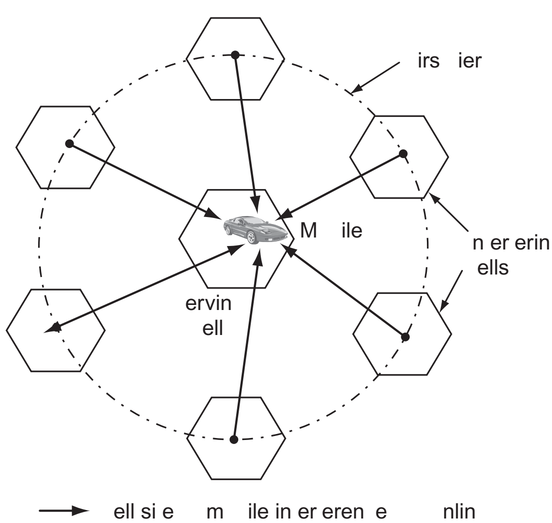

Fig. 4.17 *Test 1: Cochannel interference measurement at the mobile unit*

Let the symbol C, I, and N denote respectively the power of the desired signal, the power of the cochannel interference, and the power of the noise at the output of the receiver demodulator. Cochannel interference can be experienced both at the cell-site and at mobile units in the serving cell. If the interference is much greater then the carrier to interference ratio C/I at the mobile units caused by the six interfering cell-sites is (on the average) the same as the C/I received at the serving cell site caused by interfering mobile units in the six cells. According to the reciprocity theorem and the statistical summation of radio propagation, the two C/I values can be very close.

A channel-scanning mobile receiver records three received signals while moving in any one cochannel cell, under the following conditions:

- When only the serving cell transmits (signal recorded is termed as C)
- Cell-sites of all six cochannel cells only transmit (signal recorded is termed as I)
- No transmission by any cell-site (signal recorded is termed as N)

Let a value of $C/I = 18$ dB or greater be acceptable in a cellular system. In general, the performance of such types of interference-limited cellular system can be evaluated from the following results.

(a) If the carrier-to-interference ratio C/I is greater than 18 dB in most of the area being served by a cell, the system is said to be properly designed.
(b) If C/I is less than 18 dB and carrier-to-noise ratio C/N is greater than 18 dB in some areas, the system is said to have a cochannel interference problem.
(c) If both C/I and C/N are less than 18 dB and C/I is approximately same as C/N in a given area, the system is said to have a radio coverage problem.
(d) If both C/I and C/N are less than 18 dB and C/I is less than C/N in a given area, the system is said to have both cochannel interference as well as radio coverage problem.

In fact, the reciprocity theorem can be applied for the study of area coverage problem but not so accurately for the study of cochannel interference problem at the cell-site. Therefore, it is recommended to perform Test 2 to measure cochannel interference at the cell-site. In Test 2, the mobile unit is transmitting in its serving cell as well as six mobile units are transmitting in cochannel cells simultaneously at the same frequency channel. Figure 4.18 depicts a typical field measurement test set-up 2 to measure cochannel interference at the cell-site.

The received signal-level measurements are recorded at the serving cell-site, under the following conditions:

– When only the mobile unit in the serving cell transmits (signal recorded is termed as C)
– Up to six interference levels are obtained at the serving cell-site from six mobile units transmitting in six cochannel interfering cells (the statistical average signal recorded is termed as I)
– No transmission by any mobile unit (signal recorded is termed as N)

Then the C/I and C/N received at the serving cell site is computed. The test result analysis will be same as obtained in Test 1. From the analysis of the results, it can be easily deduced whether the cellular system has a radio coverage problem, or a cochannel interference problem or both.

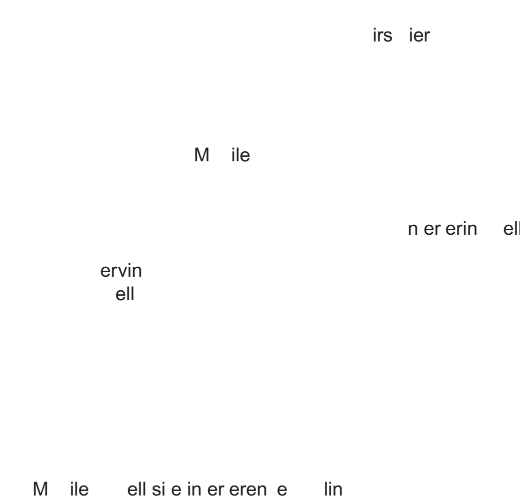

Fig. 4.18 *Test 2: Cochannel interference measurement at the cell-site*

4.8 COCHANNEL INTERFERENCE REDUCTION METHODS

Interference is the major limiting factor in the performance of cellular communication systems. Sources of interference may include another mobile operating in the same cell, other cells operating in the same frequency band, an on-going communication link in a neighbouring cell, or any noncellular system that may be leaking energy into the cellular frequency band.

Cells using the same set of frequencies are called cochannel cells, and the interference between received signals from these cells is called cochannel interference. RF channels are interference limited. If the different cells in the entire cellular system were to use different sets of frequencies, intercell interference would be kept at a minimum. However, the system capacity would be limited. When the number of mobile subscribers increase, the channels, which are limited in numbers, have to be repeatedly reused in a different area, This provides many cochannel cells, which can serve a large number of mobile subscribers simultaneously, thereby increasing the system capacity.

In fact, deployment of frequency reuse is necessary to enhance the system capacity. On the other hand, frequency reuse will introduce cochannel interference from cells using the same set of frequencies. In this situation, the received signal quality is affected by the amount of cochannel interference as well as the extent of radio coverage. Therefore, frequency reuse is required to be carefully planned so that cochannel interference is kept at an acceptable level.

Facts to Know!

In a cellular reuse structure, the effect of cochannel interference must be included in the channel capacity estimate.

Except for the cochannel cells, other adjacent cells operate at frequencies different from those of the cell under consideration so that interference from non cochannel cells is minimal. Intercell interference in the downlink (cell-site to mobile subscriber link) that affects the reception at the individual mobile subscribers may be more of a problem than uplink interference at the cell-site receiver. The reason for this can be attributed to the fact that the cell-site receiver may be more sophisticated than the receivers of the individual mobile subscribers. Intercell interference is thus dominated by cochannel interference. It is, therefore, of prime interest to assess system performance, taking into consideration interference from cochannel cells.

Assuming that the local noise is much less than the interference level and can be neglected in a cellular mobile system, cochannel interference can be reduced by a number of methods such as the following:

Method 1 *Increasing the separation between two cochannel cells, D*

As D increases, interfering signal power from cochannel interfering cell decreases drastically (since received signal power is inversely proportional to the fourth power of the distance in a mobile environment).

But it is not advisable to increase D because as D is increased, K should also be increased. A high value of K means less number of channels available per cell for a given RF spectrum. This results into reduction of the system capacity in terms of number of channels available per cell.

Method 2 *Lowering the antenna heights at the cell-site*

It is a good approach to reduce cochannel interference in some circumstances, such as on a high hill or in a valley. The effective antenna height, rather than the actual antenna height, is always considered in the system design. Moreover, the effective antenna height varies according to the location of the mobile unit in such terrains.

When the cell-site is installed on top of the hill in a given service area, the effective antenna height is more than the actual antenna height. In order to reduce cochannel interference, lower antenna height can be used without reducing the received signal power either at the cell-site or at the mobile unit. Similarly, the lowered cell-site antenna height in a valley is very effective in reducing the radiated power in a distant high-elevation area where the mobile subscriber is supposed to be present.

However, lowering the cell-site antenna height does not always reduce the cochannel interference. As an instance, in a forested area, the cell-site antenna should clear the tops of the longest trees in the vicinity, especially when they are very close to the antenna. In this case, lowering the antenna height would not be proper for reducing cochannel interference because excessive attenuation of the desired signal would occur in the vicinity of the cell-site antenna as well as in its cell boundary if the cell-site antenna were installed below the treetop level.

Method 3 *Using directional antennas at the cell-site*

The use of directional antennas (three or six) in each cell can serve the purpose of reduction of cochannel interference if the interference cannot be eliminated by a fixed separation of cochannel cells. This will also increase the system capacity even when the traffic increases.

The cochannel interference can be further reduced by intelligently setting up the directional antenna. For example, installation of a 120 directional antenna can reduce the cochannel interference in the system by eliminating the radiation to the rest of its 240 sector. However, cochannel interference can exist even when a directional antenna is used, as the serving site can interfere with the cochannel sector that is directly ahead.

> **Facts to Know!**
>
> Using umbrella pattern of a staggered discone antenna or using a parasite antenna with a single active element having no corner or a ground reflector can also reduce the cochannel interference.

The cochannel interference can also be reduced by down tilting of the antenna patterns—mechanically or electronically. The mechanical downtilting is to downtilt the antenna physically. The electronic downtilting is to change the phases among the elements of a collinear array antenna. In addition, a significant reduction of interference of the order of 7 to 8 dB or the gain of C/I in the interference-receiving cell can be achieved by using a notch in the centre of the antenna pattern at the interfering cell. The resulting weak signal coverage in a small area in the serving cell can be compensated by the use of sufficient transmitting power at the cell-site.

Method 4 *Use of diversity scheme at the receiver*

The diversity scheme applied at the receiving end of the cell-site antenna is an effective technique for reducing the interference because any measure taken at the receiving end to improve signal interference will not cause additional interference. For instance, the separation of two receiving antennas at the cell-site meeting the requirement of $h/s = 11$, where 'h' is the antenna height and 's' is the separation between two antennas, results into the correlation coefficient $\rho \leq 0.7$ for a two-branch diversity.

A selective combiner can then be used to combine two correlated signals. The mobile transmitter could undergo up to 7-dB reduction in power and attain the same performance as a nondiversity receiver at the cell-site. Thus, interference from the mobile transmitters to the cell-site receivers can be drastically reduced.

Key Terms

- Adjacent channel interference
- Cell
- Cell sectorisation
- Cellular network
- Channels
- Cluster
- Cluster size
- Cochannel interference
- Coverage area
- Directional antenna
- Diversity
- Frequency reuse
- Omni-directional antenna
- Reuse distance
- Reuse factor
- Spectrum
- System capacity

Summary

This chapter provides an overview of the cellular concept. The essential principles of cellular communication include frequency reuse, cochannel cells, and cochannel interference. Various cell parameters based on regular hexagonal cellular pattern, including cluster, frequency reuse distance, reuse factor are described. As limited frequency spectrum has been allocated for cellular communications, the frequency reuse technique is shown to enhance the spectrum efficiency as well as user capacity significantly for both FDMA and TDMA systems. However, reuse of same frequency in distant cells gives rise to cochannel interference which results in degradation of received signal quality. The methods of cochannel interference reduction with the aim of maintaining the desired signal quality and cell capacity are introduced here. In the next chapter, the emphasis is on the design techniques of cell-site antenna installation as well as significant improvement of signal quality with cell sectoring.

Important Equations

- $K = i^2 + j^2 + i \times j$ (4.13)
- $q = D \quad R$ (4.14)
- $q = D \quad R = \sqrt{3K}$ (4.15)

Short-Answer Type Questions with Answers

A4.1 Why can the actual shape of the cell may not be either a circle or a regular geometrical shape
The actual shape of the cell is determined by the desired received signal level by the mobile subscribers from its cell-site transmitter in its service area. The received signal is affected by many factors including reflections, refractions, and contour of the terrain as well as multipath propagation due to presence of natural and man-made structures. Therefore, the actual shape of the cell may not be either a circle or a regular geometrical shape but may be a zigzag shape which indicates a true radio coverage area. However, for proper planning and analysis of a cellular system, an appropriate model of a regular geometry shape is needed for the cell.

A4.2 Why can the cellular topology not be formed by using ideal circular shape
The cellular topology formed by using ideal circular shape results into overlaps or gaps between them which is not desirable in cellular communications which has to be essentially continuous. It is highly desirable to construct the cellular system architecture such that the cells do not overlap, and are tightly packed without any dead signal spots. This form of cellular configuration requires the use of regular geometrical topologies (say, a regular hexagonal pattern) instead of a circular shape.

A4.3 Distinguish between a cell and a cell-site.
A cell is a wide geographical service area which is served by the base-station radio equipment and their antennas mounted on tall towers located at the centre of the cell. A cell-site is a location or a point at the centre of the cell which provides radio signal coverage to a cell.

A4.4 What is the significance of cell size
Each cell size varies depending on the landscape. Typical size of a cell may vary from a few 100 metres in urban areas to several kilometres in the rural areas. Smaller cells are used when there is a requirement to support a large number of mobile subscribers in a small service area or when a low transmission power may be required to reduce the effects of interference. Large cells are employed in remote areas, coastal regions, and areas with few mobile subscribers.

A4.5 Mention the major limitations of conventional radio communication systems.
The inefficient spectrum utilisation, limited service area capability, inadequate system capacity, and lower grade of service. The conventional radio communication systems are usually designed for providing service in an autonomous geographic zone and by selecting RF channels from a specified allocated frequency band.

A4.6 What is the core concept of the cellular communications
Frequency reuse is the core concept of the cellular communications. The plan of dividing the large geographic service area into many small contiguous cells and using a low-power transmitter with low antenna at base station in each cell is referred to as cellular communications. The design process of selecting and allocating channel groups for all the cellular base stations within a system is called frequency reuse planning.

A4.7 List few typical technical issues for proper design and planning of a cellular network.
The technical issues for proper design and planning of a cellular network include selection of a suitable frequency reuse pattern, physical deployment

and radio coverage modeling, plans to account for the expansion of the cellular network, analysis of the relationship between the capacity, cell size, and the cost of the infrastructure.

A4.8 What complications arise due to usage of smaller cells
Smaller cells are needed in areas of the higher traffic demand. As cells become smaller, more cell-sites are needed and hand-off occurs more frequently. This requires more computing power and faster response both at the system level and in the individual mobile phone. Typically, once the radius drops below about 0.5 km the hand-offs occur so frequently that it is difficult to cope with a mobile moving at high speed.

A4.9 Assume that the allocated number of channels in a cluster are fixed and the size of each cell is kept constant. How will the system design be affected by choosing higher value of cluster size
More number of cells per cluster (that is, higher value of cluster size) would mean less number of available channels per cell, less system capacity per cell, and lower cochannel interference because cochannel cells will be located farther apart with higher value of cluster size.

A4.10 How will the system design be affected by choosing a smaller value of cluster size
Assuming that the allocated spectrum and the cell size is fixed, less number of cells per cluster (that is, smaller value of cluster size) would mean more number of available channels per cell, higher system capacity per cell, but increase in cochannel interference due to reduced distance between cochannel cells.

A4.11 List the factors to determine the minimum distance between cochannel cells.
The minimum distance between cochannel cells, which allows the same frequency to be reused in these cells, depends on many factors such as the number of cochannel cells in the vicinity of the centre cell, the type of geographic terrain contour, the cell-site antenna height and the transmitted power at each cell-site. Usually, the size of all the cells is approximately same and fixed. Then the cochannel interference is independent of transmitter power of each cell.

A4.12 What are the important parameters of the wireless communication network designed on cellular approach
The three important design parameters in cellular architecture are frequency reuse pattern (K), frequency reuse distance (D), and frequency reuse factor (q). These are related to one another by the expression $q = D/R$. The frequency reuse factor determines the minimum distance between nearest cochannel cells for repeating a set of frequency channels.

A4.13 Can the value of cluster size be increased more than 7 to minimise the effect of cochannel interference
In most mobile radio environments, use of cluster size $K = 7$ is not sufficient to avoid interference. But increasing the value of K more than 7 would reduce the number of channels per cell, assuming the available spectrum fixed in the system. This would result in reduction in the cell capacity and hence the spectrum efficiency. Therefore, there is trade-off among various system performance parameters such as system capacity, spectrum efficiency and signal quality so as to determine the optimum value of cluster size.

A4.14 How is signal quality affected by employing frequency reuse concept in cellular communication systems
The frequency reuse method is useful for increasing the efficiency of spectrum usage but results in cochannel interference which degrades the received signal quality. Cochannel interference occurs equally in all available channels in a given area assuming non-selective fading channel environment. Since the cochannel interfering signals are amplified, processed and detected in the same manner as the desired signal, the receiver is particularly vulnerable to these emissions. Thus, cochannel interference may either desensitise the receiver or override or mask the desired signal. It may also combine with the desired signal to cause serious distortions in the detected output.

A4.15 What could be the possible sources of interference which may limit the performance of cellular communication systems
Interference is the major limiting factor in the performance of cellular communication systems.

The possible sources of interference may include other cells operating in the same frequency band (cochannel interference), another mobile operating in the same cell (near-and-far interference), an on-going communication link in a neighbouring cell (adjacent channel interference), or any non-cellular system that may be leaking energy into the cellular frequency band.

A4.16 List some methods to reduce cochannel interference in cellular communication network. The methods to reduce cochannel interference in cellular communication network include using the higher value of cluster size (resulting in increased separation between two cochannel cells and hence reduction in cochannel interference), lowering the antenna heights at the cell-site in situations like high hill or a valley, using directional antennas at the cell-site (sectoring the omnidirectional cell), down tilting of the antennas at the cell-site, providing a notch in the centre of the antenna pattern at the interfering cell-site, using umbrella pattern of a staggered discone antenna or using a parasite antenna with a single active element, and use of diversity antenna system the receiver.

Self-Test Quiz

S4.1 A regular ______ shaped cell is the closest approximation to a circle which has been used for cellular system design.
(a) circular (b) triangular
(c) square (d) hexagonal

S4.2 The propagation considerations recommend the ______ as a cell shape for defining the area covered by a particular cell site.
(a) circle
(b) triangle
(c) square
(d) hexagon

S4.3 A ______ can be repeated any number of times in a systematic manner in order to cover the designated large geographical service area.
(a) channel (b) cell
(c) cell-site (d) cluster

S4.4 ______ is the major concern in frequency reuse.
(a) System noise
(b) Cochannel interference
(c) Adjacent-channel interference
(d) Intermodulation

S4.5 A mobile communication system has an allocated number of 1000 voice channels. If the service area is divided into 20 cells with a frequency reuse factor of 4, the system capacity is
(a) 1000 (b) 4000
(c) 5000 (d) 20000

S4.6 A mobile communication system is designed with a cluster size of 7. If the area of a cell is 5 km^2, the area of the cluster is
(a) 5 km^2 (b) 25 km^2
(c) 35 km^2 (d) 49 km^2

S4.7 A cellular network is reconfigured with a frequency reuse pattern of 7 instead of 4. The increase in overall system capacity is approximately
(a) 1.7 times (b) 4 times
(c) 7 times (d) 28 times

S4.8 A service area is covered with 10 clusters having 7 cells in each cluster and 16 channels assigned in each cell. The number of channels per cluster are
(a) 16 (b) 70
(c) 112 (d) 1120

S4.9 If there are J number of channels per cell and K number of cells per cluster, then the total number of channels available in a cluster is equal to
(a) J / K (b) $J \times K$
(c) K / J (d) $J \times K^2$

S4.10 In a regular hexagonal geometry pattern, the number of cells in a cluster formed by $i = 2$ and $j = 2$ are
(a) 4 (b) 7 (c) 9 (d) 12

S4.11 For a given frequency reuse ratio of 8 and the cell radius of 0.8 km, the distance between the nearest cochannel cells is
(a) 6.4 km (b) 0.8 km
(c) 0.1 km (d) 8.8 km

S4.12 The distance between the centers of two adjacent hexagonal cells, each cell having radius of 2 km, is
(a) $2\sqrt{3}$ (b) $\sqrt{3}$
(c) $3\sqrt{3}$ (d) $\sqrt{3}/2$

S4.13 For a given frequency reuse ratio of 3, the cluster size is
(a) 3 (b) 4 (c) 7 (d) 12

S4.14 Assuming each cell-site transmits the same power then as the value of cluster size K is increased, the cochannel interference will
(a) remain same
(b) increase marginally
(c) increase drastically
(d) decrease

S4.15 The results in cochannel interference measurement test at the mobile unit show that C/I is less than 18 dB and carrier-to-noise ratio C/N is greater than 18 dB in some areas. The cellular system is said to
(a) be properly designed
(b) have cochannel interference problem
(c) have radio coverage problem
(d) have cochannel interference as well as radio coverage problem

S4.16 Cells using the same set of frequencies are called
(a) neighbouring cells
(b) adjacent channel cells
(c) cochannel cells
(d) clusters

S4.17 As the separation between two cochannel cells increases, interfering signal power from cochannel interfering cell
(a) increases marginally
(b) increases significantly
(c) decreases significantly
(d) remains same

S4.18 The use of directional antennas at the cell-site will result into
(a) reduction in cochannel interference and increase in system capacity
(b) reduction in cochannel interference as well as system capacity
(c) increase in cochannel interference as well as system capacity
(d) increase in cochannel interference and increase in system capacity

Answers to Self-Test Quiz

S4.1 (d); S4.2 (a); S4.3 (d); S4.4 (b); S4.5 (c); S4.6 (c); S4.7 (a); S4.8 (c); S4.9 (b); S4.10 (d); S4.11 (a); S4.12 (a); S4.13 (a); S4.14 (d); S4.15 (b); S4.16 (c); S4.17 (c); S4.18 (a)

Review Questions

Q4.1 Define the terms cell, footprint, cluster and frequency reuse distance in a cellular system.

Q4.2 Why is a hexagonal cell shape preferred over triangular or square cell shapes to represent the cellular architecture?

Q4.3 Why is a regular octagon-shaped cell not used for design of cellular pattern although it is closer to a circle as compared to a regular hexagon-shaped cell?

Q4.4 Describe briefly the principle of frequency reuse in the context of a cellular network. Can the same frequency be repeated within a cluster?

Q4.5 Which is the main problem caused due to frequency reuse in a cellular architecture? How can it be minimised?

Q4.6 Write the procedure for locating the cochannel cells in the first tier using a regular hexagonal pattern.

Q4.7 How does frequency reuse increase spectrum efficiency in a cellular system? Explain it with the help of suitable example which compares a cellular mobile system with a conventional mobile system.

Q4.8 Define cochannel cell and adjacent channel cell. Mark the frequency channel groups in an illustration of a fully equipped first tier regular hexagonal geometrical pattern based on 4-cell cluster.

Q4.9 Define cochannel cell and cochannel interference. How does cochannel interference become a serious concern in the design of a cellular mobile system?

Q4.10 Show that lowering the cell-site antenna height on the hill does not reduce the received signal power at the mobile unit.

Q4.11 Illustrate that lowering the cell-site antenna height in the valley is very effective in reducing the received signal power in a distant high-elevation area.

Analytical Problems

P4.1 Out of the numbers 3, 8, 15, 21, 23, 30, and 47, what values are possible for a cluster size in a regular hexagonal geometry of cellular architecture? Justify the same using the relationship between the cluster size K and shift parameters i and j.

P4.2 A large city with an area of 1000 km^2 is required to be covered by a finite number of cells with a radius of 2 km each. How many cell-sites would be required, assuming regular hexagonal-shaped cells?

P4.3 A new cellular service provider decides to employ a cluster of 19 cells for frequency reuse.

(a) Draw one such cluster structure.
(b) Is there an alternative cluster of 19 cells available for Part (a)? If yes, how?
(c) Determine the reuse distance for Part (a), if the radius of a cell is 2 km.

P4.4 The size and shape of each cluster in a cellular system must be designed in such a way so as to cover adjacent areas in a contiguous and non-overlapped manner. Define such pattern for the cluster size of 4 cells, 7 cells and 9 cells.

P4.5 Calculate the number of channels per cell for a cluster size of 7 in a cellular system which has 1001 radio channels available for handling traffic. The serving area of the complete system is 2100 km^2 and the area of the cell is 6 km^2.

P4.6 A cellular service provider is allocated an RF spectrum of 30 MHz. The cellular system is configured with a cluster size of 12. It has simplex channel bandwidth of 25 kHz and 10 channels per cell are reserved as control channels.

(a) Find the reuse distance if the radius of each cell is 5 kms.
(b) How many channels per cell are available for traffic purpose?
(c) How many calls can be simultaneously processed by each cell if 8 users can share each channel?

P4.7 A cellular system is designed for a cluster size $K = 4$ and is shown in Fig. 4.19. Verify it by using the formula $K = i^2 + j^2 + i \times j$ and the procedure for locating cochannel cells. If it is not correct, draw the correct frequency plan.

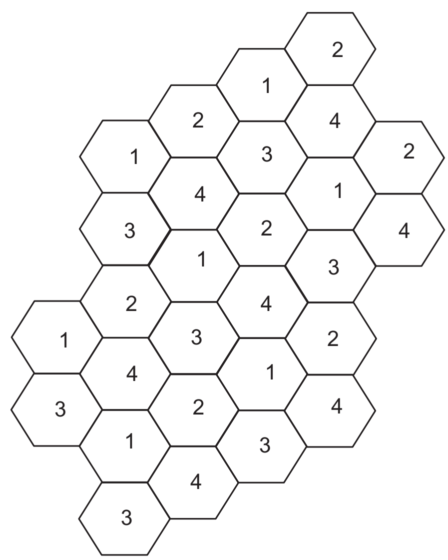

Fig. 4.19 | *For P4.7*

P4.8 A cellular system has a total RF bandwidth of 12.5 MHz and a simplex channel spacing of 30 kHz. The system contains 20 control channels. The system is required to cover a total area of 3600 km .

(a) Calculate the number of traffic channels/cell, if the cluster size is 9.
(b) Find the reuse distance for Part (a), if the area of each cell is 8 km .

(c) Determine the total number of cells needed to cover the entire area.
(d) How many calls can be simultaneously processed by each cell if 8 users can share each channel?

P4.9 Consider two different cellular systems *A* and *B*, each allocated a spectrum of 20 MHz and use 30 kHz channel spacing for uplink and downlink separately. The system *A* is configured with a frequency reuse factor of 4, and the system *B* with frequency reuse factor of 7.

(a) Suppose that in each of these systems, the cluster of cells is duplicated 16 times. Determine the number of simultaneous communications that can be supported by each system.
(b) Find the number of simultaneous communications that can be supported by a single cell in each system.
(c) What is the area covered by each system in terms of cells?
(d) Suppose the cell size is the same in both systems and a fixed area of 100 cells is covered by each system. Compute the number of simultaneous communications that can be supported by each system.

P4.10 A cellular system is allocated a total band of 33 MHz and uses two 25-kHz simplex channels to provide full duplex voice and control channels.

(a) Find the number of channels available per cell for a frequency reuse factor of (i) 4 cells, (ii) 7 cells, and (iii) 12 cells?
(b) Assume that 1 MHz is dedicated to control channels but only one control channel is needed per cell. Determine a reasonable distribution of voice channels and control channels in each cell for frequency reuse factors given in Part (a).

P4.11 Prove the following for a hexagonal cellular system with cell radius *R*, reuse distance *D* and given value of the cluster size K:

(a) $D = 3R$ for $K = 3$
(b) $D = 3.46R$ for $K = 4$
(c) $D = 4.6R$ for $K = 7$
(d) $D = 6R$ for $K = 12$

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5

Cellular Antenna System Design Considerations

In wireless communications, the signals are transmitted in the form of electromagnetic waves through unguided space (wireless) medium. Antennas couple electromagnetic energy from a coaxial cable connected to the transmitter to space for radiating the signal and from space to a coaxial cable connected to the receiver. This chapter begins with the revision of the general characteristics of an antenna, followed by specific requirements of cell-site and mobile antennas. The critical analysis of omnidirectional and directional antenna system design for a cellular system is detailed here. The concept of cell-sectoring is introduced which improves the signal quality. Finally, the effects of different antenna parameters on the cellular system performance are briefly discussed. The emphasis here is to appreciate the vital role played by antenna-system design in a cellular system.

5.1 ANTENNA CHARACTERISTICS

The antenna is an interface between an RF cable connected to transmitter/receiver units and the space. The primary function of a transmitting antenna is to convert the electrical energy (in the form of electric field between the conductors and the magnetic field surrounding them) traveling along an RF cable from a transmitter unit into electromagnetic waves in space. At the receiving antenna, the electric and magnetic fields in space cause current to flow in the conductors that make up the antenna and some of this energy is thereby transferred to the RF cable connected to it and the receiver unit.

In general, antennas are passive devices, which mean the power radiated by a transmitting antenna cannot be greater than the power entering from the transmitter. In fact, the radiating power is always less than the power at its input because of losses. It should be recalled that antenna gain in one direction results from a concentration of power and is accompanied by a loss in other directions. Secondly, antennas are reciprocal devices; that is, the same antenna design works equally well as a transmitting or a receiving antenna with the same amount of gain.

Various types of antennas differ in the amount of radiation they emit in various directions. An isotropic antenna is defined as a hypothetical loss-less antenna having equal radiation in all directions. The actual radiation pattern for the isotropic antenna is a sphere with the antenna at the center. However, radiation patterns are almost depicted as a two-dimensional cross-section of the three-dimensional pattern. An isotropic radiator is taken as a reference for expressing the directional properties of actual antennas. A directional antenna is one which has the property of radiating or receiving electromagnetic waves more effectively in some directions than in others.

Omnidirectional antennas allow transmission of radio signals with equal signal power in all directions. It is difficult to design such antennas, and most of the time, an antenna covers an area of 120 degrees or 60 degrees. These are called directional antennas, and cells served by them are called *sectored cells*. From a practical point of view, many sectored antennas are mounted on a single microwave tower located at the centre of the cell. An adequate number of antennas are placed at the cell-site to cover the whole 360 degrees of the cell. In practice, the effect of an omnidirectional antenna can be achieved by employing several directional antennas to cover the whole 360 degrees. The advantages of sectoring are that it requires coverage of smaller area by each antenna and hence requires lower power to transmit radio signals. It also helps in reducing cochannel interference in a cellular architecture, and thereby enhancing the spectrum efficiency of the overall system.

> **Facts to Know!**
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> An isotropic radiator is a theoretical perfect sphere that radiates power equally in all directions. The closest thing to an isotropic radiator is the sun. It is not possible to build a real isotropic radiator because it would need a power cable connected to it at some point on the surface of the sphere.

A brief description of the related terms and the means to quantify the basic characteristics of antennas is presented here.

Antenna Radiation Pattern or Antenna Pattern It is defined as a mathematical function or graphical representation of the radiation properties of the antenna as a function of space coordinates. An antenna will radiate power in all directions but, typically, does not perform equally well in all directions. The radiation pattern characterises this performance of the antenna. Refer Fig. 5.1.

Field Pattern A graph of the spatial variation of the electric or magnetic field along a constant distance path is called a *ie p tte* . The distance from the antenna to each point on the radiation pattern is proportional to the power radiated from the antenna in that direction. The linear dipole is an example of an omnidirectional antenna, that is, an antenna having a radiation pattern which is nondirectional in a plane.

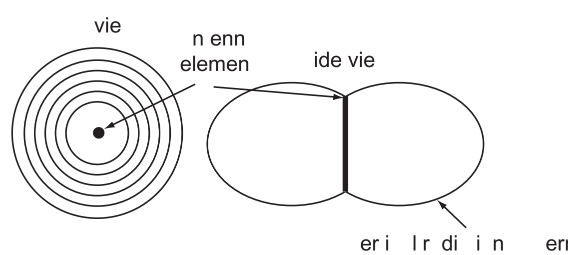

Fig. 5.1 *Omni-directional antenna radiation patterns*

EIRP The power radiated within a given geographic area is usually specified either with reference to isotropic antenna or an omnidirectional dipole antenna. The effective isotropic radiated power (EIRP) is referenced to an isotropic antenna. The effective radiated power (ERP) is referenced to an omnidirectional antenna. The ERP is greater than EIRP by 2 dB approximately.

Directivity The directivity of a transmitting antenna is defined as the ratio of the radiation intensity flowing in a given direction to the radiation intensity averaged over all directions. Directivity is sometimes referred to as *antenna gain*. It is important to note that antenna gain does not refer to obtaining more output power than input power but refers to directionality.

Absolute Gain The absolute gain of a transmitting antenna in a given direction is defined as the ratio of the radiation intensity flowing in that direction to the radiation intensity that would be obtained if the power accepted by the antenna were radiated isotropically. Note that absolute gain is closely related to directivity, but it takes into account the efficiency of the antenna as well as its direction characteristics. To distinguish it, the absolute gain is sometimes referred to as *power gain*.

Relative Gain The relative gain of a transmitting antenna in a given direction is defined as the ratio of the absolute gain of the antenna in the given direction to the absolute gain of a reference antenna (a perfect omni-directional isotropic antenna) in the same direction. The power input to the two antennas must be the same.

Facts to Know!

Antenna gain is directional gain, not power gain, due to focusing of the radiated energy in a specified direction.

Efficiency The efficiency of a transmitting antenna is the ratio of the total radiated power radiated by the antenna to the input power to the antenna.

The same antenna can be used to transmit and receive the radio signals due to the principle of reciprocity, which may be stated as 'signal transmission over a radio path is reciprocal considering that the locations of the transmitter and receiver can be interchanged without changing the wireless transmission characteristics'. This principle is based on Maxwell's equations of electro-magnetic theory, which implies that if the direction of propagation of the electromagnetic signal is reversed, the energy of the signal would follow exactly the same radio path and encounter the same propagation effects but in the reverse direction. This also implies that the transmit and receive antenna gains are equal. However, transmitter antenna gain is a measure of how well it emits the radiated energy in a certain area, while the receiver antenna gain is a measure of how well it collects the radiated energy in that area.

Effective Area The effective area or aperture of a receiving antenna in a given direction is defined as the ratio of the available power at the terminals of the antenna to the radiation intensity of a plane wave incident on the antenna in the given direction. In fact, the effective area of an antenna is related to the physical size of the antenna and its shape. The relationship between antenna gain and effective area is

$$G_r = (4\pi A_{eff}) / \lambda_c^2 \tag{5.1}$$

where G_r = receiver antenna gain
A_{eff} = effective area
λ_c = carrier wavelength

Antenna Factor It is the ratio of the magnitude of the electric field incident upon a receiving antenna to the voltage developed at the antenna's output connector (assuming a 50-ohm coaxial connector). The antenna factor is clearly related to the gain of the antenna, but is often found to be the most convenient parameter for use in the monitoring of electromagnetic emissions.

Radiation Resistance The radiation resistance of a half-wave dipole antenna situated in free-space and fed at the centre is approximately 70 Ω. The impedance is completely resistance at resonance, which usually occurs when the length of the antenna is about 95% of the calculated free-space half-wavelength value. The exact length depends on the diameter of the antenna conductor relative to the operating wavelength.

Polarisation The polarisation of a radio wave is the orientation of its electric field vector. It could be horizontal or vertical or hybrid. It is important that the polarisation should be the same at both ends of a communication link. Wireless communication systems usually use vertical polarisation because this is more convenient for use with portable and mobile antennas.

Front-to-back Ratio The direction of maximum radiation in the horizontal plane is considered to be the front of the antenna, and the direction 180 degree from the front is considered to be its back. The ratio between the gains to the front and back lobes is the front-to-back ratio. It is generally expressed in dB. It is a measure of the antenna's ability to focus radiated power in the intended direction successfully, without interfering with other antennas behind it. In other words, it is the ratio of radiated power in the intended direction to radiated power in the opposite direction. The dipole antenna viewed in the horizontal plane has two equal lobes of radiation. The more complex antenna, however, has one major lobe and a number of minor lobes. Each of these lobes has a gain and beamwidth.

5.2 ANTENNAS AT CELL-SITE

The antennas used for cellular radio systems have to fulfill the requirement of not only providing adequate radio coverage but also reducing cochannel interference arising due to frequency reuse in cellular architecture. So there is a need for high-gain (usually 6-dB or 9-dB) omnidirectional antennas (a picture of typical antenna shown alongside), and antennas with beamwidths of 60 degrees or 120 degrees for sectorised cells. Narrower beamwidths are sometimes needed for filling weak or dead signal spots. Typical cellular antennas use variations of the collinear antennas backed by reflectors for omnidirectional patterns or log-periodic antennas for directional patterns.

A typical antenna

Cellular base-station receiving antennas are usually mounted in such a way so as to obtain space diversity. Space-diversity antennas can separate only horizontally, not vertically. Two-branch space-diversity antennas are used at the cell-site to receive the same signal with different fading envelopes, one at each antenna. The degree of correlation between two fading envelopes is determined by the degree of separation between two receiving antennas. When the two fading envelopes are combined, the degree of fading is reduced.

When cell sizes are small as in high-traffic areas, directional base-station antennas are often tilted downwards in order to reduce interference to neighbouring cells, thereby reducing cochannel interference. Of course this is accompanied with slight reduction in its coverage area that can be compensated with higher transmitting power. This downtilt can either be done mechanically by mounting the antenna so that it aims downwards as a slight angle of the order of 9 degrees or so. It can also be done electronically. Figure 5.2 shows a typical picture of a cell-site panel antenna.

Typically, the separation '*s*' between two receiving antennas for a cellular system operating at 900 MHz should be greater than or equal to 8 λ_c for an antenna height of 30 metres and the separation of greater than or equal to 14 λ_c for an antenna height of 50 metres. For example, at 900 MHz, the separation of 8 λ_c between two receiving antennas creates a 2 dB difference between their received signal levels, which is tolerable for the advantageous use of a diversity scheme. Figure 5.3 shows the diversity antenna spacing at the cell-site.

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A minimum separation between two receiving antennas is necessary to reduce the antenna pattern ripple effects.

In any omnicell system, the two space-diversity receiving antennas should be aligned with the surrounding terrain. Figure 5.4 shows a typical cell-site antenna mounting arrangement.

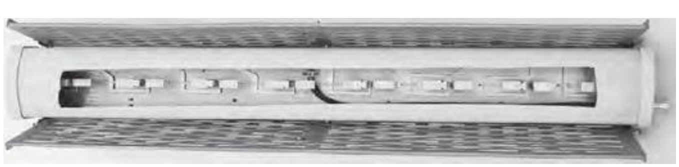

Fig. 5.2 *Cell-site panel antenna (with cutout to show internal construction)*

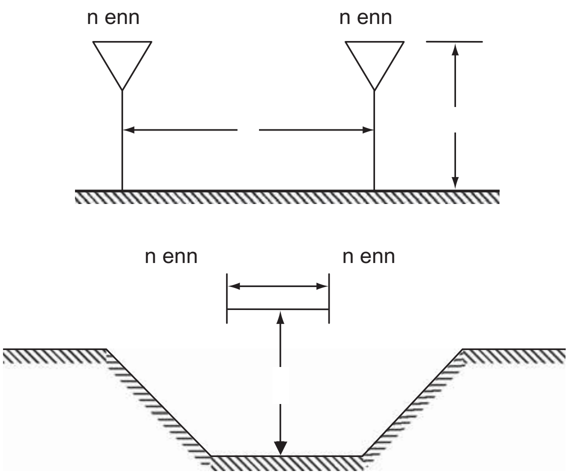

Fig. 5.3 | *Diversity antenna-mounting arrangement*

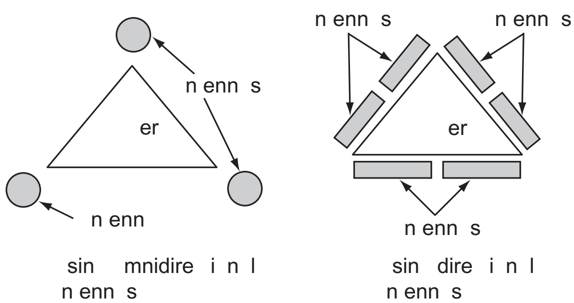

Fig. 5.4 | *Cell-site antenna mounting*

- For an omnidirectional radiation pattern, generally three receiving antennas are mounted at the corners of a tower with a triangular cross-section, as shown in Fig. 5.4(a).
- When the cell is divided into three sectors, it is common practice to mount two receiving antennas for each sector on the sides of the tower, as shown in Fig. 5.4 (b).

The minimum separation between a transmitting antenna and a receiving antenna is necessary to avoid receiver desensitisation and the intermodulation problem. However, it is possible to use a single antenna for both transmitting and receiving using a duplexer.

Figure 5.5 shows a picture of a cell-site antenna tower with various antennas mounted on it.

But in practice, the transmitting antenna is located separately at the cell-site. Only one transmitting antenna is needed with an omnidirectional radiation pattern; otherwise, one transmitting antenna is needed for each sector of a cell. Antennas can also be suitably mounted on the topside walls of the buildings where towers cannot be installed.

At the cell-site, diversity can be achieved by using two polarisations, typically at 45-degree angles to the vertical. With the use of dual-polarisation antennas, the number of antennas required for diversity can be reduced. This arrangement takes into consideration the fact that the signal polarisation may be randomised by reflections in a mobile environment. Dual-polarisation antennas can considerably reduce the number of visible structures needed at a cell site. Portable and mobile units can still use vertical polarised antennas.

When the cell-site antenna is placed in an urban or suburban environment and the mobile antenna is lower than the heights of the surroundings, the cell-site antenna pattern is quite different from that of free-space antenna pattern and is distorted, although the strongest signal reception area still coincides with the strongest signal strength of the directional antenna. In such an operating environment, the front-to-back ratio as well as beamwidth is less than the specified values. For example, for a 120 directional antenna, the backlobe (or front-to-back ratio) is about 10 dB less than the front lobe, regardless of whether a weak sidelobe pattern or no sidelobe pattern is designed in a free-space operating condition. This condition exists because the strong signal radiates in front, bouncing back from the nearby surrounding buildings so that the energy can be received from the back of the antenna.

> **Facts to Know!**
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> If the buildings are far away from the directional antenna, then the front-to-back ratio measured in the field will be quite close to the specified antenna pattern, usually 20 dB.

In antenna-site selection, reduction of interference is an important factor while meeting the radio-coverage patterns.

When a cell-site is chosen on the map, there is a 50% chance that the site location can be acquired. So an antenna location can be found within a quarter of the size of the cell $R/4$ where R is the radius of the cell. This is termed as the quarter-radius rule. For example, if the radius of a cell is 12 km, the cell-site antenna can be located within a 3-km radius ($R/4$). The change in the cell-site antenna within a 3-km radius around the centre of the cell would not affect the coverage pattern at a distance 12 km away. If the radius of the cell is 3 km then the cell-site antenna can be located within a 0.75-km radius from its centre. However, the quarter-radius rule can be applied only on relatively flat terrain, not in a hilly area.

There is a need to check the cell-site antennas regularly. Air-pressurised RF coaxial cable is generally used to connect the base-station equipment with the cell-site antennas. This enables to prevent moisture from entering the RF coaxial cable and causing excessive signal attenuation. The RF power delivered to the antenna terminal can be checked with the help of an *R Power Meter*. Alternatively, VSWR can be checked at the bottom of the RF cable antenna at the output of the base-station transmitter equipment. In this case, the loss of reflected power due to the cable itself should also be considered. However, the VSWR reading may not be accurate for a high tower due to cable connector losses. For example, if each cable connector has 1-dB loss due to energy leakage at the interconnecting point and two numbers of midsection 1-dB loss for each RF adaptor used in the transmitter system then the reflected power indicated in the VSWR meter would be 4 dB less than the real reflected power.

Fig. 5.5 *Cell-site antenna tower with various antennas mounted on it*

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To determine whether the quarter-radius rule can be applied in a particular service area or not, the point-to-point signal prediction model for estimation of propagation path loss should be used to plot the radio coverage at different possible cell-site locations. Care should be taken so that the differences in the observations should be minimal.

EXAMPLE 5.1 RF cable loss in cell-site antenna system

A cell-site transmitter generates a +15 dBm RF signal and is connected to an antenna using an RF coaxial cable that induces a 3 dB loss. The cable has two connectors at its either end that induce a loss of 2 dB each. What is the signal level at the input of the antenna?

Solution

The RF output signal level of cell-site transmitter = +15 dBm (given)

Step 1. *To determine total loss in cable connectors*

Signal loss due to one connector of RF coaxial cable = 2 dB (given)

Number of connectors on a coaxial cable = 2

Therefore, signal loss due to both connectors = $2 \times 2 = 4$ dB

Step 2. *To determine signal loss due to cable and connectors*

Signal loss due to RF coaxial cable = 3 dB (given)

Total signal loss due to cable and connectors = 3 dB + 4 dB = 7 dB

Step 3. *To determine signal level at the input of the antenna*

Signal level at the input of the antenna = +15 dBm – 7 dB

Hence, signal level at the input of the antenna = +8 dBm

In certain situations such as controlling the radiated energy in a confined area, a *discone antenna* or a *biconical antenna* is used at the cell-site. A biconical antenna is the one in which one of the cones is extended to 180 to form a disk. The diameter of the disk, the length of the cone, and the opening of the cone can be suitably adjusted to create an umbrella-type radiation pattern. Using a monopole with a top disk used at the cell-site can also develop the umbrella-pattern antenna. The size of the disk determines the tilting angle of the radiation pattern. The smaller the disk, the larger the tilting angle of the umbrella pattern. A high-gain discone antenna can be constructed by vertically stacking a number of umbrella-pattern antennas. A *parasitic antenna* is used to reduce interference in critical directions. The parasitic (insulation) element is about 1.05 times longer than the active element.

5.3 MOBILE ANTENNAS

A very common configuration of mobile antenna consists of a $\lambda/4$ antenna connected by a coil that matches impedances with a $\lambda/2$ antenna mounted collinearly above it. This arrangement has a gain of about 3 dB compared with the quarter-wave monopole which is normally used with portable phones. Figure 5.6 shows a picture of typical omnidirectional mobile antennas with mounts.

Fig. 5.6 *Blister-type omnidirectional mobile antennas*

The requirement of a mobile (vehicle-mounted) antenna is an omnidirectional antenna because in a mobile radio environment, the scattered signals arrive at the mobile unit from every direction with equal probability and from different elevation angles (the elevation angle for scattered signals received in urban areas is greater than that in suburban areas). At the most, a 2- to 3-dB gain antenna might be used at the mobile for the purpose of enhancing reception level because the cell-site antenna is rarely as high as the broadcasting antenna and out-of-sight conditions often prevail. The mobile antenna with a gain of more than 3 dB can receive only a limited portion of the total multipath signal in the elevation as measured under the out-of-sight condition.

The mobile antenna should be ideally located as high as possible from the point of reception. However, the physical limitation of antenna height on the vehicle restricts this requirement. Generally, the antenna should at least clear the top of the vehicle. The mobile antenna can either be roof-mounted or glass-mounted in the motor vehicle. The radiation pattern of a roof-mounted antenna is more or less uniformly distributed around the mobile unit when measured at an antenna range in free space. Figure 5.7 depicts a scenario in which a roof-mounted mobile antenna receives multipath signals from the cell-site at different vertical angles.

In glass-mounted mobile antennas, there is no need to drill a hole in the side-window glass of the vehicle. The signal energy

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Omnidirectional antennas have a gain because they emit or radiate a signal in two dimensions only, not in three dimensions such as an isotropic radiator would radiate, if it could be built.

is coupled through the glass with slight loss of energy through coupling. It is clear that the position of the glass-mounted antenna is always lower than that of the roof-mounted antenna that may result into 3-dB difference between these two types of mobile antennas. Also, glass-mounted antennas cannot be installed on the shaded glass found in some motor vehicles because this type of glass has a high metal content. A high-gain omnidirectional antenna is different from a directional antenna having the same gain because in a high-gain omni-antenna, the radiation pattern is suppressed horizontally whereas in the directional antenna, the antenna beam pattern is suppressed horizontally.

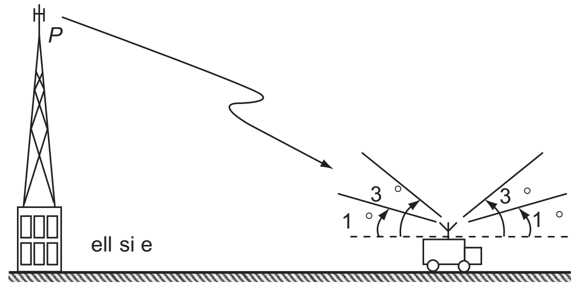

Fig. 5.7 *Roof-mounted omnidirectional mobile antenna*

In order to reduce the effect of fading and interference, a two-branch space-diversity antenna can be mounted on a motor vehicle either horizontally oriented or vertically oriented. The two vehicle-mounted antennas can be separated horizontally by 0.5λ, or vertically by 1.5λ on the rooftop of the vehicle. Theoretical analysis and measured data show that in horizontally oriented space-diversity mobile antennas, the in-line arrangement of the two antennas exhibit less fading and therefore recommended as compared to perpendicular arrangement to the direction of motion of the vehicle.

> **Facts to Know!**
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> The mobile antenna-gain range is 1 to 3 dB depending on the operating frequency. A high-gain omnidirectional mobile antenna can also be used on the mobile unit.

5.4 DESIGN OF OMNIDIRECTIONAL ANTENNA CELLULAR SYSTEM

The carrier-to-interference (C/I) ratio requirement is determined by the transmission and modulation scheme adopted in a system and is specified for an acceptable voice quality. A voice quality is termed as acceptable if 10–15% of the mobile subscribers rate the received signal quality as good or excellent. The parameter C/I is generally determined based on the subjective voice quality tests.

Let N_I be the number of cochannel interfering cells and I_i be the interference power caused by transmissions from the i^{th} interfering cochannel cell. The signal-to-cochannel interference ratio (C/I) at the desired mobile receiver is given by

$$\frac{C}{I} \approx \frac{C}{\sum_{i \to 1}^{N_I} I_i} \tag{5.2}$$

In addition to cochannel interference, there is always the inherent background noise. However, in an interference dominated mobile environment, we may neglect the background noise. It may be noted that the desired received signal power C is proportional to $r^{-\gamma}$, where r is the distance between the mobile and the serving base station and γ is a propagation path-loss exponent determined by the actual terrain environment and varies in the range $2 \leq \gamma \leq 5$. When the transmit powers from all base stations are equal and the path-loss exponent is the same throughout the geographical coverage area, the cochannel interference from the i^{th} cochannel cell, I_i, for all i, depends on D_i and γ only, D_i being the distance between the i^{th} interfering cochannel cell and the mobile unit.

When the mobile unit is located at the cell boundary having radius R (i.e., $r = R$), the worst-case cochannel interference occurs as the power of the desired signal is minimum. Since the

> **Facts to Know!**
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> The degree of cochannel interference is a function of the location of the mobile unit within the cell of the serving base station in a cell.

average received signal strength at any point decays as a power law of the distance between the transmitter and the receiver (C varies as $R^{-\gamma}$ and I varies as $D^{-\gamma}$), the C/I at a typical mobile receiver can be approximated by

$$\frac{C}{I} \approx \frac{R^{-\gamma}}{\sum_{i \to 1}^{N_I} D^{-\gamma}} \tag{5.3}$$

Assuming as a simple case, $D_i \approx D$ for $I = 1, 2, \ldots, N_I$, in (Eq. 5.3), we get

$$\frac{C}{I} \approx \frac{R^{-\gamma}}{N_I D^{-\gamma}}$$

Or,

$$\frac{C}{I} \approx \frac{D^{\gamma}}{N_I R^{\gamma}}$$

Thus, for a minimum reuse distance D and the cell radius R, the carrier-to-interference ratio due to cochannel interference is given as

$$\frac{C}{I} \approx \frac{1}{N_I} \cdot \left(\frac{D}{R}\right)^{\gamma} \tag{5.4}$$

The separation between cochannel cells, D is mainly determined by C/I that is required to achieve the desired transmission quality and the fade margin that is necessary to take care of statistical fluctuations in the desired received signal level induced by the mobile environment. Since by definition, frequency reuse ratio, $q = D \quad R$

therefore,

$$\frac{C}{I} \approx \frac{1}{N_I} \cdot (q)^{\gamma} \tag{5.5}$$

In a mobile radio environment, γ is usually taken as 4. In a fully developed hexagonal cellular system, the number of cochannel interfering cells is six in the first tier, ignoring cochannel interfering cells in the second tier, which cause weaker interference than those in the first tier. Therefore, putting $N_I = 6$ and $\gamma = 4$ in Eq. (5.5), we get

$$\frac{C}{I} \approx \frac{1}{6} \cdot (q)^{4} \tag{5.6}$$

Or,

$$q = \left(6 \times \frac{C}{I}\right)^{1/4} \tag{5.7}$$

EXAMPLE 5.2 Acceptable *C/I* vs cluster size K

(a) Determine the minimum cluster size for a cellular system designed with an acceptable value of signal-to-cochannel interference ratio C/I = 18 dB. Assume the path-loss exponent as 4 and cochannel interference at the mobile unit from six equidistant cochannel cells in the first tier.

(b) If the acceptable C/I is enhanced to 20 dB, will the cluster size determined in (a) be adequate? If not, then what should be the cluster size?

Solution

(a) *To determine the minimum cluster size, K*

Signal-to-cochannel interference ratio, $C/I = 18$ dB (given)

Step 1. *To convert given C/I in dB to C/I in ratio*

We know that C/I (dB) $= 10 \log_{10} C/I$ (ratio)

Therefore, 18 dB $= 10 \log_{10} C/I$ (ratio)

Or, C/I (ratio) $= 10^{1.8} = 63.1$

Step 2. *To determine the frequency reuse ratio, q*

We know that

$$q = \left(6 \times \frac{C}{I}\right)^{1/4}$$

$q = (6 \times 63.1)^{1/4} = 4.41$

Step 3. *To determine the cluster size, K*

We know that $q = \sqrt{3K}$

Or, $K = (q) / 3$

Or, $K = (4.41) / 3$

Or, $K = 6.48$

Hence, the nearest valid cluster size, $K = 7$

(b) *To determine C/I for K = 7 and new cluster size*

Step 4. *To determine q for K = 7*

We know that $q = \sqrt{3K}$

For $K = 7$, $q = \sqrt{(3 \times 7)} = 4.6$

Step 5. *To determine C/I for K = 7*

We know that in six equidistant cochannel cells in the first tier,

$$\frac{C}{I} = \frac{1}{6}.(q)^4$$

Or, $$\frac{C}{I} = \frac{1}{6}.(4.6)^4$$

Hence, C/I (ratio) $= 74.2$

Step 6. *To convert C/I (ratio) in C/I (dB)*

We know that C/I (dB) $= 10 \log_{10} C/I$ (ratio)

Or, C/I (dB) $= 10 \log_{10} 74.2$

Hence, C/I (dB) $= 18.73$ dB

Step 7. *To determine whether K = 7 is adequate*

Acceptable $C/I = 20$ dB (given)

The required C/I of 18.73 dB is less than the acceptable value of C/I of 20 dB for the given situation.

Hence, $K = 7$ cannot meet the desired / requirement.

Step 8. *To convert given C/I in dB to C/I in ratio*

We know that C/I (dB) $= 10 \log_{10} C/I$ (ratio)

$20 \text{ dB} = 10 \log_{10} (C/I)$

Or, C/I (ratio) $= 10^{2} = 100$

Step 9. *To determine new frequency reuse ratio, q*

The required q corresponding to $C/I = 20$ dB or 100 can be determined from the relationship

$$q = \left(6 \times \frac{C}{I}\right)^{1/4}$$

$q = (6 \times 100)^{1/4} = 4.95$

Step 10. *To determine new cluster size, K for C/I = 20 dB*

We know that $K = (q)\,/3$

Or, $K = (4.95)\,/3 = 8.165 \approx 9$

Hence, the nearest valid cluster size, $K = 9$

EXAMPLE 5.3 | ***C/I* in a hexagonal cellular architecture**

Derive and establish the following relationship between C/I and K:

$$C/I = 1.76 + 20 \log (K)$$

Assume that there are only six cochannel interferers in the first tier, which are equidistant from the mobile, and the path-loss exponent as 4. Discuss the interpretation of the expression with the help of a graphical representation.

Solution

Since there are six equidistant cochannel interferers in the first tier, and the path-loss exponent is given as 4, C/I can be given by the expression

$$\frac{C}{I} = \frac{1}{6}.(q)^4$$

Step 1. *To establish relationship between C/I and K*

Substituting $q = \sqrt{3K}$, we get

$$\frac{C}{I} = \frac{1}{6}.\left(\sqrt{3\mathrm{K}}\right)^4$$

$C/I = (3K)\,/6$

Or, $C/I = (9K\,)/6$

Or, $C/I = 1.5K$

Step 2. *To show that C/I = 1.76 + 20 log (K)*

Taking log on both sides, we get

C/I (dB) $= 10 \log (1.5) + 10 \log (K\ )$

Hence, $\qquad = 1.76 + 20 \log (K) \qquad (5.8)$

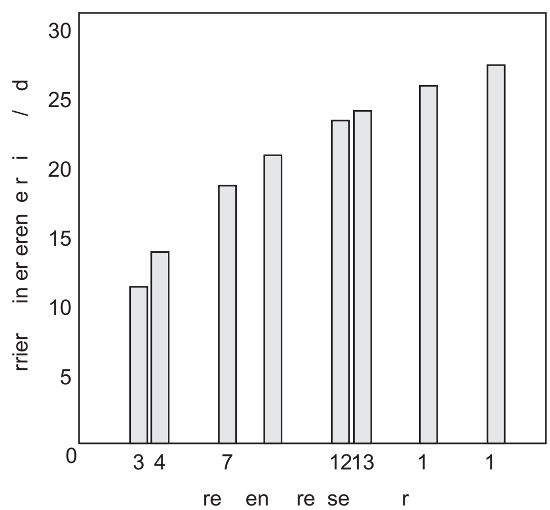

Fig. 5.8 | *Graphical representation of C/I vs. K*

Step 3. *Interpretation of the expression*

Equation 5.8 shows how the carrier-to-interference ratio varies with the cluster size K under the assumption that the mobile is equidistant from all six cochannel interferers in the first tier in a mobile radio environment. This equation is commonly used to determine the cluster size for desired performance of a cellular system.

Step 4. *Graphical representation of C/I versus K*

Figure 5.8 shows a graphical representation of the variation of C/I with the cluster size K.

As an example, for a C/I ratio of 18 dB, the serving signal must be 18 dB greater than the interfering signal. Therefore, the greater the distance between the reusing cell sites and the higher value of K for the same cell radius, the lower the interfering (I) value. This will yield an improved C/I, assuming that there is sufficient signal coverage.

Worst Case Analysis of C/I in an Omnidirectional Antenna System

Figure 5.9 illustrates a serving cell C, surrounded by six equidistant cochannel interfering cells $C1$, $C2$, $C3$, $C4$, $C5$, and $C6$ in its first tier.

Consider that the mobile is at the cell boundary (cell radius R) of its serving cell C, where it experiences worst-case cochannel interference on the forward channel. At this location, the mobile unit would receive the weakest signal from its own cell-site but strong interference from all co-channel interfering cell sites. From Fig. 5.9, it can be seen that the distance from all the six cochannel interfering cells in the first tier of $K = 7$ frequency reuse pattern would be (D is the distance between the centre of any two cochannel cells)

- Two distances of $D - R$ (that is, $D5$ and $D6$),
- Two distances of D (that is, $D1$ and $D4$), and
- Two distances of $D + R$ (that is, $D2$ and $D3$).

Then from Eq. (5.3), the C/I ratio can be expressed as under:

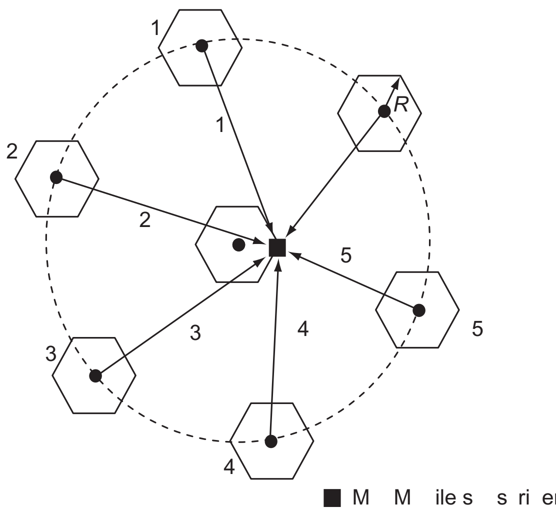

Fig. 5.9 *Worst-case cochannel interference in the first tier*

$$\frac{C}{I}(\text{omni}) \approx \frac{R^{-\gamma}}{2(D-R)^{-\gamma} + 2D^{-\gamma} + 2(D+R)^{-\gamma}} \tag{5.9}$$

Using $q = D/R$ and simplifying it, we get

$$\frac{C}{I}(\text{omni}) \approx \frac{1}{2(q-1)^{-\gamma} + 2(q)^{-\gamma} + 2(q+1)^{-\gamma}} \tag{5.10}$$

EXAMPLE 5.4 | Worst-case C/I in omnidirectional antenna system

Determine the signal-to-cochannel interference ratio C/I at the mobile receiver located at the boundary of its omnidirectional operating cell, under the influence of interfering signals from six cochannel interfering cells in the first tier in a cellular system designed with

(a) $K = 7$
(b) $K = 9$
(c) $K = 12$

Assume the path-loss exponent is 4. What would be the correct choice of K for a system requiring C/I = 18 dB?

Solution

Configuration of cellular antenna design = omnidirectional
Number of cochannel interferers = 6 (given)
The path-loss exponent, $\gamma = 4$ (given)

(a) *To determine the C/I at the mobile receiver for K = 7*

Step 1. *To determine frequency reuse ratio, q*

We know that

$$q = \sqrt{3K}$$

For $K = 7$, $q = \sqrt{21} = 4.6$

Step 2. *To determine the C/I in the worst-case analysis*

In an omnidirectional antenna design of a cellular system based on regular hexagonal pattern, the worst-case C/I, under the influence of interfering signals from six cochannel interfering cells in the first tier, is given by the relationship

$$\frac{C}{I}(\text{omni}) \approx \frac{1}{2(q-1)^{-\gamma} + 2(q)^{-\gamma} + 2(q+1)^{-\gamma}}$$

By substituting $q = 4.6$ and $\gamma = 4$, we get

$$\frac{C}{I}(\text{omni}) \approx \frac{1}{2(4.6-1)^{-4} + 2(4.6)^{-4} + 2(4.6+1)^{-4}}$$

Or, C/I (omni) = 54

Step 3. *To convert C/I (ratio) in C/I (dB)*

We know that C/I (dB) = 10 $\log_{10}$ C/I (ratio)
Or, C/I (dB) = 10 log 54
Hence, / (dB) = 17 dB

(b) *To determine the C/I at the mobile receiver for K = 9*

Step 4. *To determine frequency reuse ratio, q*

$$q = \sqrt{3K}$$

$$q = \sqrt{27} = 5.2$$

Step 5. *To determine the C/I in the worst-case analysis*

By substituting $q = 5.2$ and $\gamma = 4$ in the relationship,

$$\frac{C}{I}(\text{omni}) \approx \frac{1}{2(q-1)^{-\gamma} + 2(q)^{-\gamma} + 2(q+1)^{-\gamma}}$$

Therefore,

$$\frac{C}{I}(\text{omni}) \approx \frac{1}{2(5.2-1)^{-4} + 2(5.2)^{-4} + 2(5.2+1)^{-4}}$$

Or, C/I (omni) = 84.5

Step 6. *To convert C/I (ratio) in C/I (dB)*

C/I (dB) = 10 log 84.5
Hence, / (dB) = 19.2 dB

(c) *To determine the C/I at the mobile receiver for K = 12*

Step 7. *To determine frequency reuse ratio, q*

$$q = \sqrt{3K}$$

$$q = \sqrt{36} = 6$$

Step 8. *To determine the C/I in the worst-case analysis*

By substituting $q = 6$ and $\gamma = 4$ in the relationship

$$\frac{C}{I}(\text{omni}) \approx \frac{1}{2(q-1)^{-\gamma} + 2(q)^{-\gamma} + 2(q+1)^{-\gamma}}$$

Therefore,

$$\frac{C}{I}(\text{omni}) \approx \frac{1}{2(6-1)^{-4} + 2(6)^{-4} + 2(6+1)^{-4}}$$

Or, C/I (omni) = 179.3

Step 9. *To convert C/I (ratio) in C/I (dB)*

C/I (dB) = 10 log 179.3

Hence, / (dB) = 22.5 dB

Step 10. *Correct choice of the value of K*

Required value of C/I = 18 dB (given)

From the above results, C/I = 17 dB for K = 7, which is lower than the desired value of C/I = 18 dB. So the system design for K = 7 is imperfect.

It is observed that C/I = 19.2 dB for K = 9 as well as C/I = 22.5 dB for K = 12, which are higher than the desired value of C/I = 18 dB. Therefore in an omnidirectional cellular system K = 9 or K = 12 cell patterns would be a correct choice.

5.5 DESIGN OF DIRECTIONAL ANTENNA CELLULAR SYSTEMS

As seen in the previous section, increasing the value of K can increase C/I. When K increases, the number of frequency channels available in a cell decreases for a given frequency spectrum. This results in overall decrease of spectrum efficiency that is contrary to the advantage of frequency-reuse scheme in a cellular system. So it is needed that when the call traffic increases, the frequency spectrum efficiency should be enhanced without increasing the cluster size K.

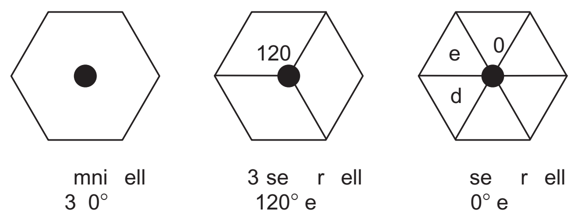

Fig. 5.10 *Sectoring of cells with directional antennas*

Instead of increasing the cluster size K beyond 7, a suitable arrangement of directional antennas in place of an omnidirectional antenna at each cell-site can be deployed to reduce cochannel interference, and thereby increasing C/I. This simply means that each omni-cell is divided into three or six sectors radially while retaining the same number of channels per cell in the 7-cell cluster and accordingly uses three or six directional antennas at a cell-site. This technique reduces cochannel interference. Figure 5.10 illustrates an omnicell (360°) and sectoring of cells having three (120° each), and six sectors (60° each).

Facts to Know!

Each sector is assigned a set of frequency channels. Sectored cells use channel-sharing and channel-borrowing schemes to increase spectrum efficiency.

5.5.1 Three-sector Cellular System Design

Figure 5.11 depicts a three-sector per cell configuration in a seven-cell reuse pattern. The mobile unit situated at the boundary of its operating cell, $C1$ (with radius R) in the worst case will now experience co-channel interference from corresponding cochannel sectors $S1$ of two cochannel cells ($C2$ and $C3$) out of six cochannel interfering cells. This is mainly due to the fact that the interference is effective in only the forward direction because the front-to-back ratio of a cell-site directional antenna is at least 10 dB or more in a mobile radio environment.

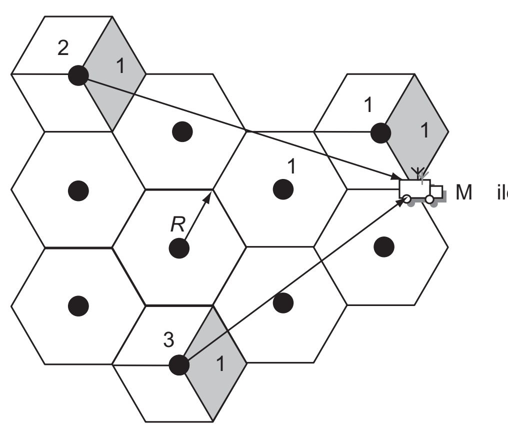

Fig. 5.11 *Worst case interference in 3-sector cellular system*

The distance from two cochannel interfering sectors of cochannel cells in the first tier of $K = 7$ frequency reuse pattern would be $D1$ and D, where D is the distance between two cochannel cells. From the hexagonal geometry, it can be computed that $D1$ is equal to $D + 0.7R$.
Then from Eq. (5.3) the C/I ratio can be expressed as under:

$$\frac{C}{I}(3-\text{sector}) \approx \frac{R^{-\gamma}}{D^{-\gamma} + (D + 0.7R)^{-\gamma}} \tag{5.11}$$

Using $q = D/R$,

$$\frac{C}{I}(3-\text{sector}) \approx \frac{1}{(q)^{-\gamma} + (q + 0.7)^{-\gamma}} \tag{5.12}$$

EXAMPLE 5.5 | Worst case *C/I* in a 3-sector cellular system for *K* = 7

A cellular system is designed with a 3-sector directional antenna cellular configuration. A cluster pattern of size 7 is deployed. Let the mobile receiver be located at the boundary of its operating cell, and be under the influence of interfering signals from two cochannel interfering cells in the first tier. Compute the worst-case signal-to-cochannel interference ratio C/I at the mobile receiver. If the C/I value for a practical system requires 6 dB higher than the theoretical value of C/I = 18 dB then comment on the results obtained. Assume the path-loss exponent as 4 in a mobile radio environment.

Solution

Configuration of cellular antenna design = 3-sector (given)
Number of cochannel interferers = 2 (given)
The path-loss exponent, $\gamma = 4$ (given)
Cluster size, K = 7 (given)

Step 1. *To determine frequency reuse ratio, q*

We know that

$$q = \sqrt{3K}$$

For $K = 7$, $q = \sqrt{21} = 4.6$

Step 2. *To determine the C/I in the worst-case analysis*

In a 3-sector directional antenna design of a cellular system based on regular hexagonal pattern, the worst-case C/I, under the influence of interfering signals from two nearest cochannel interfering cells, is given by the relationship

$$\frac{C}{I}(3-\text{sector}) \approx \frac{1}{(q)^{-\gamma} + (q + 0.7)^{-\gamma}}$$

Then by substituting q = 4.6 and $\gamma = 4$, we get

$$\frac{C}{I}(3-\text{sector}) \approx \frac{1}{(4.6)^{-4} + (4.6 + 0.7)^{-4}}$$

Or, C/I (3-sector) = 285

Step 3. *To convert C/I (ratio) in C/I (dB)*

We know that C/I (dB) = 10 log C/I (ratio)
C/I (dB) = 10 log 285
Hence, / (dB) = 24.5 dB

Comment on the Result Thus, the worst-case theoretical value of C/I received by a mobile unit from the 3-sector directional antenna design is 24.5 dB that greatly exceeds the minimum required value of 18 dB.

In a practical scenario under a heavy traffic area conditions as well as because of irregular terrain contour and imperfect site locations, there may be a reduction of 6 dB in the theoretical value of *C/I*. Even then, the remaining *C/I* of (24.5 – 6 =) 18.5 dB is still adequate to ensure desired quality level. Hence, it may be concluded that three-sector cellular design with $K = 7$ meets the desired requirement of *C/I* as compared to that of omnidirectional antenna design.

EXAMPLE 5.6 Worst case *C/I* in 3-sector cellular system for $K = 4$

(a) Compute the worst-case C/I value at the mobile receiver located at the boundary of its serving cell if it is under the influence of interfering signals from two nearest cochannel interfering cells in a cellular system. The system is designed with 3-sector directional antenna cellular system with a reuse pattern of 4.

(b) Does this system yield an adequate value of C/I for a practical system which requires 6 dB higher than the theoretical value of C/I = 18 dB?

Assume the path-loss exponent as 4 in a mobile radio environment.

Solution

Configuration of cellular antenna design = 3-sector (given)
Number of cochannel interferers = 2 (given)
The path-loss exponent, $\gamma = 4$ (given)
Cluster size, $K = 4$ (given)

(a) *To compute worst case C/I value at the mobile receiver*

Step 1. *To determine frequency reuse ratio, q*

We know that

$$q = \sqrt{3K}$$

For $K = 4,\ q = \sqrt{12} = 3.46$

Step 2. *To determine the C/I in the worst-case analysis*

In a 3-sector directional antenna design of a cellular system based on regular hexagonal pattern, the worst-case *C/I*, under the influence of interfering signals from the two nearest cochannel interfering cells, is given by the relationship

$$\frac{C}{I}(3-\text{sector}) \approx \frac{1}{(q)^{-\gamma} + (q+0.7)^{-\gamma}}$$

Then by substituting $q = 3.46$ and $\gamma = 4$, we get

$$\frac{C}{I}(3-\text{sector}) \approx \frac{1}{(3.46)^{-4} + (3.46+0.7)^{-4}}$$

Or, *C/I* (3-sector) = 97

Step 3. *To convert C/I (ratio) in C/I (dB)*

We know that *C/I* (dB) = 10 log *C/I* (ratio)
Therefore, *C/I* (dB) = 10 log 97
Hence C/I (dB) = 20 dB

(b) *Adequacy of K = 4 in a practical system*

Thus, for $K = 4$, theoretical value of *C/I* in a 3-sector directional antenna design of a cellular system comes out to be 20 dB, which exceeds the minimum required value of *C/I* = 18 dB. However, in a practical scenario under a heavy traffic area conditions as well as because of irregular terrain contour and imperfect site locations, there may be a further reduction of 6 dB in the theoretical value of *C/I*, which means *C/I* reduces to (20 – 6) = 14 dB that may not be adequate in a practical 3-sector cellular system.

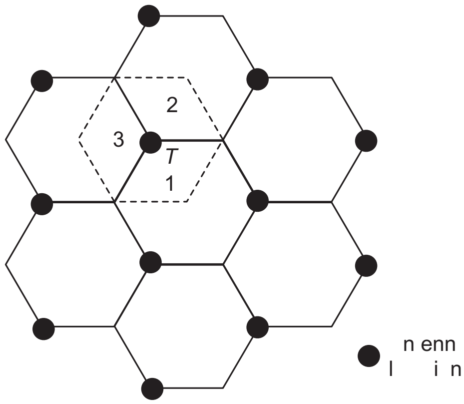

Figure 5.12 *Alternate placement of directional antennas*

There is an alternate way of configuring sectored cellular pattern by installing directional antennas at the selected corners of the regular hexagonal cellular pattern instead of installing them at the centre of the cell. Figure 5.12 depicts this arrangement of directional antennas.

For example, a directional antenna is installed at a point where three adjacent cells $C1$, $C2$ and $C3$ meet, that is, junction T, as shown in the figure. Similarly, other directional antennas are installed at the corners as shown by dark dots.

5.5.2 Six-sector Cellular System Design

A cell can be divided into six sectors by using six 60 -beam directional antennas. Figure 5.13 depicts realistic coverage of a practical 60 -beam directional antenna in one sector.

Figure 5.14 depicts realistic antenna coverage of practical 60 -beam directional antennas in all the six sectors of a cell, and overlap of antenna radiation patterns in 6-sector cellular configuration.

Figure 5.15 depicts six-sector per cell configuration in a hexagonal cellular reuse pattern.

The mobile unit situated at the boundary of its operating cell (with radius R) in worst case will now experience cochannel interference from corresponding cochannel sectors of only one out of six interfering cells. This situation is depicted by shaded sectors which use the same frequency channels.

From the hexagonal geometry, it can be easily seen that the distance from one co-channel interfering sector of $K = 7$ frequency reuse pattern would be $D1 = D + 0.7R$. Then from Eq. (5.3), the C/I ratio can be expressed as under:

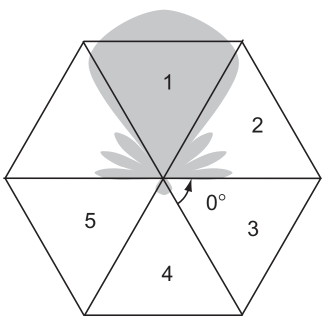

Fig. 5.13 *Realistic coverage of a 60°-directional antenna in one sector*

$$\frac{C}{I}(6-\text{sector}) \approx \frac{R^{-\gamma}}{(D+0.7R)^{-\gamma}} \tag{5.13}$$

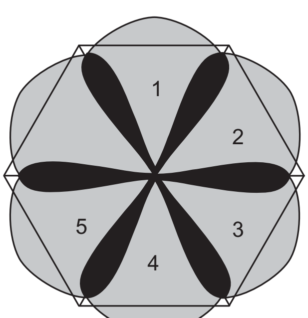

Fig. 5.14 *Overlap of antenna radiation patterns in 6 sectors*

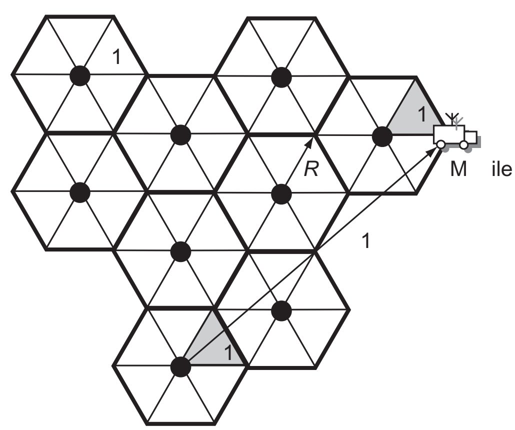

Fig. 5.15 *Worst-case cochannel interference in 6-sector configuration*

Using $q = D/R$ and simplifying, we get

$$\frac{C}{I}(6-\text{sector}) \approx \frac{1}{(q+0.7)^{-\gamma}} \tag{5.14}$$

EXAMPLE 5.7 | Worst-case *C/I* in 6-sector cellular system for $K = 7$

Calculate the signal-to-cochannel interference ratio C/I at the mobile receiver located at the boundary of its operating cell, under the influence of interfering signals from one cochannel interfering cell in the first tier in a cellular system designed with 6-sector directional antenna cellular system designed for K = 7. Comment on the results for a practical cellular system. Assume the path-loss exponent as 4.

Solution

Configuration of cellular antenna design = 6-sector (given)
Number of cochannel interferers = 1 (given)
The path-loss exponent, $\gamma = 4$ (given)
Cluster size, K = 7 (given)

te t e t e e tt e i e e ei e

Step 1. *To determine frequency reuse ratio, q*

We know that

$$q = \sqrt{3K}$$

For $K = 7,\ q = \sqrt{21} = 4.6$

Step 2. *To determine the C/I value in the worst-case analysis*

In a 6-sector directional antenna design of a cellular system based on regular hexagonal pattern, the worst-case *C/I*, under the influence of interfering signals from one nearest cochannel interfering cell, is given by the relationship

$$\frac{C}{I}(6-\text{sector}) \approx \frac{1}{(q+0.7)^{-\gamma}}$$

Then by substituting $q = 4.6$ and $\gamma = 4$, we get

$$\frac{C}{I}(6-\text{sector}) \approx \frac{1}{(4.6+0.7)^{-4}}$$

Or, *C/I* (6-sector) = 794

Step 3. *To convert C/I (ratio) in C/I (dB)*

We know that *C/I* (dB) = 10 log *C/I* (ratio)
Therefore, *C/I* (dB) = 10 log 794
Hence, C/I (dB) = 29 dB

Comment on the results Thus, for $K = 7$, the theoretical value of *C/I* in a 6-sector directional antenna design of a cellular system is 29 dB which exceeds the minimum required value of $C/I = 18$ dB. Even in a practical scenario under a heavy traffic area conditions as well as because of irregular terrain contour and imperfect site locations, where there may be a reduction of 6 dB in the theoretical value of *C/I*, which means *C/I* reduces to (29 – 6)= 23 dB that still exceeds the required value of *C/I* in a practical system. Hence, it may be concluded that six-sector cellular design with $K = 7$ meets the desired requirement of *C/I* as compared to that of an omnidirectional antenna design.

EXAMPLE 5.8 | Worst -case C/I in 6-sector cellular system for $K = 4$

Show that the worst-case C/I at the mobile receiver located at the boundary of its serving cell, under the influence of interfering signals from one cochannel interfering cell in the first tier in a cellular system designed with a 6-sector directional antenna cellular system designed for K = 4 is adequate for a practical cellular system in which there may be 6 dB performance margin needed. Assume the path-loss exponent as 4.

Solution

Configuration of cellular antenna design = 6-sector (given)
Number of cochannel interferers = 1 (given)
The path-loss exponent, $\gamma = 4$ (given)
Cluster size, $K = 4$ (given)

te t e t e e tt e i e e ei e

Step 1. *To determine frequency reuse ratio, q*

We know that

$$q = \sqrt{3K}$$

For $K = 4$, $q = \sqrt{12} = 3.46$

Step 2. *To determine the C/I in the worst-case analysis*

In a 6-sector directional antenna design of a cellular system based on regular hexagonal pattern, the worst-case C/I, under the influence of interfering signals from two nearest cochannel interfering cells, is given by the relationship

$$\frac{C}{I}(6-\text{sector}) \approx \frac{1}{(q+0.7)^{-\gamma}}$$

Then by substituting $q = 3.46$ and $\gamma = 4$, we get

$$\frac{C}{I}(6-\text{sector}) \approx \frac{1}{(3.46+0.7)^{-4}}$$

Or, C/I (6-sector) = 294

Step 3. *To convert C/I (ratio) in C/I (dB)*

We know that C/I (dB) = 10 log C/I (ratio)
Expressing it in dB, C/I (dB) = 10 log 294
Hence / (dB) = 24.76 dB

Step 4. *Adequacy of K = 4 in a practical system*

Thus, for $K = 4$, theoretical value of C/I in 6-sector directional antenna design of a cellular system is 24.76 dB which exceeds the minimum required value of C/I = 18 dB. Even in a practical scenario under a heavy traffic area conditions as well as because of irregular terrain contour and imperfect site locations, where there may be a reduction of 6 dB in theoretical value of C/I, which means C/I reduces to (24.76 – 6 = 18.76) dB, that still exceeds the required value of C/I in a practical system.

Hence it may be concluded that six-sector directional antenna design with K = 4 also meets the desired requirement of C/I even in a practical cellular system.

EXAMPLE 5.9 | Optimum value of cluster size

A TDMA digital cellular system is designed to accept a C/I value of 15 dB. Find the optimum value of K for

(a) omnidirectional antenna design

(b) six-sector 60° directional antenna design

Comment on the use of cell sectoring in this case.

Solution

$C/I = 15$ dB (given)

Step 1. *Converting given C/I in dB to C/I in ratio*

We know that C/I (dB) $= 10 \log_{10} (C/I)$

Or, $15 = 10 \log_{10} (C/I)$

Or, C/I (ratio) $= 10^{1.5} = 31.6$

(a) *To find K for omnidirectional antenna design*

Step 2. *To determine the value of q in omnidirectional antenna design*

Considering only the first tier of six cochannel interfering cells, and approximating them to be equidistant from the serving cell-site, then the frequency reuse ratio q can be determined from the relationship

$$q = \left(6 \times \frac{C}{I}\right)^{1/4}$$

$$q = (6 \times 31.6)^{1/4} = 3.7$$

Step 3. *To find the value of K in omnidirectional antenna design*

We know that $q = \sqrt{3K}$

Or, $K = (q) \;/\; 3$

Or, $K = (3.7) \;/\; 3$

Or, $K = 4.56$

The practical valid value of K which is higher than 4.56 is 7

Hence, $K = 7$

(b) *To find K for six-sector 60° directional antenna design*

Step 4. *To determine the value of q in six-sector 60° directional antenna design*

By using six-sector 60 directional antenna design in place of an omnidirectional antenna in each cell, the cochannel interference decreases drastically because there is only one cochannel interferer in the first tier. We know that

$$\frac{C}{I}(6 - \text{sector}) \approx \frac{1}{(q + 0.7)^{-4}}$$

By substituting $C/I = 31.6$, we get

$$31.6 \approx \frac{1}{(q + 0.7)^{-4}}$$

Step 5. *To find the value of K in omnidirectional antenna design*

We know that $q = \sqrt{3K}$

Therefore, $31.6 = \dfrac{1}{(\sqrt{3K} + 0.7)^{-4}}$

Solving for K, we get the practical valid value of K as 3

Hence, $K = 3$

Comment on the use of cell sectoring in this case It is observed here that by using six-sector 60 directional antenna design, the system capacity increases by a factor of $7/3 = 2.33$. Hence, if there is a requirement of capacity enhancement in the system, cell sectoring should be used.

Table 5.1 *Comparison of C/I for different antenna systems*

Type of antenna system design	*Worst-case C/I for K = 7*	*Worst case C/I for K = 4*
Omnidirectional	17 dB	11.33 dB
120 directional (3–sector)	24.5 dB	20 dB
60 directional (6–sector)	29 dB	24.76 dB

Comparison of Performance in $K = 7$ and $K = 4$ Systems

Table 5.1 summarises worst case C/I for all three types of antenna system design; Omnidirectional, 120 directional (3–sector), 60 directional (6–sector) for cluster sizes of 7 and 4.

A $K = 7$ cell-pattern system gives adequate cochannel reuse distance in an omnidirectional cellular system having normal traffic. The cochannel reuse distance is more or less adequate to give the desired C/I value.

In practice, a cell cannot be sectored ideally because ideal antenna radiation patterns cannot be implemented. Therefore, the C/I values obtained in the examples for ideal sectors are optimistic. However, it can be concluded from these examples that the use of sectoring increases the signal-to-interference ratio at the mobile units. Moreover, it is possible to reduce the frequency reuse factor from $K = 7$ to $K = 4$ by using three- and six-sector cells, respectively. This reduction in frequency reuse from seven to four would result in a capacity increase from 1.67 and 2.3 respectively, allowing an increase in the number of simultaneous communication links. The disadvantage of using sectors is that each sector is nothing but a new cell with a different shape, because channels have to be re-assigned between the different sectors of a cell. The network load is also substantially increased because of more frequent requirement of handoff to be made each time a mobile user moves from one sector of a cell to another.

When the traffic increases, a 3-sector cellular system (120 directional antennas) should be implemented using smaller size of each cell such as in urban area environment. In certain hot spots (weak signal areas) within the cell, a 6-sector cellular system (60 directional antennas) can be used locally to increase the channel utilisation.

If a given area is covered by both $K = 7$ and $K = 4$ cell patterns employing six-sector configuration, the $K = 7$ system has a total of 42 sectors whereas the $K = 4$ system has a total of only 24 sectors. So $K = 7$ system 6–sectors configuration has less cochannel interference than $K = 4$ system 6–sectors cellular systems.

One advantage of six-sector configuration with $K = 4$ is that they require fewer cell sites than a three-sector configuration with $K = 4$. The main disadvantages of six-sector configuration are that they require more antennas to be mounted on the tower as well as they often require more frequent hand-offs because there are more chances that the mobile units will travel across the six sectors of the cell.

Furthermore, assigning the proper traffic channel to the mobile unit in each sector is more difficult. This puts extra overheads on the system complexity and reliable operation. Also, in small cells, it becomes difficult to control interference and the coverage is never uniformly distributed because of irregular terrain and surroundings. Thus the use of a $K = 4$ pattern with six sectors in small cells needs to be considered only for special implementations such as indoor mobile operations in cellular systems or narrowbeam applications for increasing the traffic capacity.

EXAMPLE 5.10 **Comparison of $K = 7$ and $K = 4$ cellular configuration**

Compare the number of traffic channels per sector in the following two different cellular systems employing 312 traffic channels for use:
System A: $K = 7$ pattern, three-sector configuration
System B: $K = 4$ pattern, six-sector configuration

Solution

Number of traffic channels = 312 channels

System A: *To determine the number of traffic channels per sector*

The reuse pattern, $K = 7$ (given)

Number of sectors per cell = 3 (given)

Step 1. *To determine the number of traffic channels per cell*

Number of traffic channels per cell = number of channels / K

Number of traffic channels per cell = 312 / 7 = 45 (on an average)

Step 2. *To determine the number of traffic channels per sector*

Number of traffic channels per sector = channels per cell / number of sectors

Number of traffic channels per sector = 45 / 3 = 15

System B: *To determine the number of traffic channels per sector*

The reuse pattern, $K = 4$ (given)

Number of sectors per cell = 6 (given)

Step 3. *To determine the number of traffic channels per cell*

Number of traffic channels per cell = number of channels / K

Number of traffic channels per cell = 312 / 4 = 78 (on an average)

Step 4. *To determine the number of traffic channels per sector*

Number of traffic channels per sector = channels per cell / number of sectors

Number of traffic channels per sector = 78 / 6 = 13

Step 5. *Comparison of System A and System B*

It is seen that the $K = 7$ pattern offers higher spectrum efficiency than the $K = 4$ pattern in the above system configuration. In practice, three-sector and six-sector configurations are mixed in the same cellular system in order to customise channel distribution for uneven traffic distribution.

5.5.3 Microcell Zone Concept

The microcell zone concept is related to sharing the same radio equipment by different microcells. This results in reduction of cluster size and hence increase in system capacity. The microcell coverage zone approach is used in practice to expand the capacity of cellular communication systems.

Cell sectoring depends upon proper installation of cell-site directional antennas to improve system capacity as well as enhance signal quality. But this also results into increase in the number of handoff occurrence and trunking inefficiencies. In a 3-sector or 6-sector cellular configuration, each sector behaves like a new cell with a different cell size and shape. Channels allocated to the original cell are partitioned between the different sectors of a cell, thereby reducing available number of channels in each sector. Moreover, handoff has to be made each time a mobile subscriber moves from one sector to another sector of the same cell. This results in substantial increase of network load on switching and control link elements such as BSC and MTSO/MSC of the cellular system. When all the three or six sectored cell-site directional antennas are located at the centre of the cell, the problem of partitioning of the channels and increase in network load becomes more serious.

This problem can be reduced to a large extent by alternate arrangement of configuring a sectored cellular pattern by installing directional antennas at the selected corners of the regular hexagonal cellular pattern instead of installing them at the centre of the cell. Figure 5.16 depicts this arrangement in a 3-sector cellular configuration.

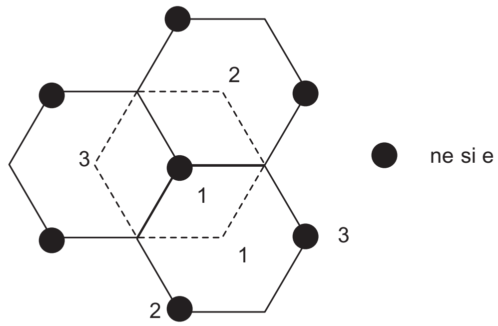

Fig. 5.16 ▌ *Location of zone-sites in sectored cells*

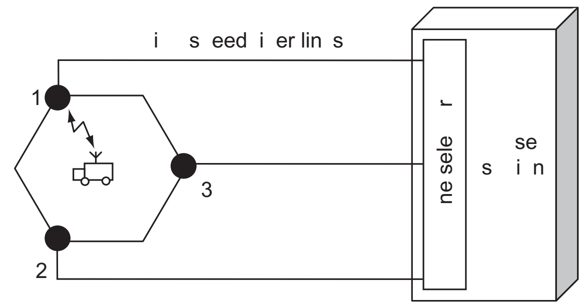

Fig. 5.17 ▌ *Lee's microcell zone concept*

Three directional antennas are installed at a junction point, 1, referred to as zone-site', where three adjacent cells $C1$, $C2$ and $C3$ meet one another. As shown in figure, 1, 2 and 3 represent three zone-sites of the cell $C1$. Each corner-excited zone-site employs three 135 directional antennas. In fact, all three zone-sites act as receivers for signals transmitted by a mobile subscriber located anywhere in the cell. These three zone-sites are connected to one common base station, as shown in Fig. 5.17. This arrangement is called *Lee's microcell zone technique* and the concept is known as *Lee's Microcell zone concept.*

The zone-sites are connected to the base station by a high-speed fiber link to avoid delay and traffic congestion. The base station determines which of the three zone-sites has the best reception from the mobile subscriber and uses that selected zone-site to transmit the signal on the downlink to the mobile subscriber. In this way, only one zone-site is active at a time for a communication link. This results in reduction of cochannel interference experienced by the mobile subscriber from a cochannel zone-site as compared to what would have been with an omnidirectional antenna configuration.

Thus, a microcell zone architecture reduces cochannel interference, increases system capacity, improves signal quality, and demands fewer handoffs. The system is easy to implement. The system capacity, based on the $K = 3$ system, is 2.33 times greater than the existing analog cellular system of $K = 7$ for the same C/I requirement of 18 dB. This microcell system can provide better voice quality than the Advanced Mobile Phone Service (AMPS) cellular system at 850 MHz. It can be used with digital cellular communication systems as well as personal communication systems, and is suitable for indoor applications. Microcell zone technique is also useful to provide cellular services along highways or in dense urban areas.

5.6 ANTENNA PARAMETERS AND THEIR EFFECTS

Many a times situations demand that coverage must be reduced to compensate for cochannel interference. There are several ways of exploiting the effects of various cell-site antenna parameters. For example, antenna radiation pattern, antenna beamwidth, antenna gain, antenna height, separation between transmitting and receiving antennas, and antenna tilting affect the cellular system design. There are some other aspects of antenna parameters such as re-orienting the direction antenna patterns, changing the antenna beamwidth, decreasing the antenna height at the cell-site which may also affect the radio coverage but reduce the system interference.

The antenna pattern can be omnidirectional, directional, or any other shape in both the horizontal and vertical planes. The antenna patterns seen in cellular systems are different from the patterns seen in the free space. For example, if a mobile unit travels around a cell-site in areas with many buildings, the omnidirectional antenna will not duplicate the omnidirectional radiation pattern. The radiation pattern is distorted in an urban or suburban environment. So the design of the antenna pattern should be based on the terrain contour, the building and forest density, and other conditions within a given area. The signal reflection mechanism in mobile radio environment is illustrated in Fig. 5.18.

For a 120 degree directional antenna, the front-to-back ratio is about 10 dB less than the front lobe. This condition exists because the strong signal radiates in front, bouncing back from the surroundings so that some part of the signal energy can be received from the back of the antenna. In such a case, the beamwidth of the directional antenna used at the cell site has no correlation between its measured values in the free space and the mobile radio environment. Control of secondary lobe formation in an antenna radiation pattern is very critical in the implementation of a directional antenna. A skewed configuration of the number of directional antennas at the same cell-site results into smoother pattern as compared to a pattern accompanied with ripples and deep null formation in case of all antennas facing outward.

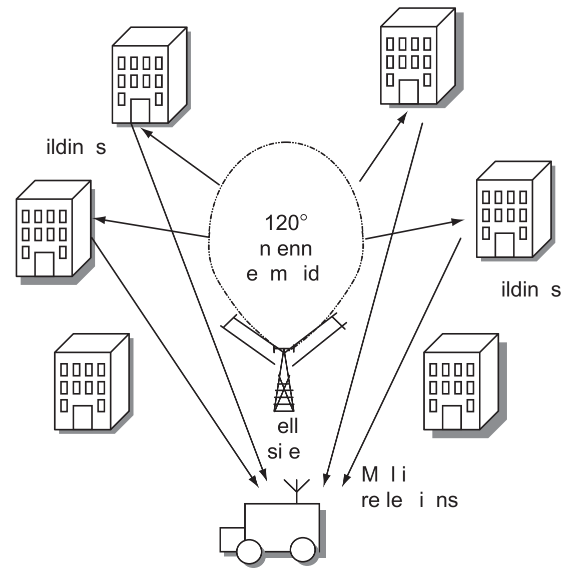

Fig. 5.18 *Signal reflection mechanism in a mobile radio environment*

The antenna gain enhances the radiated power from the antenna. The height of the cell-site antenna can affect the area and shape of the coverage in the system. Effective antenna height should be taken into consideration for received signal analysis within a cell. The effective antenna height and the actual antenna height are same only when the mobile unit is traveling on flat ground. It is easy to decrease antenna height to control coverage in a flat-terrain area. For decreasing antenna height in a hilly area, the signal strength contour is different from the situation of transmitting power decrease. Therefore, a decrease in antenna height becomes very difficult to control the overall coverage plan. Some areas within the cell may have a high attenuation while others may not.

Separation between two transmitting antennas at the same cell-site should be minimised to avoid the intermodulation. The minimum separation between a transmitting antenna and a receiving antenna must be ensured in order to avoid receiver desensitisation. For example, if the separation between a transmitting antenna and a receiving antenna is 15 metres horizontally, the signal isolation obtained from the free-space formula is 56 dB. For better performance, the transmitting antenna and receiver antenna are not mounted in the same horizontal plane, but rather these are mounted on the same vertical pole, if they are omnidirectional antennas. In case of directional antennas, this restriction can be moderated because of the directive patterns.

Facts to Know!

Antenna tilting can reduce the interference to the neighbouring cells and enhance the weak spots in the cell coverage. Different antenna patterns and antenna gains at the cell site as well as at the mobile units would affect the system performance and must be considered in the system design.

Key Terms

- Antenna
- Active antenna
- Antenna pattern
- Antenna polarisation
- Attenuation
- Bandwidth
- Cell sectorisation
- dB
- dBd
- dBi
- dBm
- Dipole
- Directional antenna
- Directional gain
- Half-wave antenna
- Monopole antenna
- Omnidirectional antenna
- Patch antenna
- Quarter-wave antenna
- Radiation pattern

Summary

This chapter provides an overview of the cellular antennas — cell-site antennas and mobile antennas, including antenna types and their characteristics. The use of omnidirectional antennas at the mobile is a must as it is expected to receive signals while on the move in any direction with respect to its serving cell-site. However, in order to reduce the effect of cochannel interference in frequency reuse cellular systems employing omnidirectional antenna at the cell-site, cells are divided into sectors and each sector of a cell is then served by its dedicated directional antenna. It is observed that the signal quality measured in terms of *C/I* ratio of the overall system and hence the capacity is significantly enhanced by sectoring of the cells. In the next chapter, the channel assignment techniques including channel sharing and channel borrowing are covered in sufficient detail so as to consolidate the advantages gained by using sectored cells.

Important Equations

- ❑ $G_r = (4\pi A_{eff}) / \lambda_c^2$ (5.1)
- ❑ $\dfrac{C}{I} \approx \dfrac{1}{6}.(q)^4$ (5.6)
- ❑ $q = \left(6 \times \dfrac{C}{I}\right)^{1/4}$ (5.7)
- ❑ $\dfrac{C}{I}(\text{omni}) \approx \dfrac{1}{2(q-1)^{-\gamma} + 2(q)^{-\gamma} + 2(q+1)^{-\gamma}}$ (5.10)
- ❑ $\dfrac{C}{I}(3-\text{sector}) \approx \dfrac{1}{(q)^{-\gamma} + (q+0.7)^{-\gamma}}$ (5.12)
- ❑ $\dfrac{C}{I}(6-\text{sector}) \approx \dfrac{1}{(q+0.7)^{-\gamma}}$ (5.14)

Short-Answer Type Questions with Answers

A5.1 What is the basic function of an antenna
The basic function of a transmitting antenna is to convert the electrical energy (available in the form of electric field between the conductors and the magnetic field surrounding them) traveling along an RF cable from a transmitter unit into electromagnetic waves in space. At the receiving antenna, the electric and magnetic fields in space cause a current to flow in the conductors that make up the antenna and some of this energy is transferred to the RF cable connected to it and the receiver unit. Moreover, antennas are reciprocal devices; that is, the same antenna design works equally well as a transmitting or a receiving antenna with the same amount of gain.

A5.2 What is meant by passive antennas
In general, antennas are passive devices, which mean the power radiated by a transmitting antenna cannot be greater than the power entering at its input from the transmitter. In fact, the radiating power from the antenna is always less than the power at its input because of losses. In fact, the antenna gain in a passive antenna results from concentration of power in one direction and is accompanied by a loss in other directions.

A5.3 Define the terms antenna radiation pattern antenna gain antenna polarisation and the front-to-back ratio.

An antenna radiation pattern is defined as a graphical representation of the radiation properties of the antenna as a function of space coordinates. The radiation pattern characterises the performance of the antenna.

The antenna gain of a transmitting antenna is defined as the ratio of the radiation intensity flowing in a given direction to the radiation intensity averaged over all directions. In fact, antenna gain does not refer to obtaining more output power than input power but refers to directionality.

The polarisation of an antenna means the orientation of the electric field vector of electromagnetic waves it radiates or receives. It could be horizontal or vertical or hybrid.

The front-to-back ratio of an antenna is the ratio of radiated power in intended direction to radiated power in opposite direction.

A5.4 Differentiate between the absolute gain and the relative gain of a transmitting antenna in a given direction.

The absolute gain or the power gain of a transmitting antenna in a given direction is the ratio of the radiation intensity flowing in that direction to the radiation intensity that would be obtained if the power were radiated by an isotropic antenna. The relative gain of a transmitting antenna in a given direction is the ratio of the absolute gain of the antenna in the given direction to the absolute gain of a reference antenna (a perfect omnidirectional isotropic antenna) in the same direction. The power input to the two antennas must be the same in both cases.

A5.5 Why are high-gain omnidirectional or directional antennas used at the cell-site

The antennas used at the cell-site for cellular radio communication systems have to fulfill the requirement of not only providing adequate radio coverage but also reducing cochannel interference arising due to frequency reuse in cellular architecture. So there is a need for high-gain (usually 6-dB or 9-dB) omnidirectional antennas, and directional antennas with beamwidths of 60 degrees or 120 degrees for sectorised cells. Typical cell-site antennas use variations of the collinear antennas backed by reflectors for omnidirectional patterns or log-periodic antennas for directional patterns.

A5.6 What is the significance of quarter-radius rule in the selection of the cell-site

In the selection of the cell-site antenna, reduction of interference is the major concern while meeting the radio coverage requirements. When a cell-site is chosen on the map, there is a 50% chance that the site location can be acquired due to logistic reasons. So an antenna location can be found within a quarter of the size of the cell $R/4$, where R is the radius of the cell. This is termed as the *quarter-radius rule*. However, the quarter-radius rule can be applied only on relatively flat terrain, not in a hilly area.

A5.7 State the reasons to prefer rooftop antennas over glass-mounted antennas at the vehicle-installed mobile unit.

In glass-mounted mobile antennas, the signal energy is coupled through the glass with slight loss of energy through coupling. Glass-mounted antennas cannot be installed on the shaded glass found in some motor vehicles because this type of glass has a high metal content. In addition, the height of the glass-mounted antenna is always lower than that of the roof-mounted antenna that may result into 3-dB difference between these two types of mobile antennas.

A5.8 Differentiate between a high-gain omnidirectional antenna and a directional antenna having the same gain.

A high-gain omnidirectional antenna is different from a directional antenna having the same gain because in a high-gain omnidirectional antenna, the antenna beamwidth pattern is suppressed horizontally whereas in the directional antenna, the antenna beamwidth pattern is suppressed horizontally.

A5.9 Why is the in-line arrangement of the two horizontally oriented space-diversity mobile antennas used

Theoretical analysis and measured data show that in horizontally oriented space-diversity mobile antennas, the in-line arrangement of the two antennas exhibit less fading. Therefore the in-line arrangement of the two horizontally oriented space-diversity mobile antennas is recommended as compared to perpendicular arrangement to the direction of motion of the vehicle.

A5.10 Why is it not recommended to choose higher value of cluster size in order to increase value

Increasing the value of cluster size can increase *C/I* but the number of frequency channels available in a cell decreases for a fixed allocated frequency spectrum. This results in overall decrease of system capacity as well as spectrum efficiency that is contrary to advantage of frequency-reuse scheme in a cellular system. So it is recommended to achieve a higher value of *C/I* without increasing the cluster size.

A5.11 Which technique could be deployed to achieve higher value of at the mobile unit in a fully equipped regular hexagonal cellular pattern

A suitable arrangement of directional antennas in place of omnidirectional antenna at each cell-site can be deployed to reduce cochannel interference, and thereby achieving a higher value of *C/I*. This simply means that each omni cell is divided into three or six sectors radially while retaining the same number of channels per cell in the 7-cell cluster. Accordingly, three or six directional antennas are used at a cell-site.

A5.12 Which antenna parameters can affect the cellular system design so as to improve the signal quality by reducing cochannel interference

In many situations, it is desirable to reduce the radio coverage in order to minimise the effect of cochannel interference. There are several aspects related to cell-site antenna parameters such as reorienting the direction antenna patterns, changing the antenna beamwidth, antenna tilting, antenna gain, decreasing the antenna height at the cell-site, separation between transmitting and receiving antennas that affect the radio coverage but reduce the cochannel interference.

Self-Test Quiz

S5.1 A(n) antenna is defined as a hypothetical loss-less antenna having equal radiation in all directions.

(a) monopole (b) quarter-wave dipole
(c) half-wave dipole (d) isotropic

S5.2 A(n) antenna is one which has the property of radiating or receiving electromagnetic waves more effectively in some directions than in others.

(a) omnidirectional (b) directional
(c) isotropic (d) smart

S5.3 The effective radiated power (ERP) is greater than the effective isotropic radiated power (EIRP) by approximately.

(a) 1 dB (b) 2 dB
(c) 3 dB (d) 6 dB

S5.4 The relationship between antenna gain, and effective area or aperture, A_{eff} of a receiving antenna in a given direction is

(a) $G_r = (4\pi\, A_{eff}) / \lambda_c^2$
(b) $G_r = 4\pi\, A_{eff} \lambda_c^2$
(c) $G_r = (4\pi\, \lambda_c^2) / A_{eff}$
(d) $G_r = (4\pi\,) / (\lambda_c^2\, A_{eff})$

S5.5 The radiation resistance of a half-wave dipole antenna situated in free-space and fed at the centre is approximately

(a) 50 Ω (b) 70 Ω
(c) 300 Ω (d) 1 MΩ

S5.6 Wireless communication systems usually use polarisation because this is more convenient for use with portable and mobile antennas.

(a) horizontal (b) vertical
(c) hybrid (d) none of the above

S5.7 Cellular base-station receiving antennas are usually mounted in such a way so as to obtain diversity.

(a) frequency (b) polarisation
(c) space (d) horizontal

S5.8 The minimum separation between a transmitting antenna and a receiving antenna is necessary to avoid the problem.

(a) cochannel interference
(b) receiver desensitisation
(c) intermodulation
(d) (b) and (c) both

S5.9 If the radius of a cell is 12 km, the cell-site antenna can be located within a radius.

(a) 0.5 km (b) 1 km
(c) 2 km (d) 3 km

S5.10 In a mobile radio environment, the frequency reuse ratio, q and the carrier-to-interference ratio C/I are related by the following expression.

(a) $q = \left(6 \times \frac{C}{I}\right)^{1/4}$ (b) $q = \left(6 \times \frac{C}{I}\right)^{4}$

(c) $q = \left(\frac{1}{6} \times \frac{C}{I}\right)^{1/4}$ (d) $q = \left(\frac{1}{6} \times \frac{C}{I}\right)^{4}$

S5.11 For a cluster size of 7, the frequency reuse ratio is approximately equal to

(a) 3 (b) 7
(c) $\sqrt{21}$ (d) 21

S5.12 For a cluster size of 7, the carrier-to-interference ratio C/I is approximately

(a) 73.5 (b) 147
(c) 1.5 (d) 7

S5.13 In an omnidirectional cellular system, the value of cluster size of would be a correct choice to meet C/I requirement of at least 18 dB in the worst-case scenario.

(a) 3 (b) 4
(c) 7 (d) 9

S5.14 Separation between two transmitting antennas at the same cell-site should be minimised to avoid

(a) cochannel interference
(b) adjacent channel interference
(c) intermodulation
(d) receiver desensitisation

S5.15 The minimum separation between a transmitting antenna and a receiving antenna at the cell-site must be ensured in order to avoid

(a) cochannel interference
(b) adjacent channel interference
(c) intermodulation
(d) receiver desensitization

S5.16 can reduce the interference to the neighbouring cells and enhance the weak spots in the cell coverage.

(a) Antenna tilting
(b) 60-degree sectorised cells
(c) 120-degree sectorised cells
(d) Reorienting the directional antenna patterns

Answers to Self-Test Quiz

S5.1 (d); S5.2 (b); S5.3 (b); S5.4 (a); S5.5 (b); S5.6 (b); S5.7 (c); S5.8 (d); S5.9 (d); S5.10 (a); S5.11 (c); S5.12 (a); S5.13 (d); S5.14 (c); S5.15 (d); S5.16 (a)

Review Questions

Q5.1 What type of information is available from the radiation pattern of an antenna?

Q5.2 What factors determine antenna gain?

Q5.3 Distinguish between the units dB, dBi, dBd, dBm, and dBW.

Q5.4 Describe the various criteria that decide placement of cell-site antennas and mobile antennas.

Q5.5 Assuming that the source of interference has been identified, suggest at least two potential solutions to resolve signal coverage problems.

Q5.6 What is a multipath condition? What effect does it have on an antenna system at the mobile?

Q5.7 Why must be an omnidirectional antenna used at the mobile unit in a mobile radio environment?

Q5.8 What are the advantages and disadvantages of cell sectoring?

Q5.9 Derive a relationship between desired C/I and cochannel interference reduction factor (q) in a 7-cell frequency reuse regular hexagonal pattern. Assume all cells are of equal size and use omnidirectional antennas at the cell-site.

Q5.10 List two types of antenna downtilting schemes.

Analytical Problems

P5.1 Determine the *C/I* value for frequency reuse factors of 4, 7, and 12, assuming an omni directional antenna with six cochannel interfering cells and a propagation path-loss slope of 40 dB/decade. Ignore the interference from second tier cochannel interfering cells.

P5.2 Consider a cellular system employing *K*-cell reuse pattern. The cell-site transmitters are located at the centre of each cell. The desired user is at the edge of the cell. Assume that the six nearest cochannel interferers (cell-site transmitters) in a cellular system cause most of the interference to the user. Determine the best value of reuse pattern *K* that achieves a minimum acceptable *C/I* value of 18 dB. Assume the path-loss exponent as 3.1.

P5.3 Calculate q for a cell radius of 2 km and the distance between two cochannel cells of 9.2 km.

P5.4 A new cellular service provider decides to employ a cluster of 19 cells for frequency reuse. Calculate the worst case *C/I* (in dB) at the mobile, when it is receiving cochannel interference from first tier cochannel cells only. Assume path loss exponent as 4 in a mobile radio environment.

P5.5 A cellular system is designed with 100 cell-sites deployed with a frequency reuse factor of 4, and an overall of 500 full duplex channels. Determine the number of channels per cell, total number of channels available to the service provider, and the minimum carrier-to-interference ratio (*C/I*) of the system in dB in a mobile radio environment.

P5.6 Draw the hexagonal grid corresponding to six-sector cells. Compute and tabulate the value of *C/I* for reuse factors of 4, 7, and 12. Compare and comment on the results.

P5.7 Prove that *C/I* performance of 3-sector cellular design improves over omnidirectional design by more than 6 dB. Assume propagation path-loss exponent as 4 and cluster size 7 in a regular hexagonal cellular pattern.

P5.8 Compare cochannel interference from the first tier of six interferers with that from the second tier of twelve interferers for a seven-cell reuse cellular pattern. Assume propagation path-loss exponent as 4. Refer Fig. 5.19 and Fig. 5.20 for reference.

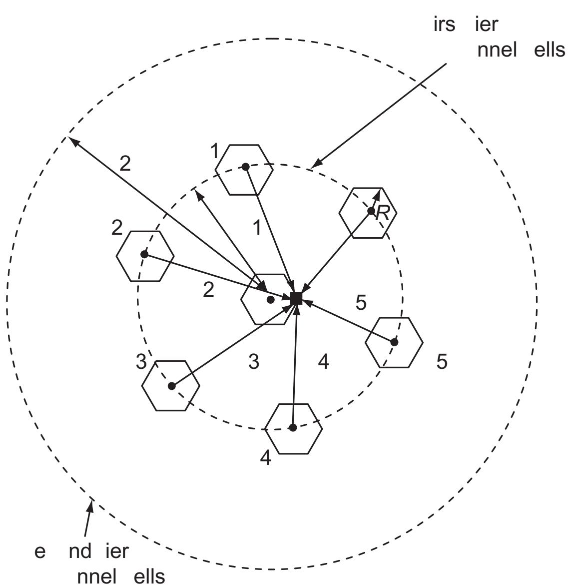

Fig. 5.19 | *For P5.8*

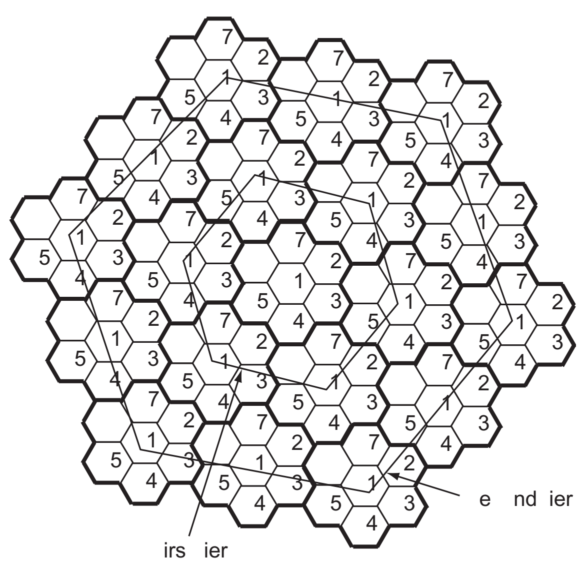

Fig. 5.20 | *For P5.8*

P5.9 Let the total service area to be covered with a number of uniform-sized cells of 5 km^2 area each of 2000 km^2. How many times would the cluster of size 4 have to be replicated in order to cover the entire cellular area?

P5.10 Show that $K = 7$, six-sectors configuration has better C/I performance than $K = 4$, six-sector cellular systems. Assume regular hexagonal cellular architecture.

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6

In cellular communications, one of the main research issues is how to achieve optimum system capacity with a limited frequency spectrum. Frequency management and channel assignment in a cellular system is important in order to achieve the basic objectives of spectrum utilisation as well as adaptability to traffic density. For many years, researchers have proposed and studied many channel-assignment strategies to increase the capacity of cellular systems. In this chapter, all the aspects leading to an efficient and effective frequency planning of cellular systems are discussed in detail. The allocated frequency spectrum is divided into a number of frequency channels depending upon the system parameters. The assignment of these channel sets to cells follows different strategies. Fixed channel-assignment techniques, dynamic channel allocation as well as hybrid strategies are covered here.

Frequency Management and Channel Assignment

6.1 FREQUENCY MANAGEMENT

Frequency management involves the assignment of proper frequency channels in different cells in such a way so as to increase spectrum efficiency. The function of frequency management is to divide the allocated RF spectrum into the total number of available channels depending upon the channel bandwidth of a cellular system. These channels are then divided into subsets, which can be assigned to each cell either in a fixed manner or dynamically (that is, in response to any channel among the total available channels).

Frequency management refers to designating the set-up (control and signaling) channels, designating the voice (traffic) channels, numbering the channels, grouping the voice channels into subsets, which is generally carried out by each system according to its preference.

6.1.1 Channel Grouping

Each communication channel has a pair of frequencies that allows for full duplex operation. That is, an uplink (or reverse channel) in which a frequency is transmitted by the mobile subscriber and received by the cell-site, and a downlink (or forward channel) in which frequency is transmitted by the cell-site and received by the mobile subscriber. These two links are always separated in frequency by a specific value termed as *duplex spacing*. For example, in the 800 MHz band cellular system, the duplex separation is specified as 45 MHz. Allocated frequency spectrum in the 800 MHz system is divided into two bands: 'A' band originally reserved for companies in the cellular business, and 'B' band reserved for landline companies to allow access to cellular business.

Facts to Know!

Frequency management, also referred to as radio resource management, is the system-level control of cochannel interference and other radio transmission characteristics in wireless communication systems.

Frequency Reuse Grouping The process of dividing available channels into the frequency channel groups is necessary to meet the requirements of the cellular network, which include the following among others:

- desired channel separation within each cell
- making cluster designing easier
- avoiding the adjacent channels in the same cell
- use of combiner-tuner network device to allow a single antenna to transmit signals from multiple cells/sectors.

EXAMPLE 6.1 | Number of channels per cell

A full-duplex wireless cellular system is allocated a total spectrum of 20 MHz and each simplex channel has 25-kHz RF bandwidth. Determine
(a) Total number of full-duplex channels available
(b) Number of channels per cell-site if $K = 4$ cell reuse pattern is employed.

Solution

Total allocated RF spectrum bandwidth = 20 MHz (given)
Channel bandwidth per simplex channel = 25 KHz (given)

(a) *To determine number of full-duplex channel*

Step 1. *To determine duplex channel bandwidth*

Channel bandwidth per simplex channel = 25 kHz
Number of channels in a duplex link = 2
Therefore, duplex channel bandwidth = $25 \times 2 = 50$ kHz

Step 2. *To determine number of full-duplex channels*

Number of full-duplex channels = total bandwidth/duplex channel bandwidth
Number of full-duplex channels = 20 MHz / 50 kHz
Hence, total number of duplex channels = 400 channels

(b) *To determine number of channels per cell-site*

Step 3. *Number of cells in one cluster, $K = 4$ (given)*

Number of channels per cell-site = total number of channels/K
= 400/4
Hence, number of channels per cell-site = 100 channels

In order to understand channel grouping and frequency reuse grouping, it is important to understand how the channels are allocated in the 800-MHz spectrum in the first generation analog cellular systems. AMPS (Advanced Mobile Phone System) is the set of rules that was developed to cover North American Analog Cellular standard. Over the years, many RF engineers worked to find optimal channel groupings.

The results are mixed but most RF engineers agree that for most omnidirectional or sectorised applications, the frequency reuse pattern $K = 7$ plan is best. However, the $K = 4$ and $K = 9$ plans are also becoming more common as the cellular industry makes more use of the cell sectorisation.

EXAMPLE 6.2 Frequency management in AMPS—Nonextended spectrum

List the standard radio specification parameters in a non-extended spectrum allocated in US-AMPS analog cellular system. Describe the grouping of channels into subsets and numbering the channels.

Solution

The standard non-extended spectrum in a US-AMPS analog cellular system has a 20 MHz bandwidth on either side of the duplex spectrum. The original voice channels available in the first AMPS system are commonly referred to as non-extended spectrum. The major radio parameters are listed as under:

Uplink frequency band (Mobile Tx)	825–845 MHz
Downlink frequency band (Cell site Tx)	870–890 MHz
RF Spectrum bandwidth (on either end)	20 MHz
Duplex separation (Tx-Rx)	45 MHz
Channel spacing	30 kHz
Channel 1 frequency (Mobile Tx)	825.030 MHz
Channel 1 frequency (Cell site Tx)	870.030 MHz
Effective RF Spectrum bandwidth	19.980 MHz
Number of channels available	19.980 MHz / 30 kHz = 666 channels

The 666 channels are divided into two groups: Block A and Block B. Each block has 333 channels—312 voice and 21 set-up or control channels. The 21 set-up channels are derived from a seven-cell frequency-reuse pattern with three 120 sectors per cell, or a total of 21 sectors in a cluster of $K = 7$, which requires at least 21 set-up channels (one set up channel per sector).

The Block A channels are numbered as below:

Voice channels Channel numbers 1–312 (total of 312 voice channels)

Set-up channels Channel numbers 313–333 (total of 21 set-up channels)

The Block B channels are numbered as below:

Set-up channels Channel numbers 334–354 (total of 21 set-up channels)

Voice channels Channel numbers 355–666 (total of 312 voice channels)

The 42 set-up channels (21 channels each in Block A and Block B) are numbered in the middle of all the assigned channels, that is, channel numbers 313–333 in Block A and 334–354 in Block B. This is deliberately done in order to facilitate scanning of these channels continuously (channels numbers 313–354) by a frequency synthesiser in the mobile receivers, which can be the same equipment operating either in Block A or Block B.

Table 6.1 summarises AMPS frequency management in a non-extended spectrum.

Table 6.1 *Frequency management in AMPS (non-extended spectrum)*

unction	*No. of channels per group*	*Block A Channel numbers*	*Block B Channel numbers*
Total number of channels per block	333	001–333	334–666
Voice channels	312	001–312	355–666
Control channels	21	313–333	334–354

Since there are 21 set-up channels for each block, it is logical to group 312 voice channels into 21 subsets of 15 voice channels each (the last subgroup has 12 voice channels only). Each subset then consists of 16 channels—15 voice channels and 1 control channel. It is obvious that in each set, the closest adjacent channel is 21 channels away. The channel separation provided in such a way is sufficient to meet the adjacent channel isolation requirement of a 16-channel transmitter combiner.

When the voice channels are not engaged by the user in a conversation, the mobile uses set-up channels to communicate with the cellular system and let it know its availability and location. The mobile does not communicate using the set-up channel during an active call. Voice channels could be used as set-up channels if needed.

When a set-up channel is used as an analog set-up channel in analog cellular systems, that particular channel cannot be used anywhere within the system as a voice channel. However, if a set-up channel is used as a digital set-up channel in digital cellular systems, it could be used as a voice channel under controlled circumstances. Generally, none of the basic 21 set-up channels in either block can ever be used as a voice channel. However, only a few of the 21 designated set-up channels are being used in any system. Theoretically, when cell size decreases, the use of set-up channels should increase.

The basic 21 set-up channels are assigned in a $K = 4$, $K = 7$, or $K = 12$ frequency reuse pattern in cellular systems. If the set-up channel antennas are omnidirectional at the cell-site, then each cell only needs one set-up channel. Thus the remaining set-up channels will be left unused. For an AMPS system, a single cell is assigned a single set-up channel, so that a seven-cell reuse scheme would have seven set-up channels assigned to seven neighbouring cells that form a cluster ($K = 7$). Two neighbouring clusters would be assigned the remaining 14 set-up channels. Thus, set-up channels follow a 21-cell reuse scheme and are assigned over three clusters before being reused, even though the voice channels may be assigned using a seven-cell reuse scheme.

> **Facts to Know!**
>
> The set-up channels of Block A and Block B are adjacent to each other. In order to avoid interference between two systems employing Block A and Block B respectively, the set-up channels in the neighbourhood of channel number 333 (block A) and channel number 334 (block B) are preferably unused.

EXAMPLE 6.3 Frequency management in AMPS—extended spectrum

List the standard radio specification parameters in an extended spectrum allocated in the US-AMPS cellular system. Describe the grouping of channels into subsets and numbering the channels.

Solution

In the new additional spectrum allocation of 10 MHz (5 MHz on either end), an additional 166 channels are available.

The major radio parameters are listed as under:

Uplink frequency band (mobile Tx)	824–849 MHz
Downlink frequency band (cell-site Tx)	869–894 MHz
RF Spectrum bandwidth	25 MHz on either end
Duplex separation (Tx-Rx)	45 MHz
Channel spacing	30 kHz
Total number of channels available	832

The total number of 832 channels are divided into two groups: Block A and Block B. Each block has 416 channels—395 voice channels and 21 setup channels.

The frequency management in AMPS for an extended spectrum is summarised in Table 6.2. It shows how the channel numbers are distributed across the complete allocated extended spectrum and the additional channels are referred to as extended spectrum.

Table 6.2 *Frequency management in AMPS (extended spectrum)*

unction	*No. of channels per block*	*Block A Channel numbers*	*Block B Channel numbers*
Total number of channels per carrier	416	001–333; 667–716; 991–1023	334–666; 717–799
Voice channels	395	001–312; 667–716; 991–1023	355–666; 717–799
Control channels	21	313–333	334–354

There are no channels assigned between channel numbers 799 and 991.

Thus, 166 additional voice channels are allocated to extended spectrum AMPS cellular system—83 voice channels for each block *A* and block *B*. It may be noted that block *A* has extended spectrum both above the block *B* band as well as below the original block *A* band. The reason for this is that most of the new frequencies came from the original allocated frequencies. Therefore, in order for the block *A* band to have the same benefit as the block *B* band, it has to accept some frequencies at the higher end of the spectrum.

6.1.2 Set-up and Voice Channels

Set-up channels, also called *control channels*, are the channels designated to set up the calls. All set-up channels carry signaling data information only. Each cell or sector of a cell requires at least one set-up channel. According to usage, set-up channels can be classified into two types:

– Access channels
– Paging channels

Access channels are used for the mobile-originating (outgoing) calls. *Paging channels* are used for network-originating calls or mobile-terminating (incoming) calls with respect to mobile subscriber units. The channel bandwidth is 30 kHz in an AMPS cellular system. All 21 set-up channels can be used as actually paging channels. The access channel can be assigned by the MTSO as a channel other than the 21 set-up channels in a cell. The mobile unit receives the access channel information from the forward paging channels. In case of increased network-originating calls, another channel in a group of voice channels can be used as an access channel. In a low-traffic system, access channels and paging channels are the same. For this reason, a set-up channel is sometimes called an access channel and sometimes called a paging channel.

EXAMPLE 6.4 Set-up and voice channels per cell

Calculate the number of set-up channels and voice channels per cell for a cellular system having a total spectrum allocation of 60 MHz which uses two 25-kHz simplex channels to provide full duplex set-up and voice channels. Assume that the system is designed with nine cell frequency-reuse pattern and 1 MHz of the total spectrum is exclusively allocated for set-up channels.

Solution

Total allocated RF spectrum bandwidth = 60 MHz (given)
Channel bandwidth per simplex channel = 25 kHz (given)

Step 1. *To determine duplex channel bandwidth*

Channel bandwidth per simplex channel = 25 kHz
Number of channels in a duplex link = 2
Therefore, duplex channel bandwidth = 25 × 2 = 50 kHz

Step 2. *To determine number of full-duplex channels*

Number of full-duplex channels = Total bandwidth/duplex channel bandwidth

Number of full-duplex channels = 60 MHz/50 kHz

Hence, total number of duplex channels = 1200 channels

Step 3. *To determine total number of set-up channels*

Allocated RF bandwidth for set-up channels = 1 MHz (given)

Duplex channel bandwidth = 50 kHz (As calculated in Step 1)

Total number of available set up channels = 1 MHz/50 kHz = 20

Step 4. *To distribute number of set-up channels per cell*

Number of cells in one cluster = 9 (given)

Total available 20 number of set-up channels can be distributed among nine cells in a cluster as

– 7 cells can have 2 set up channels each, and

– remaining 2 cells can then have 3 set-up channels each.

which means a total ($7 \times 2 + 2 \times 3 =$) 20 set-up channels in a system

Step 5. *To determine total number of voice channels*

Available RF bandwidth for voice channels = 60 MHz–1 MHz

= 59 MHz

Total number of available voice channels = 59 MHz / 50 kHz

= 1180

Step 6. *To distribute number of voice channels per cell*

Number of cells in one cluster = 9 (given)

Total 1000 number of available voice channels can be distributed among nine cells in a cluster as

– 8 cells can have 131 voice channels each, and

– remaining 1 cell can then have 132 voice channels.

which means a total ($8 \times 131 + 1 \times 132 =$) 1180 voice channels in a system.

6.1.3 Selecting a Voice Channel

The assignment of certain sets of voice channels in each cell is based on the aspect such that it causes minimum cochannel interference and adjacent channel interference. For mobile-originating calls, the mobile subscriber selects a cell-site based on its received signal-strength indicator (RSSI) value on a set-up channel. The call request from a mobile subscriber is received on the reverse set-up channel (access channel) by the cell-site. The cell-site RSSI scans the incoming signals and determines which sector out of three sectors of a cell has the strongest received signal strength at that time. The MTSO then assigns a free voice channel from among those voice channels designated in that sector.

Generally, a separate set-up channel is assigned to each sector of a cell. For mobile-terminating calls, a call is received by the mobile unit on one of the forward set-up channels (paging channel). When a mobile unit responds to the cell-site, the cell-site RSSI will measure the signal strength of the incoming signal from the three sector antennas and find the strongest sector in which the voice channel can be assigned to the mobile unit.

6.2 CHANNEL-ASSIGNMENT STRATEGIES

The main objective of channel-assignment techniques is to stabilise the fluctuations in the probability of call blockage over the entire coverage area of a cellular network over a period of time. A classical approach to frequency-assignment problems does not enable this task to be performed in an efficient way when applied to the frequency planning of cellular networks since it does not consider the cumulative effect of cochannel interferers. The criteria of frequency planning in a cellular system includes how to allocate (or assign) channels to each cell, and then how to re-assign channels with varying traffic load, while maintaining the desired signal quality.

Channel assignment refers to the allocation of specific (fixed) channels to cell-sites on long-term basis, and the allocation of specific channels to mobile units on a short-term basis during a call. One way is fixed (static) channel assignment in which the set of all channels is partitioned into 'n' groups of frequency channels (fixed) and each cell is assigned to one of these 'n' groups of frequency channels. This arrangement is acceptable as far as the cellular system has uniform distribution of mobile subscribers and fully meets the capacity requirement within the available radio resources.

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In order to avoid the interference problems from cochannel and adjacent channel interference in a cellular system, an intelligent and efficient frequency planning is needed. The frequency channels may be assigned dynamically in such a way so as not to cause any interference to on-going communications.

The mobile traffic density of a coverage area must be determined before the frequency planning of a system is done. The traffic pattern in busy hours can be confined to different zones within the service area. If the traffic pattern predominates over the simple signal coverage pattern, cell-site selection will be based on the traffic pattern. Initially, the cell sites should be located at the centre of zones of heavy mobile traffic. While the system is operating, the statistical call traffic data is updated at each cell-site to correlate with the mobile traffic data. This information is useful for determining whether new cell splitting is needed or not. If cell splitting is needed, then the location and number of transceivers at the new split cell-site is determined. These decisions are all related to frequency channel assignment.

6.3 FIXED CHANNEL ASSIGNMENT

Channel set is a collection of channels that could be assigned at one cell. In fixed channel assignment (FCA), each cell assigns its own frequency channel to the mobile subscribers within its cell. Channel assignment is primarily based on causing least cochannel and adjacent channel interference in the cellular system. The channel assignment for each voice call is determined by MTSO on a short-term basis. In a fixed channel assignment, the set-up and voice channels are usually assigned to the cell-site for relatively long periods. Channels in a channel set are usually 21 channels apart and must meet minimum frequency spacing requirements of a multi-channel transmitter combiner. Channels are usually numbered in order of increasing frequency. Regardless of the number of channels in a channel set, the highest channel set is frequency-adjacent to the lowest channel set.

There are certain advantages of fixed channel assignment such as fixed parameters (power, frequency) for transceivers, and reasonably good performance under uniform and/or high traffic loads as cells independently decide their channel allocation decisions. However, the number of channels in a channel set has to be at least seven for hexagonal cellular configuration. At run-time, channel borrowing may be necessary but the number of channels in a cell dictates the upper limits for channel borrowing. This results into major problem of hot spots or localised congestion with fixed channel assignment.

Example 6.5 | Fixed channel assignment in a cellular system

In a cellular system with a 7-cell cluster, 168 voice channels are available. Determine the assignment of voice channels to each cell/sector if

(a) omnidirectional antennas are used at the cell-site
(b) 3-sector 120° directional antennas are used at the cell-site
(c) 6-sector 60° directional antennas are used at the cell-site
Comment on the results obtained.

Solution

Number of available voice channels = 168 (given)

Number of cells in a cluster = 7 (given)

(a) *To determine the assignment of voice channels in omnidirectional cells*

Step 1. *When omnidirectional antennas are used at each cell-site, all the available voice channels are assigned in a cluster. Assuming uniform traffic distribution in each cell of a cluster, equal number of voice channels will be assigned to each cell.*

Step 2. *Number of voice channels assigned in each cell*

Number of voice channels per cell = total available channels/cells in a cluster

Number of voice channels assigned in each cell = 168/7 = 24

(b) *To determine the assignment of voice channels in 3-sectored cells*

Step 3. *When 3-sector 120° directional antennas are used at each cell-site, all the available voice channels are assigned in a cluster. Assuming uniform traffic distribution in each sector of a cell of any cluster as well as in each cell of a cluster, equal number of voice channels will be assigned to each of three sectors.*

Step 4. *To calculate number of sectors in a cluster*

Number of sectors in a cell = 3 (given)

Total number of sectors in a cluster = $7 \times 3 = 21$

Step 5. *Number of voice channels assigned in each sector*

Number of voice channels per sector = total channels/sectors in a cluster

Number of channels assigned in each sector = 168/21 = 8

(c) *When 6-sector 60° directional antennas are used at each cell-site*

Step 6. *All the available voice channels are assigned in a cluster. Assuming uniform traffic distribution in each sector of a cell of any cluster as well as in each cell of a cluster, equal number of voice channels will be assigned to each of six sectors.*

Step 7. *To calculate the number of sectors in a cluster*

Number of sectors in a cell = 6 (given)

Total number of sectors in a cluster = $7 \times 6 = 42$

Step 8. *Number of voice channels assigned in each sector*

Number of voice channels per sector = Total channels /sectors in a cluster

Number of channels assigned in each sector = 168/42 = 4

Comment on the results The above example clearly demonstrates that fixed channel-assignment scheme in a uniformly distributed traffic arrangement is quite simple.

There are different types of fixed channel-assignment schemes:

– Adjacent-channel assignment scheme
– Channel-sharing scheme
– Channel-borrowing scheme
– Overlapped cells-based channel assignment scheme

6.3.1 Adjacent Channel-Assignment Scheme

Adjacent-channel assignment scheme includes neighbouring-channel assignment scheme and next-channel assignment scheme. Neighbouring channels are up to four adjacent channels on each sides of the desired channel.

In an omnidirectional cellular system, next channels including neighbouring channels cannot be assigned in the same cell as well as in the six neighbouring cells in the cluster of size seven. Adjacent channel–assignment scheme in omnidirectional cells is shown in Fig. 6.1.

In a directional antenna cellular system design, if one channel is assigned to a sector, next channels cannot be assigned to the same sector or to the other two sectors in the same cell. The next channel is also not

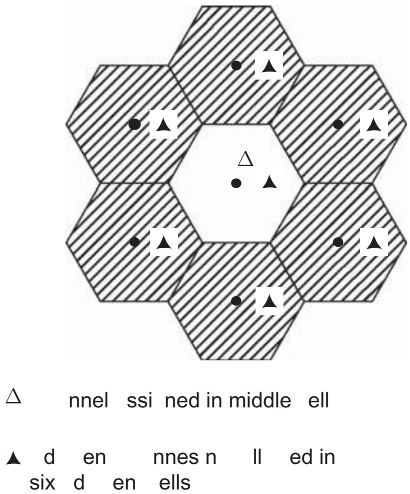

Fig. 6.1 *Adjacent channel assignment in omnidirectional cells*

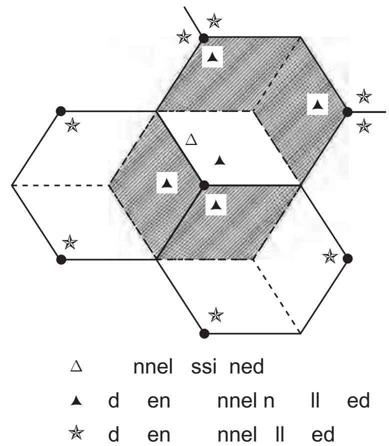

Fig. 6.2 *Adjacent channel assignment in three-sectored cells*

assigned to its immediate neighbouring sectors of other adjacent cells. Adjacent channel assignment in three-sectored cells is shown in Fig. 6.2. Adjacent channel assignments in shaded sectors are not allowed.

Sometimes the next channels are assigned in the next immediate sector of the same cell in order to increase capacity. If channel assignment scheme is properly designed, the system performance can still be maintained within acceptable limits.

EXAMPLE 6.6 Adjacent fixed channel assignment in AMPS cellular system

In the US AMPS cellular system, a bandwidth of 25 MHz is allocated for both the uplink and downlink to provide full duplex transmission. There are two service providers in any operating region. Each service provider has 12.5 MHz of spectrum available. Create a fixed channel assignment chart, assuming the cellular system is configured with a frequency reuse factor of seven.

Solution

Each service provider is allocated half of the standard 824–849 MHz band in the uplink or reverse channel, and half of the standard 869–894 MHz band in the downlink or forward channel. Some of the major air-interface parameters of the US AMPS cellular system are given as below.

Uplink frequency band (Mobile Tx)	824 MHz–836.5 MHz
Downlink frequency band (cell-site Tx)	869 MHz–881.5 MHz
RF spectrum bandwidth	12.5 MHz on either end
Duplex separation (Tx-Rx)	45 MHz
Channel spacing	30 kHz
Total number of duplex channels available	(12.5 MHz/30 kHz) = 416

In an AMPS analog cellular system, each mobile subscriber uses one duplex channel for a voice communication link. Of the total of 416 channels, 21 channels are designated as set up channels and remaining 395 channels are used for carrying voice traffic. With a frequency reuse factor of seven and employing fixed channel assignment, the seven sets of voice channels can be created to minimise the cochannel interference as given in the following chart.

Cell 1	Channel nos. [1, 8, 15,	.372, 379, 386, 393]
Cell 2	Channel nos. [2, 9, 16,	.373, 380, 387, 394]
Cell 3	Channel nos. [3, 10, 17,	...374, 381, 388, 395]
Cell 4	Channel nos. [4, 11, 18,	...375, 382, 389, ---]
Cell 5	Channel nos. [5, 12, 19,	376, 383, 390, ---]
Cell 6	Channel nos. [6, 13, 20,	.377, 384, 391, ---]
Cell 7	Channel nos. [7, 14, 21,	.378, 385, 392, ---]

The first three sets of cells will have 57 voice channels each whereas the remaining four sets of cells will have 56 voice channels (one voice channel less) from the total available 395 voice channels.

EXAMPLE 6.7 **Adjacent fixed channel assignment in GSM cellular system**

In the GSM cellular system, a pair of 25 MHz band is allocated to the uplink and downlink channel to provide full duplex transmission. Each radio carrier uses a 200 kHz bandwidth, and each carrier contains eight time slots capable of supporting eight voice users. If the cluster size is four, create a fixed channel-assignment chart.

Solution

Uplink frequency bandwidth	25 MHz
Downlink frequency bandwidth	25 MHz
Carrier bandwidth	200 kHz
Total number of carriers	25 MHz/200 kHz = 125 carriers

Potentially, there are 125 carriers, but in practice only 124 of them are used. Let the carriers or channels be numbered as 1, 2, 3, , 124.

Since the cluster size is four, four sets of channels for corresponding four cells of a cluster can be created to minimise the cochannel interference as given in the following chart.

Cell 1	Channel nos. [1, 5, 9,	...113, 117, 121]
Cell 2	Channel nos. [2, 6, 10,	.114, 118, 122]
Cell 3	Channel nos. [3, 7, 11,	.115, 119, 123]
Cell 4	Channel nos. [4, 8, 12,	.116, 120, 124]

The fixed channel-assignment technique is simple to implement. The traffic in the cellular network is uniform, so that the number of active mobile subscribers in each cell is same, and it remains constant with time. Fixed channel assignment is also an optimum channel-assignment technique. Assuming that the traffic density in all the cells is the same, the channel assignment algorithm is very straightforward. The total numbers of available channels are simply divided by the cluster size of the system and assign this number of channels to each cell uniformly. Knowing the traffic density data and the number of channels assigned in one cell, the call-blockage probability in that cell can be determined. The probability of call blockage in all other cells and consequently the average probability of call blockage in the whole cellular network will be the same as that of one cell. In fact, the channel-assignment algorithm and the calculation of probability of call blockage are performed independently.

But the fact is that in each cell, the traffic changes with time due to the movement of mobile subscribers from one cell to another. This results in higher probabilities of call blocking in some cells due to non-availability of voice channels and lower values in other cells. This leads to poor utilisation of the available frequency spectrum. To equalise the utilisation of channels in all the cells, the obvious solution is that the cells with higher traffic load should somehow use the free channels available in low traffic cells. This is possible by a nonuniform assignment of channels to cells in the first place. With a nonuniform channel-assignment technique, the call-blockage probability must be added as a criterion for the channel-assignment algorithm. The relation between the number of channels and the call-blockage probability is a complex function, thus making channel-assignment algorithm significantly more complex. Moreover, the regular frequency-reuse pattern configuration does not work practically because these are based on the assumption that the channels are fixed and the cochannel interference can be easily computed on the basis of having the same number of channels per cell. With nonuniform number of channels per cell, frequency reuse pattern configuration becomes much more complex.

6.3.2 Channel Sharing Scheme

When a particular cell needs more number of channels in order to meet the increased traffic to be served, the channels of another sector at the same cell-site can be shared to meet the short-term overload traffic. In a sectored cellular system configuration, channel sharing can be done from one of the two adjacent sectors of the neighbouring cells. Shared channels can be returned back as and when the channels become available in the shared sector. This scheme is known as an ordered channel-assignment scheme with rearrangement.

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Channel sharing is always cyclic so as to follow the adjacent-channel assignment algorithm.

Another alternate scheme is channel assignment with sharing and reassignment. This scheme ensures that channel-sharing arrangement causes minimum impact on call-blocking probability in neighbouring cells. Reassignment of shared channels is done to provide maximum assistance to the neighbouring cells to meet the temporary increased traffic demand. The channel can also be ordered based on which channels provide better performance. The configuration of transmitter channel combiner at the cell-site should be flexible in real time so as to adapt to increased number of transceivers for increased number of channels at a time. Channel sharing always increases the trunking efficiency of the channels.

The difficulty in implementing effective channel-sharing scheme is that all cell locations may not have non-parent cell-sites, and there may be low quality connections to cell-sites which are not parents. The channels may be shared between adjacent cells. It should be noted that channels are not re-allocated but re-used by a different cell. Channel sharing may be considered as a temporary channel-borrowing strategy between one cell with cell-site of a neighbouring cell. Generally, non-parent cell-sites are chosen based on close proximity. Moreover it requires local channel management between adjacent cells. Also, it requires precise control of transmitted power of mobile equipments and cell-sites because there is increased probability of adjacent channel interference.

There are relative advantages and disadvantages of different channel-sharing schemes in terms of proper utilisation of total available channels, total carried traffic by the system, and assignment complexity. Usually, decisions are made based on the system parameters and traffic behaviour.

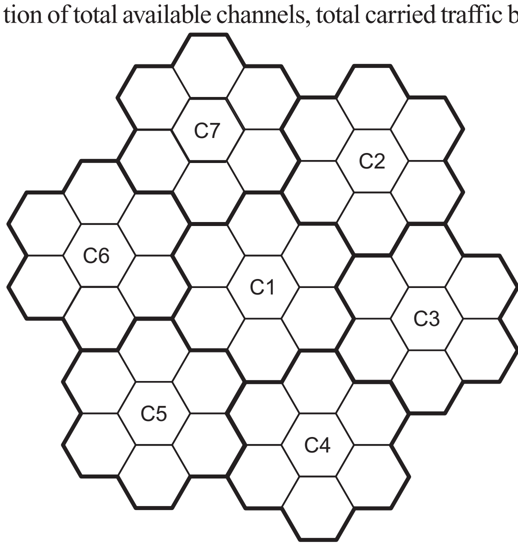

Fig. 6.3 *7-cell-based cluster omnidirectional cellular system*

6.3.3 Channel-Borrowing Scheme

The channel-borrowing scheme is used primarily for slow-growing systems on a long-term basis as an alternate to costly cell-splitting technique to handle increased traffic. One approach to address increased traffic of either mobile-originating calls or hand-off calls in a cell is to borrow free available channels from neighbouring cells. A simple channel-borrowing scheme implies that if all channels assigned to a cell have already been used, then additional channels as per the current need can be borrowed from any other cell (preferably adjacent cells) that has some free unused channels. In addition, the central cell-site can also borrow channels from neighbouring cells. The extent of borrowing channels depends on the traffic density in the area.

For example, in the seven-cell-based cluster scheme of omnidirectional cellular system as shown in Fig. 6.3, a cell of cluster C1 can borrow channels from cells of adjacent clusters C2, C3, C4, C5, C6, or C7 in the first tier.

While borrowing channels, there is a need to make sure that there is no interference with cells associated with these clusters, which are within reuse distance of the cluster C1.

EXAMPLE 6.8 | Impact of channel borrowing in omnidirectional cellular system

In a cellular system based on 7-cell reuse pattern, each cell-site is designed for omnidirectional coverage. It is implied that each corresponding cell of all clusters uses the same group of channels to maintain minimum desired frequency reuse distance or an acceptable C/I ratio.

Refer to Fig. 6.4 to demonstrate the impact of channel borrowing and cochannel interference in omnidirectional cellular system, if cell c5 of cluster C3 borrows a channel from its adjacent cell c3 of cluster C1.

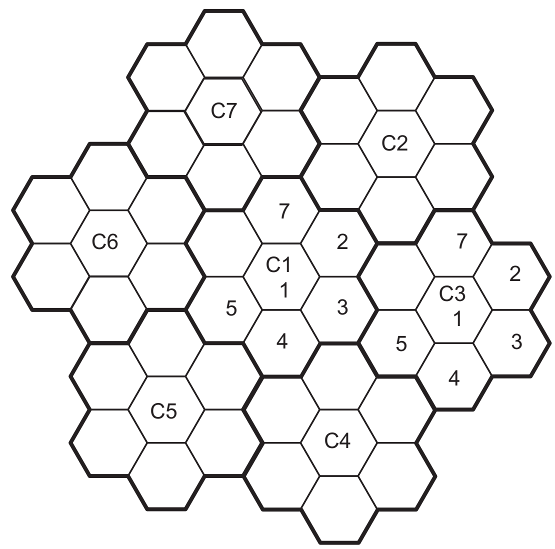

Fig. 6.4 | *Channel borrowing in an omnidirectional cellular system*

Solution

When cell c5 of cluster C3 borrows a channel from its adjacent cell c3 of cluster C1, these two cells happen to be adjacent cells although they belong to different clusters. This may cause potential violation of reuse distance, and there could be interference between the borrowed channel in cell c5 of cluster C3 with the same channels of all c3 cells of clusters C2, C4, C5, C6, and C7, even including C3 itself.

The following additional observations are made in the given scenario:

– Cluster C3 has neighbouring clusters C1, C2, and C4, whereas clusters C5, C6, and C7 are a distance apart.
– The distance between cell c5 of cluster C3, in which the borrowed channel will be used, and cell c3 of clusters C5, C6, and C7 still satisfy the frequency-reuse distance requirements.
– The distance between cell c5 of cluster C3, in which the borrowed channel will be used, and cell c3 of clusters C2, C3, and C4 violate the reuse distance from cell c5 of cluster C3.
– Moreover, the borrowed channel from cell c3 of cluster C1 is already being used in cell c3 of cluster C3 itself that can cause maximum cochannel interference as compared to that from corresponding cells c3 of clusters C2 and C4.

Similar results would be obtained if channels were borrowed from adjacent cells of the same cluster. The only solution to minimise interference in channel-borrowing scheme is the use of sectored-cells instead of omnidirectional cells.

6.3.4 Channel Borrowing in Sectored Systems

The channel-borrowing scheme can be implemented from one cell-site sector to another sector of the same cell-site. A 7-cell system is normally configured with three 120 sectors per cell. The total number of available channels per cell are divided into three equal subsets. Cell sectoring can be used to assign channels on a temporary basis. While borrowing channels from other cells, there are two criteria of verifying potential interference and possible prohibition of using them.

– The first criterion is that the reuse distance with other nearby clusters using those borrowed channels should be verified.
– The second criterion is that the directions of sectors of all cells not satisfying the reuse distance should be verified.

Such verification would ascertain any potential cochannel interference with other cells and ensure satisfactory operation of the overall system even after borrowing the channels in sectored cellular configuration. It is explained with the help of an example illustrated next.

EXAMPLE 6.9 | Impact of channel borrowing in 3-sectored cellular system

In a cellular system based on 7-cell reuse pattern, each cell is partitioned into a three-sector format. Figure 6.5 shows some of the cells with three-sectors for simple illustration.
What will be the impact of channel-borrowing and cochannel interference in this case? Comment on the clear-cut advantage that sectored cells have over omnidirectional cells.

Solution

Frequency reuse pattern = 7 (given)
Number of sectors in each cell = 3 (given)

A step-by-step procedure to illustrate the impact of channel-borrowing and cochannel interference in the given case is described below.

Step 1. *Seven adjacent clusters of the first tier of 7-cell reuse pattern employing 3-sector configuration in each cell that could have cochannel interference are shown in Fig. 6.6.*

Step 2. *Assume that each corresponding sector of all clusters uses the same group of channels to maintain minimum desired reuse distance or acceptable C/I value.*

Step 3. *Let sector 'x' of cluster C3 need to borrow channels from an adjacent cell, say, from sector 'a' of cluster C1.*

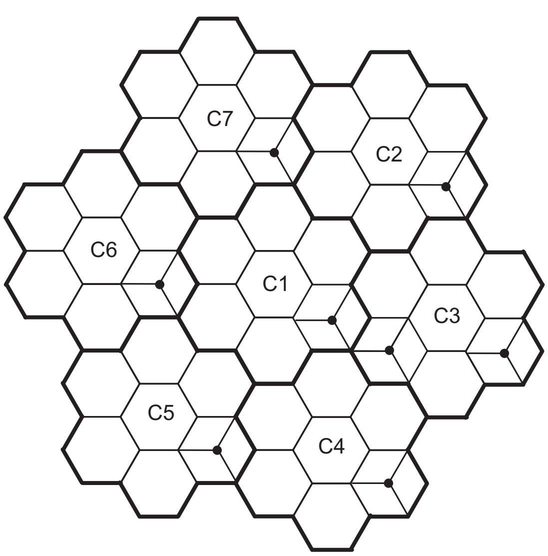

Fig. 6.5 | *A 7-cell 3-sector cellular configuration*

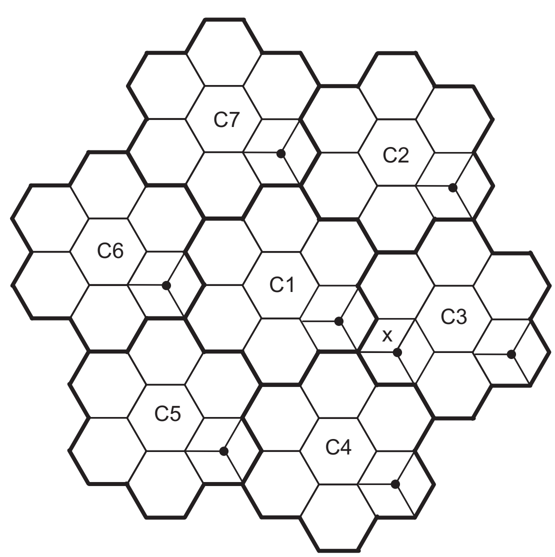

Fig. 6.6 | *Channel borrowing in 3-sectored cellular system*

Step 4. *But this may cause potential violation of reuse distance, and there could be interference between the borrowed channel in sector 'x' with the same channels of all 'a' channels of clusters C2, C3, C4, C5, C6, and C7.*

Step 5. *From Fig. 6.6, it can also be observed that the distance between sector 'x' of cluster C3 and sector 'a' of clusters C5, C6, and C7 satisfy the minimum reuse distance requirements.*

Step 6. *The distance between sector 'x' of cluster C3 and sector 'a' of clusters C2, C3, and C4 does not satisfy the minimum reuse distance requirement.*

Step 7. *Therefore, it is needed to look at the directions of sector 'a' for clusters C2, C3, and C4 also.*

Step 8. *Clearly, the 'a' sectors for both clusters C2 and C4 are in different directions from 'x' and simultaneous use of the same channels in these areas will not cause any additional interference, as expected.*

Step 9. *Now the only issue that needs to be resolved is the interference between sector 'x' and sector 'a' of cluster C3.*

Step 10. *Even though the reuse distance is not satisfied between sector 'x' and sector 'a' of cluster C3 and moreover, they belong to the same cluster C3 but their directions are such that most likely they would not interfere with each other.*

Step 11. *Similar results would be obtained if channels were borrowed from adjacent cells of the same cluster.*

Comments on the advantages of cell-sectoring over omnidirectional cell. If the cells are omnidirectional (that is, no sectorisation of cells is done), then borrowed channels will be used in the cell marked '*x*' and would cause interference with the cell '*a*, *b*, *c* sectors' of clusters C2, C3, and C4. These borrowed channels cannot be used in these clusters as well. Therefore, sectored cells have a clear-cut advantage over omnidirectional cells.

Channel coordination is much easier in cell sectorisation to avoid cochannel interference than in cell splitting. But moving from one sector to another during a call may require frequent hand-offs. This in turn requires additional resources in terms of channel availability.

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Cell sectorisation serves the same purpose as the channel-borrowing scheme. Sometimes it is referred as delayed cell splitting. However, trunking efficiency decreases in sectorisation for the same number of channels.

6.3.5 Channel Prioritising

Borrowing channel assignment with channel-ordering algorithm uses fixed channel assignment as a normal assignment condition in the beginning. When all the fixed channels are occupied, then the cell under consideration borrows the minimum required number of channels from the neighbouring cells. There is an alternate way to divide the available channels per cell into two groups, one group of channels assigned to each cell permanently (fixed channels) and the second group of channels kept reserved for borrowing by its neighbouring cells. The ratio between the two groups of channels is determined in advance, based on estimated traffic in the system. Another alternative scheme is to assign priorities to all the available channels of each cell. The highest priority channels are used in a sequential order for local calls in the cell while channel borrowing is done starting from lower priority channels. This is called borrowing with channel-ordering scheme.

Channel-borrowing scheme with directional channel locking deals with borrowing of that particular channel which is available in nearby cochannel cells in order to minimise cochannel interference under all conditions. But this scheme imposes additional constraints, and the number of available channels is reduced. Some cells get saturated with their assigned channels. In other words, they cannot allow any more incoming calls or hand-off requests, whereas neighbouring cells may have free channels available at that time.

6.3.6 Overlapped Cells-based Assignment

In practical cellular communication systems, cell structures are non-uniform because of varying traffic conditions in different cell-site locations. An underlaid small cell is sometimes commissioned at the same cell-site over a large cell. This permits the two frequency groups to reuse the channels in two different

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Channel borrowing may not be needed at all times of operation. Practical situations for channel borrowing may be required during busy traffic hours, typically, at 9 a.m. and 5 p.m.; traffic on highways outside the big cities on long weekends; traffic-jam conditions or places of accident; events like games in a stadium.

cell-reuse patterns of the same cluster size. The channels assigned in the small cells have more protection against cochannel interference. A set of frequencies used in an overlay area (coverage area of large cell) will differ from a set of frequencies used in an underlay area (coverage area of small cell) in order to avoid cochannel and adjacent-channel interference. The hand-offs are implemented between the large cells and small cells. An example of an overlapped cells-based cellular system is shown in Fig. 6.7.

Let a larger hexagonal-shaped cell be split into seven small hexagonal-shaped cells, marked as c1 to c7, with separate cell-sites placed at the centre of each small cell. In such an overlapped cellular system, the number of channels assigned to each cell (large or small) depends on the following parameters:

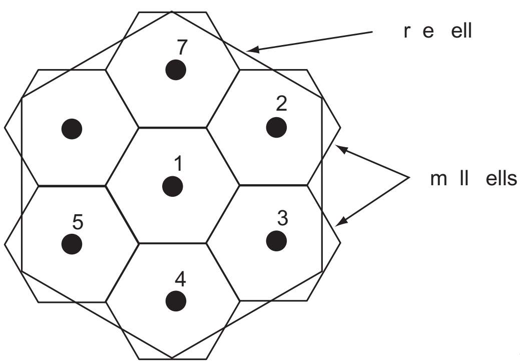

Fig. 6.7 *Overlapped cells-based system*

- total number of available channels
- the area to be covered
- the average speed of the mobiles in each cell
- the call-arrival rate
- the call duration
- desirable call-blocking probability
- desirable call-dropping rate
- the number of channels reserved for handoffs

One approach to handle increased traffic in a cell is to split it into a number of smaller cells inside a cell. Such partitioned smaller cells are called *microcells* and *picocells*. One way to assign channels for the larger cells and smaller cells is to characterise the mobility of each mobile terminal into slow-moving and fast-moving mobiles. For slow-moving mobiles, channels are assigned from one of the smaller cells, based on its present location. Fast-moving mobiles are assigned channels from the larger cell because fast-moving mobiles would have more frequent hand-offs if channels associated with the smaller cells were assigned for the same. Therefore, channel assignment from the larger cell or smaller cells is matched with the speed of the mobiles, assuming the speed will not change drastically during the call duration. As such, optimisation of designing such a system is quite complex.

Another alternative arrangement to using larger cells and smaller cells is to change the logical structure dynamically. When the traffic is small, only the larger cell is used and other smaller cells are switched off for the time being under the central control for low traffic, as traffic increases in one or more parts of a larger cell which may lead to forced call blocking. This may happen due to unacceptable level of cochannel interference (reduced *C/I* value below acceptable value) or unavailability of radio resources. At this moment, the corresponding smaller cells are activated. If traffic decreases, the respective cell-sites of smaller cells are deactivated, thereby automatically adapting to instantaneous call-traffic density and reducing the call blocking or call-dropping probability. This also has an advantage in significant reduction in the number of occurrences of hand-offs.

Facts to Know!

The optimal assignment of channels among the larger cell and smaller cells in an overlapped cell-based system is a complex function of numerous parameters. The rate at which activation and deactivation of smaller cells can be done is also very difficult to optimise.

EXAMPLE 6.10 | Overlapped cells-based system

A cellular system is originally designed with a larger hexagonal cell of radius $R_2 = 20$ km. In order to optimise the channel assignment in varying traffic, it is overlaid with seven smaller regular hexagonal cells of radius R_1 each, as shown in Fig. 6.8.
(a) What is the size of each smaller cell?
(b) How is the received signal strength influenced by such an overlaid cellular system?

Solution

Radius of larger cell, $R_2 = 20$ km (given)
Number of smaller cells = 7 (given)
Radius of smaller cell, $R_1 = ?$

(a) *To determine the size of each smaller cell*

Step 1. *To find the radius of the smaller cell, R_1*

From the geometry of the given figure, it is observed that

$$R_2 = 3\,(\sqrt{3}/2)\,R_1 = 2.6\,R_1$$

Or,
$$R_1 = R_2/2.6$$

Hence, the radius of the smaller cell, $R_1 = 20\text{ km}/2.6 = 7.7$ km

Step 2. *To determine the area of the smaller cell*

The size or area of smaller hexagonal cell $= 3\sqrt{3}/2 \times R_1^{\,2}$

Putting the value of $R_1 = 7.7$ km, we get

The area of smaller hexagonal cell $= 3\sqrt{3}/2 \times (7.7\text{ km})^2$

Hence, size of each smaller cell $= 154\text{ km}^2$

(b) *Impact on the received signal strength by such an overlaid cellular system*

In a larger cell, the maximum radio coverage range required is 20 km as compared to 7.7 km in a smaller cell. So a high-power transmitter is required at the centre of the larger cell, while low-power transmitters can be used at the centre of each smaller cells. The total number of channels available has to be dynamically assigned between the larger cell and smaller cells so as not to interfere with each other as simultaneous use of channels in larger cell and smaller cells may take place.

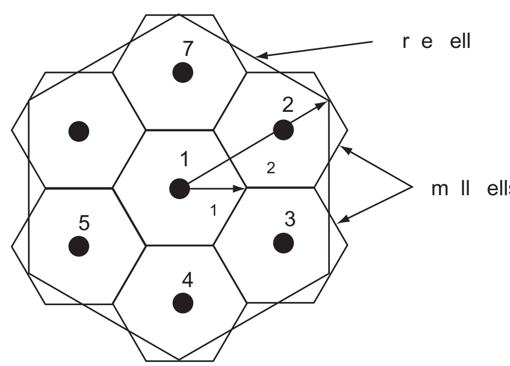

Fig. 6.8 | *Illustration of overlaid cells of Example 6.10*

6.4 DYNAMIC CHANNEL ASSIGNMENT

In order to achieve optimum system capacity with limited frequency spectrum, many dynamic channel-assignment schemes have been proposed to allocate the channels more efficiently. In a cellular system, a mobile subscriber moves from one cell to another, and continuation of communication link is ensured with suitable hand-off mechanism. This demands for additional and flexible radio resources utilisation. One way to ascertain minimum blocking probability is to increase the number of channels per cell. Then every cell would expect to have a large number of channels. However, because a limited frequency band is allocated for cellular communication, there is an upper limit to the

> **Facts to Know!**
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> During the call, the system monitors the quality of channel being used by measuring S/N ratio or BER. If the defined level of the quality is exceeded, a hand-off request is initiated. This is a typical dynamic channel assignment operation involving an intracell hand-off.

maximum number of channels, thereby restricting the number of available channels that can be assigned to each cell. Another way is nonuniform fixed channel assignment based on the amount of traffic expected to be served in different cells as per the statistical traffic data. Another alternative arrangement could be dynamic assignment of channels to different cells, as per the current demand. This may be done from a central pool of channels, or a combination of both fixed channel assignment and dynamic channel assignment.

In dynamic channel assignment, the central common pool maintains all the available channels. Channels are assigned dynamically as new requests for radio resource (for a fresh originating call or hand-off of existing call) arrive in the system. This also implies that when the use of assigned channel is completed, the channel currently in use is returned to the central pool.

Let cell C_i use any channel or a group of channels. The same channel or group of channels can be re-used by another cell C_j (where i and j are different integer values). It is mandatory that both the cells maintain minimum re-use distance as per the cluster size. In this way, it is a fairly straightforward method is to select the most appropriate channel for any new request for a channel with the aim of minimising the interference. This is possible because assignment of different traffic channels for on-going traffic is known. In dynamic channel assignment, no fixed channels are assigned to each cell. Therefore, any channel out of available channels can be assigned to any cell on need basis, as long as interference constraints in that cell can be satisfied so as to maintain the desired signal-quality requirements.

6.4.1 Channel Borrowing

A channel is borrowed from the central pool of all available channels by a cell-site for use. The assignment of a free channel must consider various aspects such as reuse distance, usage frequency of the candidate channel, average blocking probability of the overall system, future blocking probability in the vicinity of the cell, and instantaneous channel-occupancy distribution. When the call is completed, the borrowed channel is returned to the central pool. The basic dynamic channel assignment has a self-originating channel-assignment algorithm based on dynamic real-time measurement of interference levels. This is usually performed at the mobile unit in order to reduce the computational load and the complexity of the system. All cell-site transceivers have access to the complete set of available channels even if each cell-site is equipped with lesser number of transceivers.

Several variations of dynamic channel-assignment have been implemented successfully in cellular systems. Various dynamic channel assignment algorithms differ in the selection of the preferred channel among the available channels. If not-so-good-quality channels are available at the preferred cell-site, the system looks for a new suitable cell-site having a good-quality channel with a sufficient signal level. Still if no channel is found to be acceptable, the call is blocked.

The *forcible-borrowing channel-assignment algorithm* is based on assigning a channel dynamically but obeying the rule of minimum reuse distance. Forcible-borrowing channel assignment can also be applied while accounting for the forcible borrowing of the channels within a fixed channel set to reduce the possibility of cochannel assignment in a reuse cell pattern. In a similar way, the *adaptive channel-assignment scheme* employs a subset of total available channels reserved to meet the hand-off requests. This has shown significantly better results in terms of lost calls during handoffs as well as increasing the capacity of digital cellular systems by about 100 per cent as compared to the traditional fixed-channel assignment.

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Generally, it is difficult in maintaining cochannel reuse distance at the minimum required value everywhere in the system. For a given reuse distance, cells can be identified that satisfy minimum reuse distance criteria. Channel is available only if every cell within reuse distance is not using the same channel.

Channel borrowing in dynamic channel assignment impacts channel locking. Channel locking is used to prevent an increase in cochannel interference, that is, cell-sites within the required minimum channel reuse distance from a cell-site that borrows a channel cannot use the same channel. Moreover, the number of channels available for borrowing by a cell-site is limited since the channel can be borrowed by a cell-site only when it is idle in all of the cell-sites within the required channel reuse distance of the borrowing cell-site. All these cells could be assigned the same channel and thus form a set of cochannel cells. If a cell needs to support a new call, then a free channel from the central pool is used in such a way so as to maximise the number of channels in its cochannel set. Minimum channel reuse distance is difficult to maintain which leads to ineffective frequency reuse.

6.4.2 Advantages and Disadvantages

Centralised dynamic channel allocation schemes can theoretically provide near optimal performance. But the extent of large computation and communication among cell sites leads to excessive system latencies and makes them almost impractical. An alternate scheme is distributed dynamic channel-assignment schemes, which are primarily based on one of the three parameters: reuse distance, *C/I* value, and signal-to-noise ratio. Based on *C/I* values, channels are assigned to a new call if the anticipated *C/I* is above acceptable value. This could cause the *C/I* for some existing calls to deteriorate and hence those would require finding new channels that could satisfy a desired *C/I*.

Dynamic channel-assignment schemes cannot maximise overall channel reuse because they handle randomly generated new calls. Therefore, dynamic channel assignment schemes are observed to carry less traffic as compared to fixed channel assignment. There may be a need of reassigning channels and change channels for existing calls if that minimises the distance between cochannel cells.

The advantages of dynamic channel assignment include the following:

– Adaptability to traffic load changes
– Adaptability to cellular network environment
– Better utilisation of available channel resources
– Improved load balancing
– Better performance in low traffic conditions

There are certain disadvantages associated with dynamic channel assignment such as

– Need of an overhead for the locking attribute, that is, a channel can be assigned only if all the cells within the reuse distance cluster do not use the same channel
– Requirement of exhaustive checking
– Need to keep track of channels
– Efficient network channel management requirement
– Channel assignment delay
– Non-implementation of some reuse patterns due to difficulty in maintaining reuse distance because of dynamic nature of channel-allocation algorithm
– Requirement of tunable frequency base stations and mobile terminals with varying power ranges

6.5 HYBRID CHANNEL ASSIGNMENT

In hybrid channel-assignment scheme, each cell has a static channel set as well as dynamic channel-borrowing capability while maintaining the minimum reuse distance requirement. In fact, hybrid channel assignment is a combination of fixed channel assignment and dynamic channel assignment. A part of total frequency channels will use fixed channel assignment and the rest will use dynamic channel assignment. The channels assigned under fixed channel assignment scheme are exclusively used by its own cell. A request for a channel from the dynamic channel set is initiated only when a call has exhausted all the available channels in the fixed channel set. A channel from the dynamic channel set can be selected by employing any of the dynamic channel-assignment algorithms.

The relative proportion of the number of fixed and dynamic channels mainly depends on traffic characteristics. It may be desirable to vary this value as per estimates of instantaneous traffic-load distributions in the cell. A lot of computation is required if simulation is to determine the behaviour of a large cellular system and an analytical approach is desirable. For example, it has been observed that for a fixed-channel-to-dynamic-channel ratio of 3:1, the hybrid channel-assignment scheme leads to better service than the fixed channel-assignment scheme for traffic up to 50%. Beyond 50% traffic load, the fixed channel-assignment scheme performs better.

> **Facts to Know!**
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> Hybrid channel-assignment scheme has the advantages of both static and dynamic channel-assignment schemes. It has low overhead of channel management, lower run-time overhead, reduction in hot spot, and improved channel-load balancing. On the other hand, if hybrid channel assignment is not coordinated properly, the benefits of both static and dynamic channel assignment strategies are lost.

To assign a channel from a dynamic channel set effectively, a centralised control is used with up-to-date traffic-pattern information. There are two different methods used in assigning dynamic channels: scheduled and predictive. In scheduled dynamic channel assignment, a priori estimates about variation in traffic in terms of location and time are needed to schedule dynamic channels at predetermined peaks of traffic change. In a predictive dynamic channel assignment, the traffic intensity and blocking probability is monitored in each cell at all the time so that dynamic channels can be assigned to each cell according to its needs.

Channel borrowing in hybrid channel assignment has similar issues as in dynamic channel assignment. The primary performance concerns include channel locking, channel unlocking and system overheads. Because of these difficulties, dynamic channel assignment and hybrid channel assignment generally perform less satisfactorily than fixed channel assignment under heavy traffic conditions.

Dynamic channel-assignment techniques have shown significant (about 30–40 %) improvements over the simple fixed channel-assignment techniques. Under low-to-moderate traffic loads, dynamic channel-assignment schemes perform far better than fixed channel-assignment schemes that are preferred under heavy traffic-load condition. Dynamic channel assignment reduces the fluctuations in the call-blocking probabilities as well as forced call terminations. However, dynamic channel-assignment schemes need a lot of effort in real time for channel assignment. Hybrid channel assignment schemes have shown to perform better than fixed channel-assignment schemes for traffic load increase up to 50 per cent.

Key Terms

- Bands
- Bandwidth
- Channel
- Downlink
- FCC
- Forward channel
- Frequency
- Full-duplex transmission
- Radio frequency spectrum
- Reverse channel
- Spectrum

Summary

Efficient resource allocation is quite vital in a very limited available radio spectrum in order to determine the optimum performance of the wireless communication systems. This chapter has given an overview of frequency management of allocated RF spectrum and channel-assignment strategies in an analog or digital cellular system. The assignment of adequate number of channels as well as interference-free channels, when in use, in each cell result in implementation complexity. Due to varying nature of traffic in each cell with time, a lot of flexibility in channel-assignment algorithms becomes a necessity. Most of the techniques for traffic-channel allocations are discussed here from the viewpoint of FDMA/TDMA- based cellular system while many issues are equally applicable to CDMA-based systems as well. Having familiarised with fundamentals of mobile communication engineering as well as principles of cellular communication in previous chapters, cellular system design trade-offs in order to ensure a balance between performance determining aspects such as radio coverage, traffic capacity and signal quality are discussed in the next chapter.

Short-Answer Type Questions with Answers

A6.1 What is meant by frequency management
Frequency management refers to dividing the allocated radio spectrum into the total number of available channels depending on the channel bandwidth of a given cellular system. It includes the functions such as designating the set-up or control and signaling channels, designating and grouping the voice or traffic channels into subsets, and numbering the channels for frequency-planning purpose.

A6.2 Distinguish between uplink and downlink.
Each communication link involves a pair of frequency channels that allows for full duplex operation. In an uplink, the signal transmission takes place from the mobile subscriber transmitter to the cell-site receiver. Uplink is also referred to as reverse channel link. In downlink, the signal transmission takes place from the cell-site transmitter to the mobile subscriber receiver. Downlink is also referred to as forward channel link. The uplink and downlink are always separated in frequency by a specific value termed as duplex separation.

A6.3 Why is it necessary to form frequency channel groups
The frequency planning of the cellular network is required to meet specific objectives which include avoiding the assignment of adjacent channels in the same cell, keeping the desired channel separation within each cell, use of combiner–tuner network device to allow a single antenna to transmit signals from multiple cells/sectors, and making cluster designing easier. The process of forming the frequency channel groups out of available channels is necessary to meet these requirements of the frequency planning in the cellular network.

A6.4 State the reason for assigning 21 set-up channels each from channel numbers 313–333 in Block A and 334–354 in Block B in S-AMPS analog cellular system.
A total of 42 set-up channels (21 channels each in Block A and Block B) are numbered in the middle of all the assigned channels, that is, channel numbers 313–333 in Block A and 334–354 in Block B. This is deliberately done in order to facilitate scanning of all these set-up channels continuously by a frequency synthesizer in the mobile receivers which can be same equipment operating either in Block A or Block B.

A6.5 What is the need of set-up channels Classify them.
Set-up channels are control channels, which are primarily used to set up the calls. All set-up channels carry signaling data information only. Each cell or sector of a cell requires at least one set-up channel. According to the nature of their usage, set-up channels can be classified as access channels and paging channels. An access channel is used by the mobile subscribers to access the system for the mobile-originating calls and paging channels are used by the base stations for mobile-terminating calls.

A6.6 How are voice channels assigned for establishment of voice calls
The assignment of certain sets of voice channels in each cell is based on causing minimum cochannel and adjacent channel interference. For mobile-originating calls, the mobile unit selects a cell-site based on its received signal-strength indicator (RSSI) value on a set-up channel. The call request from a mobile unit is received by the cell-site on the reverse set-up channel (access channel). The cell-site scans the RSSI of incoming signals and determines which sector out of three sectors of a cell has the maximum RSSI value. The MTSO then assigns a suitable voice channel from the available voice channels in that sector.

A6.7 Define channel assignment. Differentiate between static and dynamic channel assignment.
Channel assignment refers to the allocation of specific channels to cell-sites on long-term basis, and to mobile units on a short-term basis during a call. In static channel assignment, the available channels are divided into number of groups of fixed channels each cell is assigned to one of these groups of channels. In dynamic channel assignment, the central pool of channels keeps all the available channels. Channels are assigned dynamically as new requests for radio resource (for a new originating call or hand-off of existing call) arrive in the system.

A6.8 nder what circumstances is static channel assignment normally used
The static channel assignment procedure is acceptable as far as the cellular system has uniform distribution of mobile subscribers in a given service area and fully meets the capacity requirement within the available radio resources.

A6.9 List few advantages and disadvantages of fixed channel-assignment scheme.
There are certain advantages of fixed channel-assignment scheme such as fixed characteristics (power, frequency) for transceivers, reasonably acceptable performance under uniform and/or high traffic loads, and non-requirement of run-time coordination as cells independently decide their channel-assignment decisions. The disadvantages of fixed channel-assignment scheme, the problem of hot spots or localised traffic congestion which necessitates the channel borrowing from neighbouring cells.

A6.10 Assuming constant traffic density in all the cells how the channels are assigned to each cell using fixed channel-assignment scheme
The channel-assignment algorithm in fixed channel assignment (FCA) scheme is very straightforward. The total numbers of available channels are simply divided by the cluster size of the system and assign this number of channels to each cell. The FCA strategy is simple to implement, and it remains unchanged with time.

A6.11 Fixed channel-assignment scheme does not utilise the available spectrum efficiently. Justify this statement.
In practice, traffic in each cell is not constant and it changes with time due to the movement of mobile subscribers from one cell to another. This results in higher probabilities of call blocking in some cells due to non-availability of voice channels and lower probabilities of call blocking in other cells. This results in poor utilisation of the available spectrum.

A6.12 Suggest some means to equalize the poor utilisation of channels in all cells under fixed channel-assignment scheme in practical cellular systems.
The simplest solution is that the cells with higher traffic density should somehow use the free channels available in low traffic density cells. This is possible by a nonuniform assignment of channels to cells. With a nonuniform fixed channel-assignment technique, the channel-assignment algorithm takes into consideration the expected traffic and the call-blockage probability in each cell. Other means to improve the channel utilisation includes channel sharing and channel borrowing strategies with the assistance of neighbouring cells.

A6.13 List few channel-sharing algorithms in fixed channel-assignment scheme.
The most popular channel-sharing algorithms include ordered channel assignment scheme with rearrangement, channel assignment with sharing and reassignment, and temporary channel borrowing. There are relative advantages and disadvantages of different channel-sharing algorithms in terms of total channel utilisation, total carried traffic, and assignment complexity. Implementation of a particular scheme is based on the traffic behaviour and system parameters.

A6.14 Which channel-assignment approach can be effectively deployed to handle increased traffic situation
One approach to address increased traffic of either originating calls or hand-off calls in a cell is to borrow free available channels from neighbouring cells. Only those channels should be borrowed which do not cause any interference with cells within reuse distance associated with these clusters. Alternatively, a larger cell is to split it into a number of smaller cells inside a cell in an overlapped cells-based system with different channels assigned to them.

A6.15 What are different ways to implement borrowing channel-assignment scheme
One way could be to use fixed channel assignment in the beginning. When all the fixed channels are occupied, then the cell borrows channels from the neighboring cells. Another way could be to divide the available channels per cell into two groups, one group assigned to each cell permanently and the second group kept reserved for borrowing by its neighbouring cells. Another way could be borrowing with channel ordering in which priorities to all channels of each cell are assigned, and then used in a sequential priority order.

A6.16 On what basis channels are assigned in an overlapped cells-based system?

The channel assignment from the larger or smaller cells in an overlapped cells-based system is matched with the speed of the mobiles, assuming the speed will not change drastically during the call duration. For slow-moving mobiles, channels are assigned from one of the smaller cells, based on its present location, and for fast-moving mobiles, from one of the larger cell.

A6.17 List few advantages and disadvantages of dynamic channel-assignment scheme.

The advantages include better utilisation of available channel resources, better performance in low traffic conditions, its adaptability to traffic-load changes, and improved load balancing. There are certain disadvantages associated with it such as the need of exhaustive checking for occurrence of interference, to keep track of channels, efficient network channel management, channel-assignment delay, need of tunable frequency base stations and mobile units with varying power ranges.

Self-Test Quiz

S6.1 In the 800 MHz band cellular system the duplex separation is specified as

(a) 20 MHz (c) 45 MHz
(b) 25 MHz (d) 80 MHz

S6.2 The original standard non-extended spectrum in US-AMPS analog cellular system has __________ allocated bandwidth on either side of the duplex spectrum.

(a) 10 MHz (c) 25 MHz
(b) 20 MHz (d) 30 MHz

S6.3 The channel spacing in standard US-AMPS analog cellular system is

(a) 10 kHz (c) 30 kHz
(b) 25 kHz (d) 200 kHz

S6.4 In extended spectrum US-AMPS cellular standard, the uplink frequency band (Mobile Tx) is specified as

(a) 824 MHz – 849 MHz
(b) 825 MHz – 845 MHz
(c) 869 MHz – 894 MHz
(d) 870 MHz – 890 MHz

S6.5 The total number of channels available in extended spectrum US-AMPS cellular standard are

(a) 312 (c) 666
(b) 416 (d) 832

S6.6 There are total 84 voice channels available in a cellular system configured with a 7-cell omnidirectional cluster pattern. Assuming the uniform distribution of channels, the number of voice channels in a cell is

(a) 7 (c) 84
(b) 12 (d) 588

S6.7 There are a total of 168 voice channels available in a cellular system configured with a 7-cell, 3-sector cluster pattern. Assuming the uniform distribution of channels in each sector, the number of voice channels in a sector is

(a) 3 (c) 8
(b) 7 (d) 12

S6.8 There are total 120 voice channels available in a cellular system configured with a 4-cell, 6-sector cluster pattern. Assuming the uniform distribution of channels in each sector, the number of voice channels in a sector is

(a) 5 (c) 30
(b) 20 (d) 120

S6.9 In the GSM cellular system, a pair of 25 MHz band is allocated to the uplink and downlink channel to provide full duplex transmission. Each radio carrier uses a 200-kHz bandwidth. If the cluster size is four, the number of channels in each cell is

(a) 4 (c) 31
(b) 8 (d) 124

S6.10 The __________ scheme is used primarily for slow-growing systems on a long-term basis as an alternate to costly cell splitting.

(a) channel sharing
(b) channel-borrowing
(c) dynamic channel assignment
(d) overlapped cells-based channel assignment

S6.11 The solution to minimise interference in channel borrowing scheme is the use of ___.
(a) omnidirectional cells
(b) 3-sector cells
(c) 6-sector cells
(d) either (b) or (c)

S6.12 In order to verify potential interference and possible prohibition of borrowed channels from other cells, the system designer must check
(a) the reuse distance with other nearby clusters
(b) the directions of sectors of all cells not satisfying the reuse distance
(c) either (a) or (b)
(d) both (a) and (b)

S6.13 Cell sectorisation serves the same purpose as the ___ scheme.
(a) channel sharing
(b) channel-borrowing
(c) cell splitting
(d) hybrid channel assignment

S6.14 The distributed dynamic channel assignment schemes is primarily based on
(a) frequency reuse distance
(b) C/I ratio
(c) signal-to-noise ratio
(d) either (a) or (b) or (c)

S6.15 The optimal ratio between the number of fixed and dynamic channels in hybrid channel assignment mainly depends on
(a) traffic characteristics
(b) availability of channels
(c) blocking probability
(d) system overheads

Answers to Self-Test Quiz

S6.1 (c); S6.2 (b); S6.3 (c); S6.4 (b); S6.5 (d); S6.6 (b); S6.7 (c); S6.8 (a); S6.9 (c); S6.10 (b); S6.11 (d); S6.12 (d); S6.13 (b); S6.14 (d); S6.15 (a)

Review Questions

Q6.1 Distinguish between frequency management and channel assignment.

Q6.2 Mention any two significant points between set-up channels and voice channels.

Q6.3 What is the necessity of channel allocation/assignment in cellular mobile communication?

Q6.4 Compare fixed channel assignment, dynamic channel assignment, and hybrid channel assignment schemes in a cellular system.

Q6.5 Under what circumstances are channel sharing and channel borrowing schemes used in cellular systems? Explain.

Q6.6 If cell sectoring is not done in a cellular system, is it still possible to borrow channels from adjacent cells? What could be the implication in such situations?

Q6.7 What precautions have to be taken to prevent cochannel interference between cells, when borrowing channels from other cells?

Q6.8 Explain the following channel-allocation strategies in specialised cellular systems:

(a) Reuse partitioning-based channel allocation
(b) Underlay overlay arrangement
(c) Channel allocation in one-dimensional system (like a highway)
(d) Overlapped cells-based channel allocation schemes

Q6.9 Explain how different channel-allocation strategies maintain the frequency reuse distance, while borrowing channels from other cells. Illustrate with the aid of suitable diagrams and examples.

Q6.10 Explain the forcible-borrowing channel-assignment scheme.

Analytical Problems

P6.1 A cellular system is installed with 100 cell-sites employing a frequency reuse factor of 7 and 500 full duplex channels.
(a) How many numbers of channels per cell are available?
(b) Calculate the total number of channels available in the entire system.
(c) Repeat part (a) and (b) for $K = 12$.

P6.2 In the GSM cellular system operating in the 900-MHz band, 124 carrier channels are available to the uplink and downlink channel separately in order to provide full duplex communication.
(a) If the cluster size is seven, create a fixed channel-assignment chart.
(b) Each radio carrier contains eight time slots capable of supporting eight voice users. How many simultaneous communication links can be established?

P6.3 The US AMPS analog full duplex cellular system is allocated 50 MHz RF spectrum in the 800-MHz range and provides a total number of 832 duplex channels. Forty-two of those channels are designated as set-up channels and the rest are used as voice channels. The uplink frequency is exactly 45 MHz greater than the downlink frequency.
(a) What is the bandwidth for each channel?
(b) How are channels distributed between the cell-site and the mobile?
(c) Assume a cell-site transmits control information on channel number 150, operating at 874.5 MHz. Compute the transmit frequency of a mobile unit transmitting on this channel.

P6.4 In a seven-cell frequency-reuse cellular pattern with 3-sectors per cell, each cell contains three subsets of channels, i.e., $iA + iB + iC$, where i is an integer from 1 to 7.
(a) Prepare a frequency management chart for Block A channels (voice and setup) as defined for AMPS.
(b) Show that the minimum separation between three subsets of a cell is 7 channels.
(c) Show that the minimum channel separation between channels of the same sector is 21.

P6.5 In a cellular system with a 9-cell cluster, 162 voice channels are available. Show the fixed-channel assignment in each cell if omnidirectional antennas are used at the cell-site.

P6.6 A cellular system is configured with a 12-cell cluster arrangement. There are a total of 288 voice channels available. Illustrate the fixed-channel assignment in each sector if 3-sector 120 directional antennas are used at the cell-site.

P6.7 In a cellular system using omnidirectional antennas, a 7-cell cluster is employed, as shown in Fig. 6.9.

The cell marked c1 at the centre of the cluster has a lot more traffic than other cells of the cluster. Suggest the strategy to meet this requirement within the allocated channels of the same cluster or outside the cluster.

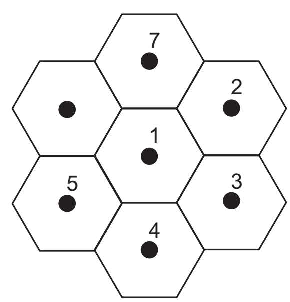

Fig. 6.9 *A 7-cell cluster omnidirectional cellular system*

P6.8 A cell is partitioned into six-sector format, s1 to s6, as shown in Fig. 6.10.

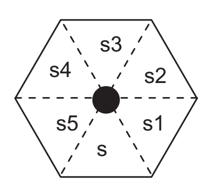

Fig. 6.10 *A 6-sector cell*

What will be the impact of channel-borrowing and cochannel interference within the sectors of a cell as well as outside the cell?

P6.9 In a cellular system with a 4-cell cluster, what would be the impact of channel borrowing if each cell deploys a 3-sector configuration.

P6.10 A cellular system is designed with 6-sector 60 directional antennas with a 4-cell cluster arrangement. What would be the impact of channel borrowing?

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7

Cellular System Design Trade-offs

As the demand for wireless communication service increases, the number of channels assigned to a cell eventually becomes insufficient to support the required number of subscribers to establish simultaneous communication links. There are several cellular-design techniques which help to achieve enhanced system capacity within the allocated limited frequency spectrum at an acceptable signal quality. This chapter covers numerous methods and system parameters to achieve these objectives. The emphasis here is to understand the cellular-system design trade-offs among various techniques, ranging from cell splitting for varying traffic densities; modulation and speech-coding techniques for spectrum efficiency; equalisation, diversity, and channel-coding techniques to combat errors, and hand-off mechanisms, leading to successful planning, design and implementation of cellular mobile communication systems.

7.1 SYSTEM PARAMETERS TO INCREASE CELL COVERAGE

The following approaches are used in a cellular system to increase the radio coverage within a cell.

– Increasing the transmitted power at the cell-site
– Increasing cell-site antenna height
– Using a high-gain omnidirectional antenna at the cell-site
– Using directional antennas at the cell-site
– Proper selection of cell-site locations
– Engineering the antenna patterns at the cell-site
– Using a diversity receiver
– Using a low-noise receiver
– Lowering the threshold level of received signals
– Using wireless repeaters or/and signal enhancers

From the knowledge of propagation models in a mobile operating environment, the received signal power at the mobile subscriber, P_r is given by

$$P_r \propto P_t G_t h_t r^{-4} \tag{7.1}$$

where P_t is the cell-site transmitter power, G_t is the cell-site transmitter antenna gain, h_t is the cell-site transmitter antenna height, and r is the distance between cell-site transmitter and mobile receiver, which signifies the coverage radius. The path-loss exponent γ in a mobile radio environment is taken as 4. Keeping all other factors unchanged, the increase of transmitter power will result into increase in coverage area.

EXAMPLE 7.1 Increase in cell coverage by increasing transmitted power

The cell-site transmitted power increases by 3 dB. For the same minimum acceptable received signal power and all other factors remaining unchanged, prove that the coverage area increases by 1.4 times. Assume mobile radio operating environment conditions.

Solution

Step 1. *Let the initial cell-site transmitted power be P_{t_1} and the minimum acceptable received signal power be P_{r_1} at a cell radius of r_1. Then,*

$$P_{r_1} \propto P_{t_1} r_1^{-4} \tag{7.2}$$

Step 2. *Let the changed cell-site transmitted power be P_{t_2} and the corresponding minimum acceptable received signal power be P_{r_2} and the cell radius be r_2. Then,*

$$P_{r_2} \propto P_{t_2} r_2^{-4} \tag{7.3}$$

Step 3. *It is given that the minimum acceptable received signal power remains unchanged even after increasing the transmitted power, that is,*

$$P_{r_1} = P_{r_2}$$

Step 4. *Therefore, equating Eq. (7.2) and Eq. (7.3),*

$$P_{t_1} r_1^{-4} = P_{t_2} r_2^{-4}$$

Or, $$P_{t_1} r_2^{4} = P_{t_2} r_1^{4}$$

Or, $$r_2^{4} = (P_{t_2} / P_{t_1}) r_1^{4}$$

Or, $$r_2 = (P_{t_2} / P_{t_1})^{1/4} r_1 \tag{7.4}$$

Step 5. *It is given that the cell-site transmitted power increases by 3 dB, which means it is doubled. That is,*

$$P_{t_2} = 2\, P_{t_1}$$

Or, $$(P_{t_2} / P_{t_1}) = 2$$

Step 6. *Substituting it in Eq. (7.4), we get*

$$r_2 = (2)^{1/4} r_1$$

Or, $$r_2 = 1.19\, r_1 \tag{7.5}$$

Step 7. *Let the initial coverage area be A_1 and the new coverage area be A_2. As we know that the coverage area (circular or hexagonal) is directly proportional to square of the radius,*

$$A_2 = (r_2 / r_1)^2 A_1$$

Using Eq. (7.5), $$A_2 = (1.19)^2 A_1$$

Hence, $$A_2 = 1.4\, A_1 \tag{7.6}$$

Comments on the result It is proved that when the transmitted power is increased by 3 dB (or doubled), the coverage area increases by 1.4 times for the same minimum acceptable received signal power and all other factors remaining unchanged.

In a flat operating terrain, doubling the cell-site antenna height results into an increase in gain by 6 dB (or four times), that is, the 6-dB/octave rule is applicable. In a hilly terrain contour, the increase in gain may be more or less than 6 dB for doubling the cell-site antenna height, depending on the location of the mobile unit. In such situations, an effective antenna height should be used to compute the increase in coverage area.

Using a high-gain omnidirectional antenna or a directional antenna at the cell-site, the similar increase in coverage area would be obtained as seen in case of increasing transmitted power. Similarly, use of multiple directional antenna patterns offered by adaptive antenna arrays and smart antennas at the cell-site can also result into increase in coverage area.

Facts to Know!

With a given transmitted power and actual antenna height at the cell-site, coverage area can be increased by proper selection of cell-site location. In principle, a high location around the planned site location should always be selected.

A diversity receiver is very useful in reducing the multipath fading, and thereby increase in received signal level at the mobile subscriber unit. For the same minimum acceptable received signal level, effective radio coverage area is increased. In the same way, use of a low-noise receiver can result into increase in radio coverage area because it can receive the minimum acceptable received signal at the farther distance. When the minimum acceptable threshold signal level is reduced at the receiver, the radius of the cell increases, thereby increasing the radio coverage area. The extent of increase in coverage area for a specified reduction in the threshold received signal level is illustrated in Example 7.2.

Example 7.2 | Increase in coverage by reducing threshold received level

Let the minimum acceptable threshold signal level be reduced at the receiver by 6 dB. For the same cell-site transmitted power and all other factors remaining unchanged, prove that the coverage area is doubled. Assume mobile radio operating environment conditions.

Solution

Step 1. *Let the initial cell-site transmitted power be P_{t_1} and the minimum acceptable received signal power be P_{r_1} at a cell radius of r_1. Then,*

$$P_{r_1} \propto P_{t_1} r_1^{-4}$$

Or,
$$P_{t_1} \propto P_{r_1} r_1^{4} \tag{7.7}$$

Step 2. *Let the changed cell-site transmitted power be P_{t_2} and the corresponding minimum acceptable received signal power be P_{r_2} and the cell radius be r_2. Then,*

$$P_{r_2} \propto P_{t_2} r_2^{-4}$$

Or,
$$P_{t_2} \propto P_{r_2} r_2^{4} \tag{7.8}$$

Step 3. *It is specified that the cell-site transmitted power remains unchanged though the minimum acceptable received signal power is reduced, that is,*

$$P_{t_1} = P_{t_2}$$

Step 4. *Therefore, equating Eq. (7.7) and Eq. (7.8),*

$$P_{r_1} r_1^{4} = P_{r_2} r_2^{4}$$

Or,
$$r_2^{4} = (P_{r_1} / P_{r_2})\, r_1^{4}$$

Or,
$$r_2 = (P_{r_1} / P_{r_2})^{1/4} r_1 \tag{7.9}$$

Step 5. *It is given that the minimum acceptable received signal power is reduced by 6 dB that means it is reduced by four times (every 3-dB reduction corresponds to half of the original signal power). That is,*

$$P_{r_2} = (1/4)\, P_{r_1}$$

Or,
$$P_{r_1} = 4\, P_{r_2}$$

Or,
$$(P_{r_1} / P_{r_2}) = 4$$

Step 6. *Substituting it in Eq. (7.9), we get*

$$r_2 = (4)^{1/4} r_1$$

Or, $$r_2 = 1.414\, r_1$$

Or, $$r_2 / r_1 = 1.414 \tag{7.10}$$

Step 7. *Let the initial coverage area be A_1 and the new coverage area be A_2. As we know that the coverage area (circular or hexagonal) is directly proportional to square of the radius,*

$$A_2 = (r_2 / r_1)^2 A_1$$

Step 8. *Using Eq. (7.10), we get $A_2 = (1.414)^2 A_1$*

Hence, $$_2 = 2 \quad _1$$

Comments on the results Hence it is proved that when the minimum acceptable received signal power is reduced by 6 dB (or four times), the cell coverage area is doubled for the same cell-site transmitted power and all other factors remaining unchanged.

Wireless repeaters or signal enhancers are often used to extend the coverage area. Repeaters are usually bi-directional in nature, and simultaneously send signals to and receive signals from a serving cell-site. Wireless repeaters may be installed anywhere and are capable of repeating the complete allocated cellular frequency band. Upon receiving signals from a cell-site forward link, the repeater amplifies and reradiates the cell-site signals to the specific extended coverage area.

7.2 COVERAGE HOLE FILLERS AND LEAKY FEEDERS

Coverage hole is an area within the radio coverage footprint of a wireless communication system in which the received RF signal level is below the specified threshold value. Coverage holes, also called *weak received signal spots*, are usually caused by physical obstructions such as buildings, hills, dense foliage as well as hard-to-reach areas such as within buildings (indoor), or in valleys or in tunnels. Because the earth is not flat, many coverage holes are created during transmission of radio signals.

The radio coverage is sometimes blocked in the outdoor wireless network applications which contain high buildings, hills and tunnels, and thus shadow regions are created. It is most desirable to fill in these coverage holes. The received signals are required to be extended by simple means into these coverage hole areas. Deploying a new cell-site in this area could be one possible solution but it is not only expensive, but also requires new channel assignment or rearranging the frequency plan. So it may not be economically justifiable from the revenue point of view from that area which may still be strategically important.

Among the various techniques available for filling the coverage holes such as wireless enhancers or repeaters (wideband and channelised), passive reflectors, diversity receivers, and cophase combiners (feedforward and feedback), wireless repeaters can provide the simplest and cost-effective solution to reradiate the amplified signal so as to reach the coverage hole areas. Unfortunately, the received noise and interference is also reradiated by the repeater on both the forward and reverse link. Therefore, care must be taken to properly place the repeaters, and to adjust the various forward and reverse-link amplifier levels and antenna patterns. In practice, particularly in tunnels or high-rise buildings, directional antennas or distributed antenna systems are connected to the inputs or outputs of repeaters for localised weak-spot coverage.

Facts to Know!

In the deployment of any wireless communication network, the radio coverage provided by any given cell-site does not cover 100% locations within the cell boundary, determined by the minimum acceptable received signal levels.

Facts to Know!

The repeater does not add capacity to the system. It simply serves to reradiate the received signal into specific coverage-hole locations.

By modifying the coverage of a serving cell by using wireless repeaters, a service provider is able to dedicate a certain amount of the cell-site's traffic for the areas covered by the repeater. The two physical considerations for a successful repeater deployment are isolation and line-of-sight conditions. There must be sufficient isolation between the donor antenna and the coverage antenna to prevent feedback oscillation for on-frequency repeaters where the transmitted frequency is the same as the received frequency. In order to meet this requirement, the two antennas — the donor antenna and the coverage antenna — must be physically separated from each other. This physical separation can be realised if

- there is a tower of sufficient height to separate the antennas vertically while pointed in opposite directions—one towards the actual cell-site transmitter and another towards the coverage hole region,
- the antennas are mounted on separate masts with sufficient horizontal separation and pointed in opposite directions as stated above, or
- there is a physical structure such as a building that can provide the needed physical isolation when the antennas can be mounted on opposite sides of the building.

At the repeater location, there must be line-of-sight condition to both the coverage hole area and the desired donor cell-site. It is desirable that the downlink signal from the repeater must be strong enough to be received by the mobile subscribers located in the coverage hole region while the receiver sensitivity of repeater plus that of the cell-site must be sufficient enough to process the signal received from the mobile subscribers.

Figure 7.1 illustrates deployment of the repeater with two different directional antennas mounted on the same tower, one in the direction of the main cell-site and another in the direction of a mobile operating in a coverage hole region.

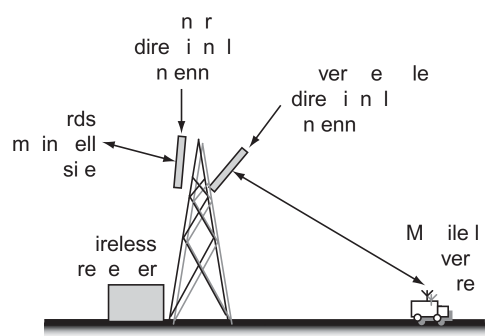

Fig. 7.1 *Illustration for deployment of wireless repeater*

One of the main concerns is that there is overlap between the repeated signal and the primary cell-site signal. The repeated signal has significant added delay. In some cases, it is important to limit the coverage antenna pattern to the area needed and minimise the overlap. If the undesired signal received by the donor directional antenna is transmitted back to the cell-site, cochannel and adjacent-channel interference may occur. For shadow areas such as behind the hills or inside tunnels, this is generally not a problem. Otherwise, an alternative way is to use a high-gain donor antenna to bring in the signal from a neighbour cell. In this case, the mobile subscribers located in coverage-hole regions would hand-off to the neighbour cell in the repeater coverage area and there would be no interference.

Instead of using a wideband enhancer, a channelised enhancer can be effectively deployed which would amplify only the selected channels. Geographic terrain contour should be considered in proper installation of a channelised enhancer. The separation between two antennas at the enhancer is very critical. If this separation is inadequate, the signal from the coverage antenna can be received by the donor antenna or vice versa. This may result in jamming of the system instead of filling the coverage hole. Likewise, the distance between the enhancer location and the serving cell-site should be as minimum as possible to avoid spread of radiations into a large area in the vicinity of the serving cell-site and beyond.

Leaky Feeders In some areas such as in tunnels, underground garages, coal mines or within a cell of less than 1-km radius, leaky-feeder techniques provide adequate coverage with minimum interference. The most popular types of leaky feeders are

- Leaky waveguide or fast-wave antenna
- Leaky coaxial feeder cable

In leaky waveguides, only the fractional energy will be leaking constantly through the opening slots (apertures) of the waveguide structure supporting $\lambda g > \lambda c$, where λg is the wavelength of the transmitted waveguide and λc is the wavelength of the transmitted signal in free space. The *leakage rate* is a function of position of the slot in the waveguide. The radiation pattern of a leaky waveguide can serve a larger area along the waveguide. These are generally used above 3 GHz.

Leaky cables are easily implemented in the tunnels because their energy is confined within the tunnel. Leaky coaxial feeder cables are used at frequencies below 1 GHz because the cable loss is 6 dB/100 m only. The RF powers cannot exceed a maximum of 500 mW in order to prevent any inflammatory sparks for safety considerations. The leaky feeder is characterised by transmission and coupling losses. For proper operation of a leaky coaxial feeder cable, use of high-coupling loss cables near the transmitter is recommended as little energy will leak out and thus have low-transmission loss.

Facts to Know!

Low temperature and snow accumulation around slots causes an increase in transmission loss. Periodic spacing of slots along the leaky coaxial feeder cable causes the intensive radiation pointing to a specific direction.

7.3 SYSTEM PARAMETERS TO REDUCE INTERFERENCE

In a wireless communication environment, no matter whether the signal is transmitted outdoors or indoors, the phenomenon of multipath interference is unavoidable. In a cellular system network, the frequency reuse is exploited extensively in order to increase system capacity by efficient use of allocated RF spectrum, thereby causing significant system interference. Even though interfering signals are often generated within the cellular system, they are difficult to control in practice due to random propagation effects. Moreover, the transmitters from competing cellular service operators are often a significant source of out-of-band interference that is even more difficult to control, since other cellular service operators often locate their cell-sites in close proximity to one another in order to provide comparable cell coverage to mobile subscribers.

Facts to Know!

The two major types of system-generated cellular interference are cochannel interference and adjacent-channel interference

A mobile subscriber unit is capable of receiving a downlink signal from each of a number of cell-sites as well as transmitting an uplink signal to a number of cell sites through a wireless channel. The combined effect of multipath interference and cochannel interference becomes a challenging task to maintain the desired performance of cellular systems. Thus, interference is the major limiting factor in achieving the desired performance of cellular radio systems.

Sources of interference may include another mobile active in the same cell, a call in progress in an adjacent cell, other cell-sites operating in the same frequency band, or any non-cellular wireless communication system, which inadvertently leaks energy into the cellular frequency band. Interference on voice channels causes cross talk, and interference on control channels leads to missed and blocked calls due to errors in the signaling. Interference is more severe in urban areas, due to the greater RF noise floor and availability of a large number of active cell-sites and mobiles. Interference has been recognised as a major bottleneck in increasing system capacity and is often responsible for dropped calls.

In most situations, the methods adopted for increasing the coverage area also causes additional interference if cochannel and adjacent channels are used in the system. So there is always a trade off between the methods used for increasing coverage area and reducing interference so as to ensure the overall system quality. Therefore, it is desirable to reduce interference in the cellular network. The following approaches are used in a cellular system to reduce the interference.

– A good frequency-management plan
– An intelligent real-time channel assignment scheme

- Assignment of a proper voice channel to a particular mobile unit
- Design of an antenna pattern on the basis of a desired direction
- Tilting antenna patterns by electronic or mechanical downtilting techniques
- Reducing the antenna height
- Choosing the cell-site location properly
- Reducing the transmitted power at the cell-site

requency management, or *radio resource management*, is the system-level control of cochannel interference and other radio-transmission characteristics in wireless communication systems. Frequency management involves strategies and algorithms for controlling system parameters such as radiated power, channel assignment, handover criteria, modulation scheme, error-coding scheme, etc. The objective of a good frequency management scheme should be to utilise the limited RF spectrum resources and wireless network infrastructure as efficiently as possible without exceeding the acceptable limits of interference.

Intelligent frequency planning is needed in order to avoid problems of cochannel and adjacent-channel interference in a cellular system. The channels may be assigned dynamically in such a way so as not to cause any interference to on-going communications. Depending on the active channels at any time, some free channels may be noisy, some free channels may be quiet, and some free channels may be vulnerable to channel interference. These factors should be considered in the assigment of voice channels to a particular mobile subscriber unit among a set of available channels.

> **Facts to Know!**
>
> The type and installation of a cell-site antenna system plays a critical role in determining the impact of system interference. Depending on the traffic demand, a strong signal may be needed in some directions and no signal may be needed in some other directions.

The design of an antenna pattern—omnidirectional or directional—including downward tilting will enable to confine the radiated signal energy within a small area, thereby reducing interference. Proper selection of cell-site location as well as optimum antenna height depending upon the terrain conditions along with reducing transmitted power can be more effective in reducing interference while trading off with the coverage area. In practical cellular communication systems, the power levels transmitted by every mobile subscriber unit are under constant control by the serving cell-site. This is done to ensure that each mobile subscriber transmits the minimum power necessary to maintain a good quality link on the reverse channel as well as to reduce the interference.

7.4 METHODS TO INCREASE TRAFFIC CAPACITY

Traffic capacity, or user capacity, can be simply defined as the maximum number of simultaneous users that can be supported by a system for a given performance requirement. The capacity of a cellular system is directly proportional to the number of times a cluster is replicated in a designated service area. As the demand for cellular service increases, the number of channels assigned to a cell eventually becomes insufficient to serve the required number of subscribers. At this point, various cellular-design methods are needed to provide more number of channels per unit coverage area to expand the capacity of cellular systems. The following methods may be considered to increase the traffic capacity of a cellular system.

- Enhanced frequency spectrum for new subscribers
- Proper channel-assignment strategies
- Mutli-access and modulation schemes
- Use of smart antennas
- Cellular hierarchy

Obtaining enhanced frequency spectrum for new subscribers is a very simple but too expensive method to increase traffic capacity. For example, an additional spectrum allocation of 10 MHz means an increase of 166 voice channels in US AMPS system.

Changing the channel-assignment strategies can result in increase in capacity of a cellular network. In practice, the distribution of the subscribers in the area is not uniform over all cells at all time. So, instead of distributing available channels equally among all cells, it is possible to use a nonuniform distribution of channels among different cells according to their respective traffic requirements. The user capacity of each cell is dynamically changing by the geography of the service area and with time. The dynamic channel assignment to different cells also reduces the probability of call blockage which results in accepting a higher number of mobile subscribers over the coverage area. This can be considered equivalent to expansion of the network capacity through availabilty of additional channels.

Facts to Know!

By using smart antennas at cell-sites, mobile subscribers in the same cells can use the same physical communication channel as long as they are not located in the same angular region with respect to a cell-site. Such a multi-access scheme, referred to as space division multiple access, can be achieved by the cell-site directing a narrow beam toward a mobile subscriber unit communicating with it. The interference between cochannel cells is also greatly reduced because the antenna radiation patterns are extremely narrow.

Another effective method to increase the capacity is to change the mutli-access and modulation scheme. Use of digital cellular technology such as TDMA (Time Division Multiple Access) and CDMA (Code Division Multiple Access) air interface with digital modulation schemes results into significant increase in system capacity as compared to analog cellular technology using FDMA (Frequency Division Multiple Access) with FM modulation. However, this changeover requires the installation of new infrastructure by service providers as well as use of a new mobile terminal by the subscribers.

Another method to increase the capacity of the cellular network is changing the cellular architecture and hierarchy such as *cell sectoring* using directional antennas, *cell splitting* (use of small cell size), *microcell zone techniques*, and using multiple reuse factors called *reuse partitioning*. These techniques change the size and shape of the coverage of the cells by adding new cell-sites or modifying the type and installation of cell-site antennas to increase the overall capacity.

7.5 CELL SPLITTING

The prime objective of implementing a cellular mobile system is to enhance the system capacity along with improvement in the spectrum efficiency. Frequency reuse and cell splitting are two main concepts in cellular systems required to achieve this objective. Cell splitting is the process of dividing a larger congested cell into smaller cells, each with its own cell-site with a corresponding reduction in transmitter power and antenna height. This is usually done to make more voice channels available to accommodate traffic growth in the area covered by the original cell. When the traffic in an area increases, larger cells are split into smaller cells so that frequency can be reused more frequently. By defining new smaller cells which have a smaller radius than the original larger cells and by installing these smaller cells (called microcells) between the existing cells, the system user capacity increases due to availability of additional number of channels per unit service area as well as the number of times that frequency channels are reused. In other words, the increased number of cells would increase the number of channels as well as number of clusters over the coverage area, which in turn would increase the overall system capacity.

It has been seen earlier how the concept of frequency reuse leads to a cellular architecture that can allow for almost limitless expansion in the geographic area and the number of mobile subscribers that the system can serve. In configuring a cellular layout, there are two most important key parameters—the cell radius R and the cluster size K. Although a cellular system can be expanded in a geographic coverage area simply by adding cells at the periphery, yet it is worth considering as to how a system can expand in meeting the requirement of increasing subscriber density.

The cell radius governs both the geographic radio coverage area served by a cell-site located at its centre and also, for a given subscriber density, the number of subscribers that the cell must service. Simple economic considerations suggest that the cell size should be as large as possible. Since installation and commissioning of every cell requires an investment in a antenna tower, land on which the tower is placed, and radio transmission equipment, a large cell size minimises the average infrastructure cost per subscriber. The cell size is ultimately determined by the received signal quality requirement that an acceptable signal-to-noise ratio be maintained over the coverage area. A number of system parameters such as transmitter power, cell-site antenna height, receiver noise figure, and receiver sensitivity are involved in determining the signal-to-noise ratio. Transmitter power is particularly limited in the reverse direction, as the mobile subscriber units have low transmitted power, are battery powered, and small in size.

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Cell splitting is an attractive feature of the cellular concept. When the call traffic in a particular cell could no longer support a reasonable grade of service with the existing allocation of channels in that cell, that cell would be subdivided into a number of smaller cells — with lower transmitter power and new (on a smaller scale) frequency reuse pattern — fitting within the area of the former cell.

Cell splitting allows an orderly growth of the cellular system as per the requirement of varying traffic density. Cell splitting increases the number of cell sites in order to increase the system capacity. Thus, cell splitting allows a system to grow by replacing large cells with smaller cells, with changing the frequency reuse plan or channel-allocation scheme required to maintain the minimum cochannel reuse ratio between cochannel cells.

7.5.1 Techniques of Cell Splitting

There are two techniques of cell splitting — permanent cell splitting and dynamic cell splitting. In *permanent cell splitting*, the installation of every new split cell has to be planned in advance. The transmitter power, the number of channels per cell, the assigned frequencies, selection of the cell-site, and the traffic load should all be considered. The channel assignment scheme should follow the rule based on frequency-reuse ratio with the transmitter power adjusted accordingly. Selecting a proper site location for a small cell is not easy. The cell-site antenna can be mounted on a monopole or erected by a mastless antenna arrangement. When ready with the plan of split cells, the actual changeover of service with split cells should take place during the lowest traffic period, usually at a midnight or at a weekend, to avoid no-service conditions or a large number of call drops.

The *dynamic cell-splitting scheme* is based on utilising the allocated spectrum efficiency in real time. The algorithm for dynamically splitting the cells is a tedious job and it should be implemented gradually to prevent dropped calls at heavy traffic hours. The size of the splitting cells is also dependent on the radio aspect and the capacity of the switching processor. The radio aspects include how well the coverage pattern can be controlled and how accurately mobile locations would be known. A high capacity of the switching processor is needed to handle more hand-offs due to smaller cells.

7.5.2 Effects of Cell Splitting

Cell splitting involves the changes in the following aspects of cellular architecture.

- Reduction in the coverage area of a split cell
- Reducion in the cell-site transmitter power of a split cell
- Increase in traffic load after cell splitting
- Changing frequency reuse plan
- Changing the channel assignment

When the traffic density starts to build up and the number of assigned channels in each cell cannot provide enough mobile calls simultaneously, the original larger cell can be split into smaller cells. Usually, the radius of a new split cell is one-half of the original larger cell, that is,

New cell radius = (original cell radius) / 2

If the radius of the original cell is designated as R_0 and the radius of the split cell is designated as R_1 then

$$R_1 = R_0 / 2$$

There are two ways of cell splitting:

- The one in which the original cell-site is not used and is completed replaced with new split cells (Fig. 7.2 depicts this situation with circular cell areas).
- The second way of cell splitting is in which the original cell is also used along with new cells, as shown in Fig. 7.3.

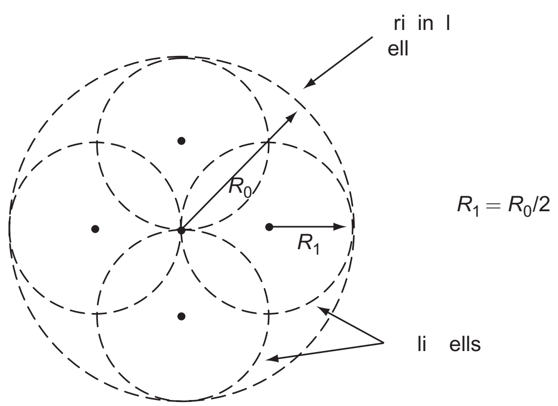

Fig. 7.2 | *Cell splitting without using original cell*

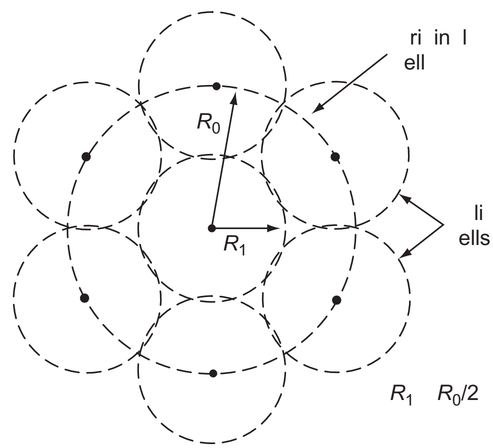

Fig. 7.3 | *Cell splitting using original cell*

EXAMPLE 7.3 | **Coverage area after cell splitting**

Let the radius of split cells be 50% of the radius of the original cell (before splitting). Show that the coverage area of a split cell is one-fourth the coverage area of the original cell. Comment on the results obtained.

Solution

Step 1. *Let the radius of the original cell* $= R_0$

The reduction in the radius of split cell is 50% of the radius of the original cell.

Then, the radius of the split cell, $R_1 = R_0/2$

Step 2. *Assuming the regular hexagonal cells,*

Area of a regular hexagonal original cell, $A_0 = (3\sqrt{3} / 2) \times R_0^2$

Step 3. *Area of a regular hexagonal split cell,* $A_1 = (3\sqrt{3}/2) \times (R_1)^2$

Substituting $R_1 = R_0/2$, $A_1 = (3\sqrt{3} / 2) \times (R_0/2)^2$

Or, $A_1 = (3\sqrt{3} / 2) \times R_0^2 \ / 4$

Hence, Area of a split cell, $_1 = \ _0 / 4$

Comment on the results It is seen that the coverage area of the split smaller cell is one-fourth of the coverage area of the larger original cell. It means that in order to cover the original service area (being served by original larger cell with radius R_0) with split cells (having radius $R_0/2$), four times as many split cells would be required. The increased number of cells would increase the number of clusters over the coverage region, which in turn would increase the number of channels, and thus capacity, in the coverage area. However, the increase in capacity due to split cells is achieved at the cost of installing additional infrastructure.

EXAMPLE 7.4 Cell-site transmitter power after cell splitting

For an identical received power at the boundaries of original larger cell with radius R_0 and the new split cell with radius $R_0/2$, prove that the cell-site transmitter power of the split cell must be 12 dB less than the cell-site transmitter power of the original larger cell. Assume path-loss exponent as 4 in a typical mobile environment. Comment on the results obtained.

Solution

For the split cells of smaller size, the transmit power of these cells must be reduced for an identical received power at the boundaries of the cells. The transmit power of the new split cells with radius half that of the original larger cells can be computed by examining the received signal power, P_r at the cell boundaries of the original and split cells respectively, and then setting them equal to each other. This is necessary to ensure that the frequency reuse plan for the new split cells behaves exactly as for the original cells in order to maintain the cochannel interference levels.

Step 1. *Let P_{t_0} be the cell-site transmitter power of the larger original cell with radius R_o,*
Then, at the cell boundary of the original cell,

$$P_r \propto P_{t_0} R_0^{-4} \tag{7.11}$$

Step 2. *Let P_{t_1} be the cell-site transmitter power of the smaller split cell with radius $R_0/2$,*
Then, at the cell boundary of the split cell,

$$P_r \propto P_{t_1} (R_0/2)^{-4} \tag{7.12}$$

Step 3. *Equating Eq. (7.11) and Eq. (7.12), we get*

$$P_{t_1} (R_0/2)^{-4} = P_{t_0} R_0^{-4}$$

Or, $$P_{t_1} (1 / 2^{-4}) = P_{t_0}$$

Or, $$P_{t_1} = P_{t_0} /16$$

Step 4. *Expressing it in logarithmic form, we get*

$$P_{t_1} \text{(dBm)} = P_{t_0} \text{(dBm)} - 10 \log (16)$$

Or, $$P_{t_1} \text{(dBm)} = P_{t_0} \text{(dBm)} - 12 \text{ (dB)} \tag{7.13}$$

Comment on the results It means that the cell-site transmit power of a split cell with radius $R_0/2$ must be reduced by 12 dB than the cell-site transmitter power of original cell with radius R_0 in order to fill in the original coverage area with split cells, while keeping the signal-to-interference ratio requirement same.

In case an original larger cell is split repeatedly 'n' number of times, and every time the radius of the split cell is one-half of its immediate previous cell, that is,

$$\text{the radius of } n^{\text{th}} \text{ split cell, } R_n = R_0 /n$$

where R_0 is the radius of the original cell, and R_n is the radius of the n^{th} split cell

Then, cell-site transmitter power of the n^{th} split cell can be given by

$$_t \text{ (dBm)} = \ _{t0} \text{(dBm)} - 12 \text{ (n) dB} \tag{7.14}$$

EXAMPLE 7.5 Cell-site transmitter power after multiple cell splitting

For an identical received power at the boundaries of the original larger macrocell with radius R_0 and the new split minicell with radius $R_0/2$ and further split microcell with radius $R_0/4$, prove that the cell-site transmitter power of the split microcell must be 24 dB less than the cell-site transmitter power of the original larger macrocell. Assume path loss exponent as 4 in a typical mobile environment.

Solution

Radius of the original larger macrocell $= R_0$ (given)

Radius of the new split mini cell $= R_0 / 2$ (given)

Radius of the split microcell $= R_0 / 4$ (given)

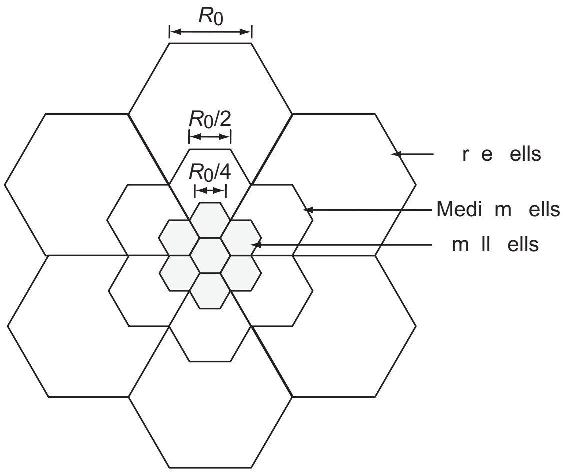

Fig. 7.4 *Repetitive cell splitting*

Step 1. *Since the cell splitting is carried out from original radius R_0 to $R_0/2$ and then once again to $R_0/4$, the cell splitting is carried out two times, which means n = 2.*

Step 2. *Figure 7.4 illustrates the cell splitting from radius R_0 to $R_0/2$ and then to $R_0/4$.*

Step 3. *The reduction in transmitted power of smallest microcell after two times splitting is given by the Eq. (7.14) as*

$$P_{t_2}\,(\text{dBm}) = P_{t_0}\,(\text{dBm}) - 12 \times 2 \text{ dB}$$

$$\text{Or,}\quad P_{t_2}\,(\text{dBm}) = P_{t_0}\,(\text{dBm}) - 24 \text{ dB}$$

Hence the reduction in transmitted power of the split cell should be by 24 dB.

When cell splitting occurs, the value of the frequency-reuse factor or cochannel interference reduction factor $q = D/R$ is always held constant since both D and R are split by the same factor.

EXAMPLE 7.6 Increase in system capacity after cell splitting

The radius of the split cell is one-half of that of the original cell (before splitting). Show that the overall system capacity increases by four times.

Solution

Radius of the original larger macrocell = R_0 (given)

Radius of the new split minicell = $R_0 / 2$ (given)

Step 1. *The area of a cell is proportional to the square of the radius of the cell.*

Step 2. *Therefore, when the original cell with radius R_0 is split into smaller cells, each with radius $R_0/2$, the coverage area of the split cell is one-fourth of the coverage area of the original cell.*

Step 3. *It means that in order to cover the original service area with these split cells, four times as many split cells would be required, while maintaining the same reuse factor or cochannel interference ratio as well as the minimum acceptable received signal level at the boundary of the split cell as that of the original cells.*

Step 4. *Assuming that the number of channels assigned per cell remains same in split cells as that of original cell, the frequency reuse planning is redone.*

Step 5. *This will result into proportionate increase in the number of clusters with split cells to cover the entire region as designated with the original cells.*

Step 6. *Thus, the total number of available channels with split-cell configuration increases by four times.*

Step 7. *Therefore, the overall system capacity also increases by four times in the coverage area.*

Hence, new system capacity = 4 × (system capacity with original cells)

If each split cell can again be split into four sub-split cells with further radius reduction by one-half then the traffic load would increase by 16 times. As the cell splitting continues every time with reduction in radius of a split cell being one-half of that of its immediate previous cell, the general formula for increase in overall system capacity with cell splitting can be expressed as

$$\text{New system capacity} = 4^n \times (\text{system capacity of original cell}) \tag{7.15}$$

where 'n' is the number of times of uniform cell splitting.

For example, if $n = 4$, it means an original cell has been split four times. The new capacity will be

$$\text{New system capacity} = 4^4 \times (\text{system capacity of original cell})$$

Or, $$\text{New system capacity} = 256 \times (\text{system capacity of original cell})$$

EXAMPLE 7.7 Channel capacity after cell splitting

Determine

(a) The channel capacity for a cellular system service area comprised of seven macrocells with 16 channels per macrocell

(b) Channel capacity if each macrocell is split into four minicells

(c) Channel capacity if each minicell is further split into four microcells

Solution

(*a*) *To determine the channel capacity for macrocell configuration*

Number of macrocells per system = 7 (given)

Number of channels per macrocell = 16 (given)

Number of channels per macrocell system = 7 × 16

Hence, the channel capacity of macrocell system = 112 channels

(*b*) *To determine the channel capacity for minicell configuration*

Number of macrocells per system = 7 (given)

Number of channels per macrocell = 16 (given)

Number of minicells per macrocell = 4 (given)

Number of channels per minicell system = 7 × 16 × 4

Hence, the channel capacity of minicell system = 448 channels

Alternately,

$$\text{New channel capacity} = 4 \times (\text{channel capacity with original cells})$$

Hence, $$\text{New channel capacity} = 4 \times 112 \text{ channels} = 448 \text{ channels}$$

(*c*) *To determine the channel capacity for microcell configuration*

Number of macrocells per system = 7 (given)

Number of channels per macrocell = 16 (given)

Number of minicells per macrocell = 4 (given)

Number of microcells per minicell = 4 (given)

Number of channels per minicell system = 7 × 16 × 4 × 4

Hence, the channel capacity of minicell system = 1792 channels

Alternately,

New channel capacity = 4^n × (channel capacity of original cell)

where 'n' is the number of times of uniform cell splitting.

Here a macrocell is split into 4 minicells and each minicell is further split into 4 microcells. Therefore, $n = 2$

New channel capacity = 4^2 × (channel capacity of original cell)

Hence, New channel capacity = 16 × 112 channels = 1792 channels

Example 7.8 Increase in system capacity with cell splitting

Consider the cellular system shown in Fig. 7.5, in which the original cells with a radius of 2 km are split into smaller cells, each with a radius of 1 km.

Let each cell-site be assigned 120 channels regardless of the cell size. How many times will the number of channels contained in a 6 × 6 km^2 area (shown with dotted lines in Fig. 7.5) centered around small cell 'S' be increased with cell splitting as compared to without cell splitting?

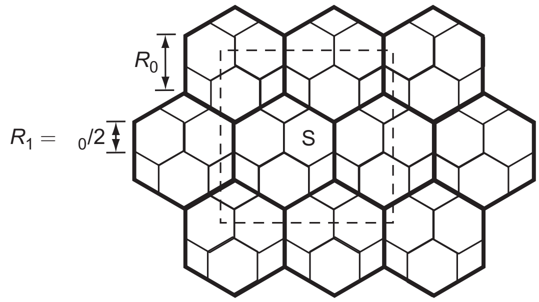

Fig. 7.5 *An example of cell splitting*

Solution

The number of channels assigned in a large cell = 120 (given)

Step 1. *Since the number of channels assigned in a split cell is the same as that in a large cell, therefore,*

The number of channels assigned in a split cell = 120

Step 2. *Coverage area centered around the small cell 'S' = 6 × 6 km^2 (given)*

From the figure, it is seen that to cover the given area, it is needed to cover 3 km each to the left, right, top and bottom of the cell-site.

Step 3. *As evident from the given figure, it can be easily observed that the 6 × 6 km^2 centered around the small cell 'S' contains more number of split cells than that of large cells.*

Step 4. *However, because of difference in edges of a square area and a regular hexagonal area, the number of either types of cells contained within the given square area cannot be accurate, it can only be an estimate. A reasonable approximation from visual observation is that there are nearly 4 large cells within this area.*

Number of large cells within the given area = 4
Number of channels per large cell = 120
Number of channels without cell splitting = 4 × 120 = 480 channels

Step 5. *Radius of the large cell, $R_0 = 2$ km (given)*

Radius of the split cell, $R_1 = 1$ km (given)

Number of split cells within the square = $(R_0 / R_1)^2$ × number of large cells

Or, Number of split cells within the square = $(2 / 1)^2 \times 4 = 16$ small cells

Step 6. *With a finite square area under consideration, it is necessary to take care of edge effect. So the number of small cells contained within the given square area would be slightly less than 16. A reasonable estimate would be 15 small cells.*

Step 7. *Thus, Number of split cells within the given area = 15*

Number of channels per split cell = 120
Number of channels with cell splitting = 15 × 120 = 1800 channels

Step 8 *Hence, increase in number of channels = 1800 / 480 =* **3.75 times**

Comments on the results Therefore, with an estimate of 15 split cells in the given square area, the number of channels contained in that area, with cell splitting, is 3.75 times more than the number of channels without cell splitting. This is quite close to the theoretical limit of 4 times increase in system capacity as given by Eq. (7.15) by putting $n = 1$. It is obvious that cell splitting increases the number of channels within the same coverage area, and thereby the overall system capacity.

Antenna downtilting, which deliberately focuses radiated energy from the cell-site towards the earth (rather than towards the horizon), is often used to limit the radio coverage of newly formed microcells.

7.5.3 Cell Sizes in Cellular Architecture

In practice, all cells in a cellular system are not split at the same time. Due to difficulties in installation of split cells at certain locations, different cell sizes will exist simultaneously. In such situations, special care in the system design needs to be taken to keep the distance between cochannel cells at the minimum required value. Hence, channel assignments become more complicated. To the same extent, large and small cells can be isolated by selecting a group of frequencies that will be used only in the cells located between the large cells on one side and the small cells on the other side. This is needed in order to eliminate the interference being transmitted from the large cells to the small cells. Hand-off issues must also be addressed so that high-speed and low-speed mobile subscribers can be simultaneously accommodated.

Figure 7.6 shows a typical cell distribution in a cellular network, in which the service area includes low-traffic (rural), medium-traffic (suburban), high-traffic (town or highways areas), thereby requiring a mix of large-as well as small-sized cells.

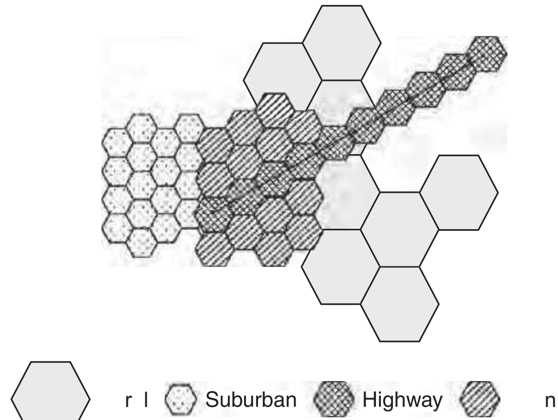

Fig. 7.6 *Cell distribution in a cellular network*

There are a number of cell sizes which are deployed in a cellular network to provide comprehensive area coverage. This is needed to support traffic variations in different geographical areas having various applications. The cells are distributed to form a hierarchy with different cell sizes such as *femtocells* (smallest cell used for interconnecting personal wireless devices such as cellphones, laptops, notepads, within a few metres), *picocells* (small cells inside a building to support WLANs within tens of metres), *microcells* (used in urban areas to support PCS within hundreds of metres), *macrocells* (to cover metropolitan areas of several kilometres), and *megacells* (used with satellites to cover nationwide areas within hundreds of kilemetres).

An ideal cellular network has a mixed hierarchy of different sizes of cells to achieve a comprehensive coverage of a variety of applications. For example, femtocells are used to connect personal wireless devices, picocells for indoor users, microcells for pedestrians in the streets or office/residential buildings in dense urban areas, macrocells for vehicle drivers in suburban/rural areas, and megacells to cover users in airplanes.

There are certain advantages of using different sizes of cells in a hierarchical cellular architecture such as

– to extend the coverage of the areas that are difficult to cover by a large cell
– to increase the capacity of the cellular network for those areas that have a higher density of subscribers
– to provide services to increased number of wireless devices and the need for communication between these devices, that is, connecting laptops with cellphones

Microcells cover small distances, usually less than 100 metres, using cell-sites with antennas mounted on street light posts or on the sides of buildings. The reason for this is that the cells designed to cover suburban areas have antennas on tall towers or rooftops of high buildings to cover a large area. However, signals from these antennas cannot propagate into urban areas or indoor environments. So there is a need to install low-power base stations with their antennas mounted on the walls to cover a smaller area, resulting in the creation of a smaller sized cell. The limitation of a small cell is based on the accuracy of cell-site locations and control of the radiation patterns of cell-site antennas. In microcell systems, a practical hand-off problem called *cell dragging* results from pedestrian users which provide a very strong signal to the cell-site. To resolve this issue, hand-off threshold levels and radio-coverage parameters must be adjusted carefully.

7.6 REVIEW OF MODULATION TECHNIQUES

Modulation is the process of encoding information signals from a message source in a manner suitable for wireless transmission. It involves translating a baseband information signal to a bandpass signal at radio carrier frequencies that are very high compared to the baseband frequency. The baseband signal is called the

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The goal of a modulation scheme is to transport the information signal through the wireless radio channel with the best possible quality while occupying least amount of RF spectrum. Modulation is a difficult process, given the hostile mobile radio channels affected by small-scale fading and multipath conditions.

modulating signal, and the bandpass signal is called the *modulated signal*. Demodulation is the process of extracting the baseband information from the modulated carrier signal at the receiver.

7.6.1 Need of Modulation

Modulation is needed for transmission of analog information as well as digital data through a wireless channel. Since the mode of transmission of a signal in a wireless medium is via electromagnetic waves which are analog in nature, the information signal (analog speech or digital data or digitised speech) has to be translated to another form of analog signal suitable for wireless transmission. Digital data is first translated into the equivalent baseband analog signal prior to modulation to enable transmission over a wireless medium. There are two main reasons for analog modulation of analog information signals:

- For unguided wireless transmission, it is virtually impossible to transmit baseband signals because the size of the required antennas would be of impractical extremely large.
- Modulation allows frequency division multiplexing (FDM) for transmission of multiple baseband signals having wider bandwidths simultaneously.
- The wireless medium has certain characterisitics such as propagation path loss, reflection, diffraction, attenuation through obstacles which depend upon the frequency of transmission of the electromagnetic waves carrying the baseband information. Different applications require appropriate carrier frequencies, for example, low frequency bands for submarines, medium-frequency bands for handheld devices (AM/FM/TV broadcast and cellular) , high-frequency bands for direct microwave communications, and very-high-frequency bands for satellite links, etc.

The types of signal conversions in the context of wireless communications can be categorised in three different forms as given below.

Analog-to-Analog Typically, a baseband analog signal such as speech must be modulated onto a much higher frequency analog carrier signal for wireless trannsmission.

Digital-to-Analog Digital data and digital signals such as computer-generated data must be converted to analog signals for wireless transmission.

Analog-to-Digital-Analog Analog voice signals are normally digitised prior to transmission over either wire-line or wireless media to improve quality and to take advantage of TDM schemes. The digitised voice data must be modulated onto an analog carrier signal for wireless trannsmission.

If the baseband information is of the form of an analog signal then the modulation technique is referred to as *analog modulation*. But if the baseband information is of the form of digital data then the modulation trechnique is referred to as *digital modulation*. In both cases, the carrier signal is a high-frequency analog signal so that the wireless transmission is possible using suitable antennas. Thus, the property that distinguishes digital modulation communications systems from conventional analog communications modulation systems is the nature of the modulating signal—digital data or analog signal respectively. Both analog and digital modulation systems use high-frequency analog carrier signals to transport the information through wireless medium.

7.6.2 Analog Modulation (AM, FM, PM)

Consider the analog information signal, also called modulating signal, is represented by

$$m(t) = A_m \cos (2\pi f_m t) \tag{7.16}$$

where A_m is its amplitude, and f_m is the modulating frequency.

As discussed in the previous section, the carrier signal used in the modulation process has to be necessarily an analog signal for wireless communications. The carrier signal can be considered of the form represented by the cosine function as

$$c(t) = A_c \cos \; 2\pi f_c t + \theta(t) \tag{7.17}$$

The carrier signal has three components as the amplitude A_c, the frequency f_c, and the phase angle $\theta(t)$; any one of them can be varied in accordance with the instantaneous value of the analog information signal, resulting into amplitude, frequency, and phase modulation respectively.

Amplitude Modulation (AM) According to the definition of the amplitude modulation, the amplitude of the carrier signal is varied linearly with the instantaneous value of the modulating signal $m(t)$, and can be represented mathematically by the expression (ignoring the phase angle of the carrier signal):

$$S_{AM}(\mathrm{t}) = \; A_c + m(t) \; \cos(2\pi f_c t) \tag{7.18}$$

Or, $$S_{AM}(t) = \; A_c + A_m \cos(2\pi f_m t) \; \cos(2\pi f_c t)$$

Or, $$S_{AM}(t) = A_c \; 1 + (A_m / A_c) \cos(2\pi f_m t) \; \cos(2\pi f_c t)$$

Or, $$S \quad (t) = \quad 1 + \quad \cos(2\pi \quad t) \; \cos(2\pi \quad t) \tag{7.19}$$

where $m_a = (A_m / A_c)$ is defined as the amplitude modulation index of an AM signal, and often expressed as a percentage, and is called *per-cent modulation*. Practically, the value of m_a is kept as less than or equal to unity, otherwise it will result into overmodulation causing distortion in the AM signal.

From the above expression, it is clear that AM involves the multiplication of the modulating signal by the carrier signal. The AM signal contains a dc component at carrier frequency, and two sidebands at $(f_c + f_m)$ and $(f_c - f_m)$. Thus, the bandwidth of an AM signal is limited to $2 f_m$. The total transmitted power in an AM signal is given by the expression

$$= \quad 1 + (\quad ^2 / 2) \tag{7.20}$$

where P_c is the transmitted power in the carrier signal. It is preferred to have a larger possible value of m_a (not exceeding 1) so that most of the signal power is used to carry information. This is the reason that AM transmitters are generally operated with the per-cent modulation as close to 100% as possible. Thus, the value of amplitude modulation index influences the transmitted power but does not affect the bandwidth of the AM signal. There is a linear relationship between the transmitted signal power and the received signal quality.

It can be easily deduced that an AM signal contains unnecessary components because the carrier signal does not carry the information signal and each of the two sidebands contains the complete envelope of the modulating signal. A variant of AM, known as *double sideband suppressed carrier* (DSBSC), filters out the carrier frequency and sends both sidebands. This saves some transmitted power but uses the same bandwidth as the original AM. Another popular variant of AM, known as *single sideband* (SSB), transmits only one of the sidebands, eliminating the carrier signal and one of the sidebands. DSBSC AM is not often used on its own as a modulation scheme, but is used as the basis for generating SSBSC or SSB signals. It requires much less transmitted power and the bandwidth is also reduced to half that of the original AM signal. However, the disadvantage of suppressing the carrier signal is that the carrier signal can be used for synchronisation purpose. So a compromise approach is a *vestigial sideband* (VSB), which uses a reduced power carrier signal and one sideband.

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AM is widely used in medium-and high-frequency bands for long-distance radio broadcast applications, aircraft communications in the VHF frequency range, and citizen-band radio. SSB is used in point-to-point long distance telephony in HF band, and VSB is used in colour-television broadcast and stereo FM applications.

Angle Modulation As mentioned earlier, only three parameters of an analog carrier signal can be changed or modulated in order for it to carry information: amplitude, frequency, and phase. Frequency and phase are closely related since frequency (expressed in radians per second) is the rate of change of phase angle (expressed in radians). If either frequency or phase is changed in a modulation process, the other will change as well. That is why both frequency and phase modulation are grouped together in the term 'angle modulation.'

The most important advantage of angle modulation over amplitude modulation is the possibility of a greatly improved SNR value. However, SNR improvement is achieved in increased bandwidth because an FM signal may occupy several times as much bandwidth as that required for an AM signal. Frequency modulation is used more extensively for radio broadcasting applications in the 88 MHz – 108 MHz band, for modulating the sound signal in TV transmission, for full-duplex fixed and mobile wireless communication systems, for satellite communications, and analog cellular communication systems such as AMPS. Phase modulation is used extensively in data communications, in some FM transmitters as an intermediate step in the generation of FM.

Frequency Modulation (FM) According to the definition of the frequency modulation, the frequency of the carrier signal is varied linearly with the instantaneous value of the modulating signal $m(t)$, and can be represented mathematically by the expression:

$$S \quad (t) = \quad \cos \; (2\pi \quad t) + + \quad \sin \; (2\pi \quad t) \tag{7.21}$$

where m_f is the frequency modulation index of FM signal, which defines the relationship between the amplitude of the modulating signal and the bandwidth of the FM transmitted signal, and is given by

$$= (\qquad) / \quad = \Delta \; / \tag{7.22}$$

where k_f is the frequency deviation constant measured in Hz/V, A_m is the peak value of the modulating signal, f_m is the maximum frequency of the modulating signal, and Δf is the peak frequency deviation of the FM signal. Since $S_{\ M}(t)$ is a nonlinear function of $m(t)$, the spectrum of an FM signal contains a carrier frequency and an infinite number of sidebands located on its either side, spaced at integer multiples of the modulating frequency f_m. The RF bandwidth of an FM signal is given by the approximate relationship as

$$B_{\ M} \approx 2\,(m_f + 1)\,f_m \tag{7.23}$$

Or,

$$B_{\ M} \approx 2\,(\Delta f + f_m) \tag{7.24}$$

The above expression is called *Carson's rule*. An FM signal has 98% of the total transmitted power within the RF bandwidth $B_{\ M}$. The value of the frequency-modulation index does not affect the average transmitted power but is directly proportional to the bandwidth of the FM signal.

Phase Modulation (PM) According to definition of phase modulation, the phase angle $\theta(t)$ of the carrier signal is varied linearly with the instantaneous value of the modulating signal $m(t)$, and can be represented mathematically by the expression

$$S_{PM}(t) = A_c \cos \; 2\pi f_c t + + k_\theta\, m(t) \tag{7.25}$$

where k_θ is the phase deviation constant, measured in radians/volt. A PM signal can be generated by first differentiating the modulating signal $m(t)$ and then using this output as the input to a frequency modulator. The phase modulation index, m_p is given by

$$m_p = k_\theta\, A_m = \Delta\theta$$

where $\Delta\theta$ is the peak phase deviation of the PM signal.

The RF bandwidth of a PM signal is same as that of an FM signal and both have greater bandwidth than an AM signal.

7.6.3 Comparison of AM, FM, and PM

Table 7.1 provides a comparative study of some of the features of AM, FM, and PM.

Table 7.1 Comparison of the features of AM, FM, and PM

S. No.	Amplitude modulation (AM)	requency modulation (M) or phase modulation (PM)
1.	Amplitude and transmitted power of an AM signal change with modulation.	Amplitude and transmitted power of an FM or PM signal do not change with modulation.
2.	An AM signal has an envelope that is used to demodulate the modulating signal.	An FM or PM signal does not have an envelope, and the demodulator need not to respond to amplitude variations.
3.	AM signals are more susceptible to noise, because information is stored as amplitude variations rather than frequency variations.	FM/PM signals are less susceptible to atmospheric noise, because information is stored as frequency/phase variations rather than amplitude variations.
4.	Modulation index cannot be changed automatically.	The modulation index can be varied to obtain greater SNR.
5.	AM signals occupy lesser bandwidth.	FM or PM signals occupy more bandwidth.
6.	An AM transmitter uses class A or AB amplifiers due to linearity requirement.	An FM or PM transmitter can use efficient class C amplifiers since amplitude linearity is not a concern.
7.	Modulation can be accomplished at low power or high power levels.	Modulation can be accomplished at low power levels.

7.6.4 Digital Modulation (ASK, FSK, PSK)

Digital modulation is the transmission of digitally modulated analog carrier signals in wireless communication systems. Digital modulation systems offer several outstanding advantages over traditional analog modulation systems such as easier and faster signal processing as well as multiplexing, and greater noise immunity and robustness to channel impairements.

Digital modulation is a simple case of transmitting digital data using analog signals. As mentioned earlier, the process of modulation involves operation on one of the three characteristics of the analog carrier signal: amplitude, frequency, and phase. Accordingly, there are three basic modulation techniques for transforming digital data into analog signals as amplitude shift keying (ASK), frequency shift keying (FSK), and phase shift keying (PSK) respectively. In all these digital-modulation techniques, the modulated signal occupies a bandwidth centered on the carrier frequency.

Amplitude Shift Keying (ASK) In ASK, the two binary values, logic 1 and logic 0, of the information data (modulating signal) are represented by two different amplitudes of the carrier signal, $c(t) = A_c \cos(2\pi f_c t)$. The ASK signal can be mathematically expressed as

$$S_{ASK}(t) = \begin{Bmatrix} A_c \cos(2\pi f_c t) & \text{for binary 1} \\ 0 & \text{for binary 0} \end{Bmatrix} \tag{7.26}$$

The transmission bandwidth for an ASK signal is given by

$$B_{ASK} = (1 + r)R \tag{7.27}$$

where r (typically $0 < r < 1$) is related to the technique by which the signal is filtered to establish a bandwidth for transmission; and R is the bit rate. Thus, the bandwidth is directly related to the bit rate.

This is the simplest form of a digital-modulation scheme. But it is an inefficient digital-modulation technique. ASK is susceptible to sudden amplitude variations due to noise, multipath propagation and interference. So ASK is not used in wireless transmission. It requires low bandwidth, and is typically used only up to 1200 bps data rate on voice-grade lines as in telemetry. It is used to transmit digital data over optical fiber for LED transmitters, and wireless infrared transmissions using a directed beam or diffused light in wireless LANs applications.

Frequency Shift Keying (FSK) The most common form of FSK is binary FSK (BFSK), in which the two binary values, logic 1 and logic 0, of the information data are represented by two different frequencies of the carrier signal, $c(t) = A_c \cos(2\pi f_c t)$ near the carrier frequency f_c. The FSK signal for one bit duration can be mathematically expressed as

$$S_{B\ \ SK}(t) = \begin{Bmatrix} A_c \cos(2\pi f_1 t) & \text{for binary 1} \\ A_c \cos(2\pi f_2 t) & \text{for binary 0} \end{Bmatrix} \tag{7.28}$$

where f_1 and f_2 are typically offset from the carrier frequency f_c by equal but opposite values. The transmission bandwidth for BFSK is same as that of ASK. BFSK is less susceptible to error than ASK but it requires larger bandwidth compared to ASK. The peak frequency offset is constant and always at its maximum value, and the highest fundamental frequency is equal to half the information bit rate. BFSK has a poorer error performance than PSK or QAM and is rarely used for high-performance digital transmission systems. Therefore, BFSK is used in low-performance, low cost, asynchronous data modems that are used for data communications over analog voice-band telephone lines. It is typically used up to 1200 bps data rate on voice-grade lines in high frequency (HF – 3 MHz to 30 MHz) radio transmissions.

Multiple FSK (MFSK) is a higher level version of the FSK modulation technique, in which more than two frequencies are used. In MFSK, each signaling element represents more than one bit. The MFSK signal for one-signal-element duration can be expressed as

$$S_{M\ \ SK}(t) = A_c \cos(2\pi f_i t) \quad \text{for } 1 \le i \le M \tag{7.29}$$

where $f_i = f_c + (2i - 1 - M) f_d$; f_c being the carrier signal frequency, f_d being the difference frequency, and M being the number of different signal elements ($= 2^L$, L is the number of bits per signal element). It occupies a bandwidth of $2 M f_d$. MFSK is less susceptible to errors. It finds applications in wireless LANs.

Phase Shift Keying (PSK) PSK is a constant-amplitude digital modulation scheme. When higher data rates are required in a band-limited channel that cannot be achieved with FSK, PSK is often used. The simplest form of PSK is binary PSK (BPSK), in which the two binary values, logic 1 and logic 0, of the information data are represented by two different phases separated by 180 of the carrier signal, $c(t) = A_c \cos(2\pi f_c t)$. The BPSK signal for one-bit duration can be mathematically expressed as

$$S_{BPSK}(t) = \begin{Bmatrix} A_c \cos(2\pi f_c t) & \text{for binary 1} \\ A_c \cos(2\pi f_c t \quad \pi) & \text{for binary 0} \end{Bmatrix} \tag{7.30}$$

BPSK is also called *biphase modulation* or *phase reversal keying*. It is a form of square-wave modulation of a CW signal. PSK has good SNR but must be demodulated synchronously, which means a reference carrier signal is required to be received to compare with the phase of the received signal, which in turn makes the demodulator complex. The transmission bandwidth for BPSK is same as that of ASK but the bandwidth efficiency is better. An alternate form of BPSK is differential PSK (DPSK) in which a binary 1 is represented by sending a signal bit of opposite phase (by 180) to the preceding one and a binary 0 is represented by sending a signal bit of the same phase as the previous one. The term 'differential' refers to the fact that the phase shift is with reference to the previous bit. The information can be represented in terms of the changes between successive data bits. This enables the coherent detection at the receiver more reliable. The phase of each symbol is compared with that of the previous symbol, rather than a constant reference phase. The advantage of DBPSK is that carrier recovery circuit is not needed which makes it simple to implement. The disadvantage is the requirement of additional 1–3 dB SNR to achieve the same BER performance as that of conventional PSK scheme.

Facts to Know!

Among the many digital-modulation techniques in use, some are responsive primarily to the goal of spectral efficiency, while others focus on the objective of achieving a narrow power spectrum.

The large variation in signal amplitude due to Rayleigh fading encountered, render the digital amplitude-modulation technique almost inoperative. There are basically two broad digital-modulation strategies emerging that emphasise the use of PSK or MSK derived digital-modulation techniques.

7.6.5 Linear Digital Modulation (QPSK, OQPSK, QAM)

The family of linear digital-modulation techniques requires a high degree of linearity in modulating the carrier by baseband signal and RF power amplification before transmission. Linear digital modulation is more spectral efficient, but requires a linear power amplifier so as to avoid the signal amplitude variations which may result in intermodulation products. The most important linear digital-modulation techniques are QPSK, OK-QPSK, π/4-shift QPSK and higher-level PSK. The QPSK-type linear digital-modulation techniques offer higher spectral efficiency and are considerably simpler to be implemented.

Quadrature Phase Shift Keying (QPSK) QPSK is an M-ary constant-amplitude digital-modulation scheme in which the number of bits is 2 and number of signaling elements is 4. Each signaling element is represented by two bits. This is called *dibit system*, which can carry twice as much data in the same bandwidth as can a single-bit system, provided the SNR is high enough. Accordingly, QPSK has 4 different phase shifts, separated by multiples of π/2 or 90 of the carrier signal, $c(t) = A_c \cos(2\pi f_c t)$. The QPSK signal for one symbol duration consisting of two bits, each of which can be mathematically expressed as

$$S_{QPSK}(t) = \begin{cases} A_c \cos(2\pi f_c\ t \quad \pi/4) & \text{for } 11 \\ A_c \cos(2\pi f_c\ t \quad 3\pi/4) & \text{for } 01 \\ A_c \cos(2\pi f_c\ t - 3\pi/4) & \text{for } 00 \\ A_c \cos(2\pi f_c\ t - \pi/4) & \text{for } 10 \end{cases} \tag{7.31}$$

Figure 7.7 shows the QPSK modulation scheme. The input is a data stream of binary digits with a data rate of $R = 1/T_b$, where T_b is the bit duration. This data stream is converted into two separate bit streams by taking alternate bits at the rate of $R/2$ bps.

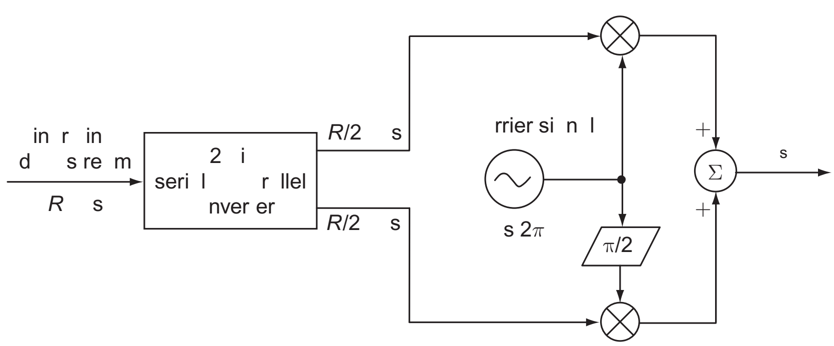

Fig. 7.7 | *QPSK modulator*

QPSK is characterised by two parts of the baseband data signal—the inphase signal $I(t)$ and the quadrature signal $Q(t)$. For this reason, the data rate of individual signals, $I(t)$ and $Q(t)$, is half that of the baseband signal. This also cuts the bandwidth requirement of QPSK to half as compared to BPSK. The modulated signal has 180 and 90 phase shifts. The QPSK signal can be expressed as

$$S_{\quad SK}(t) = 1/\sqrt{2} \quad (t)\cos(2\pi \quad t) - \quad (t)\sin(2\pi \quad t) \tag{7.32}$$

Facts to Know!

QPSK modems are used in systems applications where the 1 bps/Hz theoretically spectral efficiency of BPSK modems is insufficient for the available bandwidth.

The signal transitions are abrupt and unequal and this causes large spectrum dispersion. QPSK also needs more power and higher C/N ratio (approximately 3 dB) than BPSK, to obtain the same performance for the same probability of error. A typical differential QPSK (DQPSK) uses phase shifts with respect to the phase of the previous symbol.

Offset Keyed QPSK (OQPSK) The difference between QPSK and OQPSK is in the alignment of the in-phase signal $I(t)$ and the quadrature signal $Q(t)$. OQPSK is obtained by introducing a shift or offset equal to one bit delay (T_b) in the quadrature signal $Q(t)$, as shown in the previous figure. The OQPSK signal can be expressed as

$$S_{OQPSK}\,(t) = 1/\sqrt{2}\quad I(t)\cos(2\pi f_c\, t) - Q(t - T_b)\sin(2\pi f_c\, t) \tag{7.33}$$

This ensures that the $I(t)$ and $Q(t)$ signals have signal transitions at the time instants separated by $T_b/2$, where T_b is the symbol period. Figure 7.8 shows the OQPSK modulation scheme.

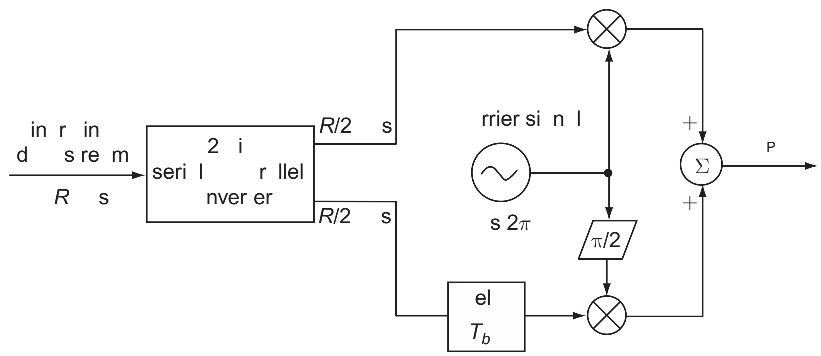

Fig. 7.8 *OQPSK modulator*

Facts to Know!

The design of coherent QPSK and O-QPSK demodulators is quite complicated, particularly if very fast modem synchronisation is needed. DQPSK avoids the need for a complex carrier recovery circuit, thereby improving the synchronising speeds of the demodulator.

The modulated signal transitions are 90 degree maximum. It has no 180-degree phase shift and this results in reduction of out-of-band radiations. However, the abrupt phase transitions still remain. Another disadvantage of OQPSK is that changes in the output phase occur at twice the data rate in either I or Q channels. Consequently, the minimum bandwidth is twice that of QPSK for a given transmission data rate.

π/4-Phase Shift QPSK π/4-QPSK is a compromise between QPSK and OK-QPSK. It can be regarded as a modification to QPSK and has carrier phase transitions that are restricted to 45 and 135 . Like OQPSK, the carrier phase does not undergo instantaneous 180 phase transition, therefore, avoiding the use of a less-efficient linear amplifier. In BPSK and QPSK, the input data stream is encoded in the absolute position in the constellation. In π/4-QPSK, the input data stream is encoded by the changes in the amplitude and direction of the phase shift. π/4-QPSK uses two QPSK constellations offset by π/4. Signaling elements are selected in turn from the two QPSK constellations. Transition must occur from one constellation to the other one. This ensures that there will always be a phase change for each input symbol.

Thus, the main advantage of π/4-QPSK is that of the reduced envelope fluctuation as compared to that of QPSK. Moreover, π/4-QPSK can be noncoherently demodulated. The BER performance of the π/4-QPSK is controlled by cochannel interference rather than fading. The π/4-QPSK digital modulation technique is widely used in most second-generation cellular systems such as United States Digital Cellular (IS 54) TDMA systems, PCS systems, and Japanese Digital Cellular (JDC) standards.

Facts to Know!

An advantage of π/4-DQPSK modulation is that the signal can easily be differentially demodulated, whereas it is difficult to differentially demodulate the OQPSK.

Multilevel or M-ary PSK (MPSK) It is possible to transmit more than two bits at a time using multiple different phase angles, resulting in the Multilevel PSK (MPSK) digital-modulation technique. Further, each angle can have more than one amplitude. For example, a standard 9600-bps modem uses 12 phase angles, four of which have two amplitude values for a total of 16 different signal elements. The transmission bandwidth for MPSK is given by the expression

$$B_{MPSK} = \left(\frac{1+r}{L}\right)R = \left(\frac{1+r}{\log_2 M}\right)R \tag{7.34}$$

where r (typically $0 < r < 1$) is related to the technique by which the signal is filtered to establish a bandwidth for transmission; L is the number of bits encoded per signal element, R is the bit rate, and M is the number of different signal elements. The bandwidth efficiency is much better as compared to QPSK. For example, if tribits are formed producing eight different output phases, it is called 8-PSK.

Quadrature Amplitude Modulation (QAM) QAM is a form of digital modulation similar to PSK except the digital information is contained in both the amplitude and the phase of the modulated signal. It is an efficient way to achieve high data rates with a narrowband channel by increasing the number of bits per symbol, and uses a combination of amplitude and phase modulation. QAM can either be considered a logical extension of QPSK or a combination of ASK and PSK. In QAM, two different signals are sent simultaneously on the same carrier frequency, by using two identical copies of the carrier frequency, one shifted by the other by 90 with respect to the other. Figure 7.9 shows the QAM modulation scheme.

The input is a data stream of binary digits at a rate of R bps. This data stream is converted into two separate data streams of $R/2$ bps each, by taking alternate bits for the two data streams. One data stream is ASK modulated on a carrier frequency f_c, and the other data stream is ASK modulated by the same carrier signal shifted by 90 . The two modulated signals are then added and transmitted. The QAM signal can be expressed as

$$S_{QAM}(t) = d_1(t)\cos(2\pi f_c t) + d_2(t)\sin(2\pi f_c t) \tag{7.35}$$

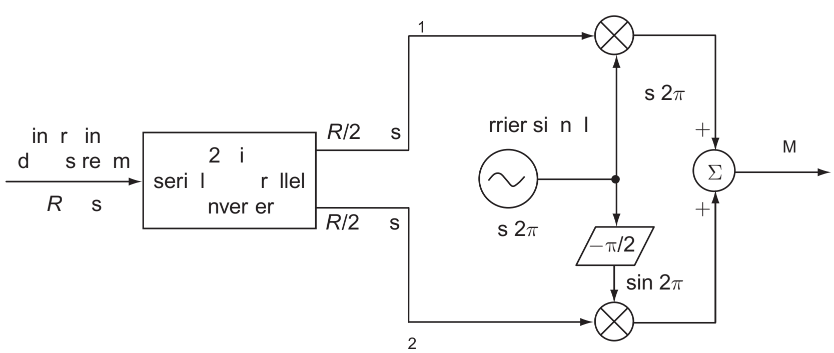

Fig. 7.9 *QAM modulator*

For a given system, there are finite numbers of allowable amplitude-phase combinations. If a 2-level ASK is used, then each of the two data streams can be in one of two states and the combined data stream can be in one of 4 states. This is essentially QPSK. If a 4-level ASK is used then the combined data stream can be in one of 16 states, called 16-QAM. Higher-level QAM is used in fixed terrestrial microwave digital radio, digital video broadcast cable, and modems. Similarly, 64-QAM and 256-QAM can be implemented. The more the number of states, the higher the data rate that can be supported within a given bandwidth, at the expense of higher probability of channel error. The ASK and PSK are combined in such a way that the positions of the signaling elements on the constellation diagram are optimised to achieve the largest distance possible between elements. This reduces the likelihood of one element being misinterpreted as another element, thereby reducing the probability of occurrence of errors. The m-ary QAM having more than 16 levels are more bandwidth efficient than BPSK, QPSK or 8-PSK and are used for high data rate transmission applications. However, the complexity to implement m-ary QAM increases requiring the use of linear amplifiers, and so the susceptibility to errors due to noise and distortion. QAM is more efficient in terms of bandwidth than QPSK, but it is also more susceptible to noise due to amplitude variations.

Facts to Know!

In general, linearly amplified m-ary QAM systems such as 16-QAM, 64-QAM, and 256-QAM are more spectrally efficient than linearly amplified QPSK, which has a theoretical fundamental limit of 2 bps/Hz.

EXAMPLE 7.9 Comparison of ASK, FSK, m-PSK and m-QAM

Define bandwidth efficiency. Compare the minimum bandwidth requirement and bandwidth efficiency for ASK, FSK, m-ary PSK and m-ary QAM digital modulation techniques in a tabular form.

Solution

Bandwidth efficiency or spectral efficiency is defined as the ratio of the transmission bit rate to the minimum bandwidth required for a particular digital-modulation technique. It is often used to compare the performance of different digital-modulation techniques. It is generally normalised to a 1-Hz bandwidth and, therefore, indicates the number of bits that can be transmitted for each hertz of bandwidth through a transmission medium.

Table 7.2 summarises the minimum bandwidth required and bandwidth efficiency for ASK, FSK, m-ary PSK and m-ary QAM digital modulation techniques. f_b represents the input-data bit rate.

Table 7.2 *Performance comparison of digital-modulation techniques*

Modulation scheme	*Encoding scheme*	*Possible outputs*	*Minimum bandwidth*	*Bandwidth efficiency*
ASK	Single bit	2	f_b	1
BFSK	Single bit	2	f_b	1
BPSK	Single bit	2	f_b	1
QPSK	Dibits	4	$f_b/2$	2
8-PSK	Tribits	8	$f_b/3$	3
8-QAM	Tribits	8	$f_b/3$	3
16-PSK	Quadbits	16	$f_b/4$	4
16-QAM	Quadbits	16	$f_b/4$	4
32-PSK	Five bits	32	$f_b/5$	5
64-QAM	Six bits	64	$f_b/6$	6

7.6.6 Continuous Envelope Modulation (MSK, GMSK)

Continuous envelope digital modulation, also called continuous-phase digital modulation (CPM), techniques avoid the linearity requirements of RF power amplification at the transmitter. Digital-modulation techniques derived from the CPM family have quite narrow power spectra. On the other hand, the spectral efficiency is somewhat lower. Among the important constant-envelope or continuous-phase digital-modulation techniques are MSK and GMSK.

Minimum Shift Keying (MSK) MSK can be regarded either as a special case of OQPSK or as a form of FSK modulation. As a form of MFSK, the MSK signal is given by the expression

$$S_{MSK}(t) = \left\{ \begin{array}{ll} \sqrt{\frac{2E_b}{T_b}} \cos \; 2\pi f_1 \; t \quad \theta(0) & \text{for binary 1} \\ \sqrt{\frac{2E_b}{T_b}} \cos \; 2\pi f_2 \; t \quad \theta(0) & \text{for binary 0} \end{array} \right\} \tag{7.36}$$

where E_b is the transmitted signal energy per bit, and T_b is the bit duration. The phase θ(0) denotes the value of the phase at time $t = 0$. The two frequencies f_1 and f_2 satisfy the following equations:

$$f_1 = f_c + \frac{1}{4T_b} \text{; and } f_2 = f_c - \frac{1}{4T_b}$$

The baseband signal is filtered sinusoidaly to produce a graceful transition from one binary state to another. MSK is a binary modulation with symbol interval T_b and frequency deviation is equal to 1/4 T_b. And there is phase continuity of the modulated RF carrier at the bit transitions. RF phase varies linearly exactly 90 with respect to the carrier over one bit period T_b. The spacing between f_1 and f_2 is the minimum that can be used and allow successful detection of the received signal at the receiver. That is why it is called minimum shift keying digital-modulation technique.

When MSK is viewed as a special case of OQPSK, the carrier signal is multiplied by a sinusoidal function. The MSK signal can be expressed as

$$S_{MSK}(t) = I(t) \cos\left(\frac{\pi t}{2T_b}\right) \cos(2\pi f_c t) + Q(t - T_b) \cos\left(\frac{\pi t}{2T_b}\right) \sin(2\pi f_c t) \tag{7.37}$$

The MSK is also known as continuous phase FSK (CPFSK) because the phase is continuous during the transition from one bit time to the next bit time. It has a better BER performance than conventional binary FSK for a given SNR value. Its disadvantage is that it requires synchronous circuits and is, therefore, more expensive.

Facts to Know!

The MSK demodulator operates in the same manner as the OQPSK receiver. However, a different filtering technique is required to ensure ISI-free transmission.

Gaussian Minimum Shift Keying (GMSK) GMSK is a binary digital modulation technique which can be considered as a special case of MSK. The use of a pre-modulation low-pass filter with Gaussian characteristics with the MSK approach achieves the requirement of uniform envelope, in addition to spectral containment and reduction in the transmitted bandwidth of the signal. The Gaussian filter is used to suppress out-of-band noise, to reduce the sidelobe levels of the power spectrum and adjacent channel interference considerably. GMSK uses less bandwidth than conventional FSK because the Gaussian filter causes the transmitted frequency to move gradually between two frequency offsets. Thus, baseband Gaussian filtering stabilises the instantaneous frequency variation over time and smoothens the phase profile of the MSK signal.

In GMSK, each transmitted symbol spans several bit periods. GMSK can be noncoherently detected as in FSK, or coherently detected as in MSK. GMSK provides high spectrum efficiency, excellent power efficiency, and a constant amplitude envelope that allows class C power amplifiers to be used minimising power consumption. The premodulation Gaussian filtering introduces marginal ISI in the transmitted signal, but the degradation in performance can be minimised by selecting the appropriate 3-dB-bandwidth-bit duration product $B \times T_b$ of the filter. The power spectrum of GMSK with a $B \times T_b$ value of infinity is equivalent to that of MSK. As the $B \times T_b$ product decreases, the sidelobe levels fall off very rapidly. For example, for a $B \times T_b = 0.5$, the peak of the second power lobe is more than 30 dB below the main power lobe. However, reducing $B \times T_b$ beyond a certain value, increases the irreducible error rate generated by the Gaussian filter due to ISI. If this irreducible error rate is less than that produced by the mobile channel due to Doppler shift produced by the mobile speed, GMSK modulation can be effectively used in mobile radio systems. GMSK is widely used in the GSM cellular radio and PCS systems.

7.6.7 Choice of Digital-Modulation Techniques

A digital-modulation technique used for mobile environment should utilise the transmitted power and RF channel bandwidth as efficiently as possible. This is because the mobile radio channel is both power-and band limited. To conserve power, efficient source-encoding schemes are generally used but this is at the cost of bandwidth. Whereas to save the spectrum in bandlimited systems, spectrally efficient digital-modulation techniques are used. Spectral efficiency is defined as the number of bits per second per hertz (bps/Hz). Spectral efficiency influences the spectrum occupancy in a mobile radio system. Theoretically, an increase in the number of modulation levels results into higher spectral efficiency. But the precision required at the demodulator to detect the phase and frequency changes also increases exponentially. This results into higher *S/N* requirement to achieve same BER performance. The objective of spectrally efficient digital modulation technique is to maximise the bandwidth efficiency. It is also desirable to achieve the bandwidth efficiency at a prescribed BER with minimum transmitted power.

> **Facts to Know!**
>
> Power efficiency and spectral efficiency are among the most important requirements of digital cellular mobile radio system. Non-linear amplifiers are used for cost-effective and small size requirements.

One of the effects produced by mobile environment is the distortion of the signal due to delay spread, which results in intersymbol interference. It is known that if the effects of delay spread are ignored or equalised, the spectral efficiency improves for higher level modulation. And this is realised at the cost of high SNR. Theoretically, the impact of delay spread can be reduced by adopting higher level digital-modulation techniques. For example, a 4-level modulation signal transmitted at the data rate of 200 kbps would have a symbol duration of about 10 μs as against 20 μs in 16-level modulation. However, under the influence of delay spread, the simulations indicate that no significant performance improvement can be achieved as the level of modulation exceeds 4, even if SNR approaches infinity. It is indicated that BER performance depends strongly on the rms value of delay spread. Thus, a 4-level modulation technique seems to be reasonable when delay spread is significant.

The energy of a modulated carrier is distributed in the main lobe and side lobes. The power spectrum is a determinant of adjacent channel interference. A power-efficient digital communication system will have to be non-linearly amplified for cost-effective utilisation of power. When a bandlimited linearly modulated carrier undergoes non-linear amplification, the filtered side lobes re-appear. This causes severe adjacent channel and cochannel interference. The BER performance is also degraded. Hence, such systems may not efficiently utilise the available frequency spectrum. Power efficiency gained is lost as a result of nonlinear amplification.

Constant-phase digital-modulation techniques can be nonlinearly amplified without significant spectral regeneration. The difference in time alignment of the bit streams for QPSK and OQPSK does not change the power spectral density distribution. The difference in power efficiency using QPSK or π/4-shift QPSK is very small. The power spectrum of MSK signal concentrates more around the main lobe and decreases more quickly in the other side lobe areas. MSK is more spectrum efficient than QPSK but it has a wider main lobe. The spectrum occupancy can be controlled by the normalized 3 dB $B \times T_b$ product of the Gaussian pre-modulation filter. If $B \times T_b$ is infinite this gives a phase change of 90 degrees in each interval, which is MSK modulation. The value of $B \times T_b$ can be selected considering overall spectrum efficiency requirements. The spectral spread is nearly the same till $1/T_b$ but beyond that both QPSK and GMSK differ significantly. Among all these schemes, the complexity of QPSK is quite high.

In a channel without any bandwidth restriction, the QPSK modulated carrier has a uniform envelope but a wide spread of sideband energy. To contain this, post filtering can be adopted. The filtering is acceptable to receivers, but the effect of filtering has been to introduce carrier amplitude variations. If the signal is transmitted through a nonlinear power amplifier in the transmitter then the amplitude variations result in the regrowth of the side lobes, which effectively expand the bandwidth again. For a nonfiltered MSK signal, 99% of the power is contained within a channel of certain bandwidth. Whereas for GMSK ($B \times T_b = 0.5$), it is contained within a channel of lesser bandwidth, and for GMSK (with $B \times T_b = 0.3$) in a channel of even much lesser bandwidth. The out-of-band emission performance of QPSK and π/4-shift QPSK is almost the same when a reasonably linear amplifier is used. For QPSK, power is contained in a channel bandwidth much larger than that for nonfiltered MSK signal. Thus, less post-modulation filtering will be required for MSK than for QPSK to reduce the out-of-band power to a given value.

Table 7.3 shows the comparison of spectral efficiency and the required S/N (for BER of 1 in 10^6) for PSK and MSK digital modulation techniques.

Table 7.3 *Performance comparison of digital modulation techniques*

Digital modulation technique	*Spectral efficiency*	*Required S/N*
BPSK	1 bps/Hz	11.1 dB
QPSK	2 bps/Hz	14.0 dB
16-PSK	4 bps/Hz	26.0 dB
2-MSK	1 bps/Hz	10.6 dB
4-MSK	2 bps/Hz	13.8 dB

If most efficient bandwidth utilisation and moderate hardware complexity is the keynote, then π/4-QPSK is a better choice, whereas continuous phase-modulation techniques offer constant envelope, narrow power spectra, and good error-rate performance. Therefore, when out-of-band signal power, tolerance against filter parameter and nonlinear power amplifiers are important features, compromise in channel separation is permissible and higher circuit complexity is of less consideration then GMSK is the solution.

Table 7.4 summarises various digital-modulation techniques adopted in second-generation cellular and cordless telephone systems.

It is observed that though linear digital-modulation techniques offer better spectral efficiency, GMSK is considered the most promising digital-modulation technique. The European GSM system is based on GMSK modulation technique with a bit rate of 270.833 kbps and $B \times T_b = 0.3$, where B is the base bandwidth. The DECT (Digital European Cordless Telephone) system uses GMSK with $B \times T_b = 0.5$ at a data rate of 1,572 Mbps. The choice between linear digital modulation and constant-envelope digital modulation technique is influenced by the fact that how much maximum spectral efficiency must be stressed in comparison with the requirement of adjacent channel selectivity.

Table 7.4 *Spectral efficiency of digital cellular and cordless systems*

System	*Digital modulation technique*	*Channel bandwidth, kHz*	*Data rate, kbps*	*Spectral efficiency, bps/Hz*
USDC	π/4-QPSK	30	48.6	1.62
JDC	π/4-QPSK	25	42.0	1.68
GSM	GMSK ($B \times T_b = 0.3$)	200	270.8	1.35
CT-2	GMSK	100	72.0	0.72
DECT	GMSK ($B \times T_b = 0.5$)	1728	1572.0	0.67

7.7 EQUALISATION

Equalisation is used to overcome intersymbol interference due to channel time dispersion and diversity is used to overcome flat fading. Equalisation techniques are widely used to improve wireless link performance and received signal quality. In a wireless mobile communication system, transmitted signals often experience time dispersion and multipath signal fading. In addition to amplitude variations and carrier phase distortion, the channel introduces propagation delay dispersion.

Time-dispersive channels can cause intersymbol interference (ISI). For example, in a multipath scattering environment, the receiver sees delayed versions of a symbol transmission, which can interfere with other symbol transmissions. To overcome the effect of propagation delay dispersion due to multipath, channel equalisers for narrowband FDMA and TDMA systems and Rake receivers for wideband CDMA systems are used. The HIPERLAN-1 wireless LAN standard specification recommends using equalisation in indoor radio channels to achieve data rates of up to 23 Mbps. An equaliser attempts to mitigate ISI and thus improve the receiver's performance.

Facts to Know!

Equalisation is a method of amplitude and phase compensation that is used to minimise the impact of inter-symbol interference (ISI) on BER degradation. Frequency-selective fading is a major cause of ISI and BER degradation.

7.7.1 Equalisation Techniques

The equalisation is a technique used to help the demodulator to recover a baseband pulse with the best possible SNR, free of any ISI. Equalisation embodies a sophisticated set of signal-processing techniques, making it possible to compensate for channel-induced interference. The device which equalises the dispersive effect of a channel with memory is referred to as an equaliser. For channel equalisation at baseband, an equaliser is placed between the demodulator and decision-making device such that the output is ISI free. The principle of operation of an equaliser can be stated as follows:

- If a linear equaliser is used then ideally the transfer function of the equaliser should be the inverse of the transfer function of the wireless channel so that the effect of the channel characteristics at the input of the decision-making device at the receiver is completely compensated for by the linear equaliser.
- For a time-variant wireless channel, the channel-transfer function changes with time. Therefore, the equaliser should be able to adapt to the channel variation for ISI-free transmission. The adaptive equaliser needs to be trained by a known fixed length of training sequence bits for a minimum BER.

Facts to Know!

Generally, an adaptive channel equaliser should be used to improve transmission accuracy over a time-dispersive wireless channel. It will be well effective up to a coherence time due to time-invariant nature of the channel.

The equalisers require periodic training bits, followed immediately by the uncoded or encoded user data bits, to combat ISI effectively. For Rayleigh channels, it may be necessary to include a new training sequence with every block of data. It implies that equalisers must track the time-varying nature of the mobile fading channel. This increases considerable overhead but is justified by the amount of error rates encountered in a wireless mobile environment. For example, the GSM standard recommends using equalisation with 26-bit training sequence in each transmitted packet. The adaptive equaliser at the receiver utilises a recursive algorithm to evaluate the channel and estimate the filter coefficients to compensate for ISI caused by the multipath in the wireless channel.

There are two distinct classes of equalisers, each with a different overall structure, based on how the output of an adaptive equaliser is used for subsequent feedback: linear equalisers and nonlinear equalisers. A *linear equaliser* is of the simplest type and can be implemented as an FIR filter. In this, the present and the past values of the received signal are linearly weighted by the filter coefficient and summed up to produce the output. The linear equaliser can be implemented either as the simple transversal filter or as a complicated lattice filter. Linear equalisers increase the noise present in near vicinity of the spectrum, so they are not very effective on channels having too severe distortion.

Depending upon type of algorithms used for decoding the data, equalisers can be categorised as adaptive linear equalisers and maximum-likelihood sequence estimation (MLSE) equalisers. Linear equalisers and decision-feedback equalisers are adaptive equalisers that use an adaptive algorithm when operating. An MLSE equaliser uses the *Viterbi algorithm*.

7.7.2 Adaptive Linear Equalisers

In a mobile cellular environment, the characteristics of the wireless dispersive fading channel experience rapid changes randomly with time. For an equaliser to effectively combat ISI, the equaliser coefficients should also change in accordance with the channel characteristics so as to track the channel variations. Such an equaliser is called an adaptive equaliser since it adapts to the channel variations. In practice, a deterministic pseudorandom training sequence is transmitted before the information data to compute the initial optimum tap coefficients of the adaptive equaliser. This sequence is known to the receiver, and is used to adjust its tap coefficients to the optimum value. After the training sequence, the adaptive equaliser then uses the previously detected information data symbols to update the tap coefficients using appropriate algorithm and to estimate the equalisation error.

Facts to Know!

Another class of adaptive equalisers, known as blind equalisers or self-adaptive equalisers, operate only in the tracking mode and with a slower convergence rate than adaptive equalisers and thus does not require a training sequence.

Symbol-spaced equalisers and fractionally spaced equalisers are examples of linear equalisers. For each of the adaptive equaliser classes, there are a number of adaptive algorithms such as least mean square (LMS), signed LMS including signed regressor LMS and sign-sign LMS, normalised LMS, variable-*step*-size LMS, recursive least squares (RLS), and constant modulus algorithm (CMA). The adaptive algorithm also lets to specify certain properties of the algorithm.

A Symbol-Spaced Linear Equaliser (SSE) consists of a tapped delay line that stores samples from the input signal. Once per symbol period, the equaliser outputs a weighted sum of the values in

Facts to Know!

The LMS equaliser is more robust, and maximises the signal-to-distortion ratio to its output within the constraints of the equaliser time span and the delay through the equaliser. The equaliser coefficients are chosen to minimise the mean squared error, that is, the sum of squares of all the ISI terms plus the noise power at the output of the equaliser.

the delay line and updates the weights to prepare for the next symbol period. This class of equaliser is called symbol-spaced' because the sample rates of the input and output are equal. The algorithms for the weight setting and error-calculation blocks are determined by the adaptive algorithm chosen. The new set of weights depends on the current set of weights, the input signal, the output signal, and for adaptive algorithms other than constant modulus algorithm, a reference signal whose characteristics depend on the operation mode of the equaliser. In typical applications, the equaliser begins in training mode to gather information about the channel, and later switches to decision-directed mode.

A Fractionally Spaced Equaliser (FSE) is a linear equaliser that is similar to a symbol-spaced linear equaliser. By contrast, however, a fractionally spaced equaliser receives input samples before it produces one output sample and updates the weights, where is an integer. In many applications, is 2. The output sample rate is $1/T_b$, while the input sample rate is Z/T_b, where T_b is the bit duration. The weight-updating occurs at the output rate, which is the slower rate. Figure 7.10 illustrates a general approach to implement an equaliser using a linear equaliser circuit.

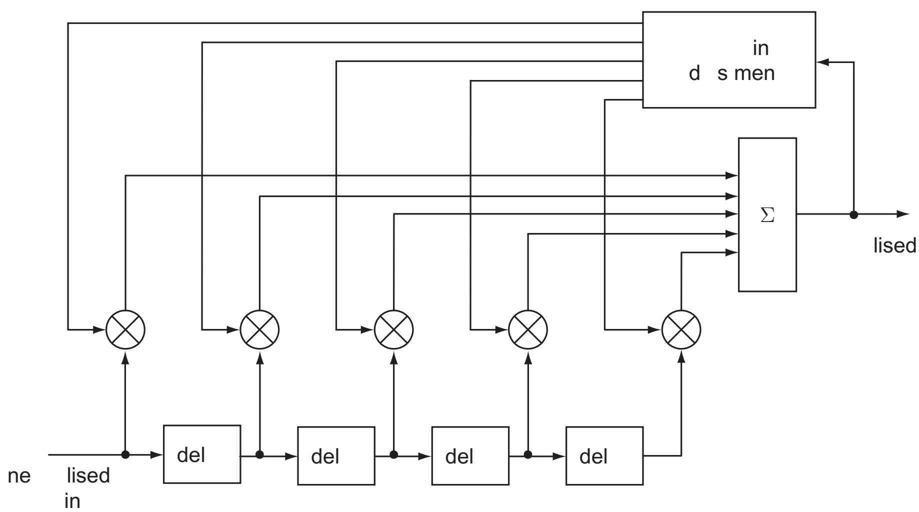

Fig. 7.10 | *Linear equaliser schematic*

A Decision-Feedback Equaliser (DFE) is a nonlinear equaliser that contains a forward filter and a feedback filter. The forward filter is similar to the linear equaliser of symbol-spaced equalisers, while the feedback filter contains a tapped delay line whose inputs are the decisions made on the equalised signal. The purpose of a DFE is to cancel intersymbol interference while minimising noise enhancement. By contrast, noise enhancement is a typical problem with the linear equalisers described earlier. Figure 7.11 depicts a typical structure of DFE which comprises of two filters, referred to as the forward and the feedback equalisers.

The received signal is the input to the forward equaliser. The input to the feedback equaliser is the stream of the detected symbols. The tap gains of this section are the estimates of the channel-sampled impulse response, including the forward equaliser. Due to past samples, this section cancels the ISI. *Decision-directed mode* means that the equaliser uses a detected version of its output signal when adapting the weights. Adaptive equalisers typically start with a training sequence and switch to decision-directed mode after exhausting all symbols in the training sequence. CMA equalisers are an exception, using neither training mode nor decision-directed mode. For non-CMA equalisers, the equalise' function operates in decision-directed mode when the syntax does not include a training sequence, and the equaliser has exhausted all symbols in the training sequence and still has more input symbols to process.

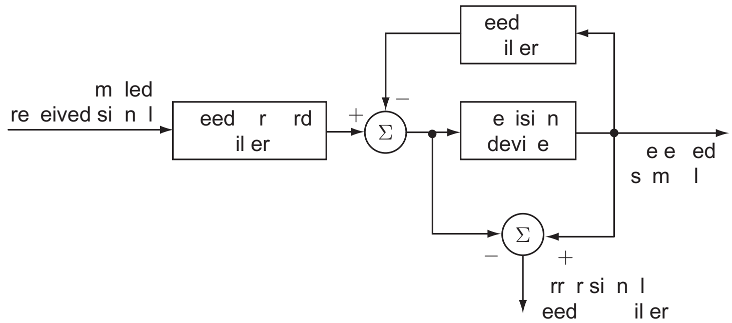

Fig. 7.11 *Functional schematic of DFE*

For proper equalisation using adaptive algorithms other than CMA, the reference tap should be set in such a way so that it exceeds the delay, in symbols, between the transmitter's modulator output and the equaliser input. Because the channel delay is typically unknown, a common practice is to set the reference tap to the centre tap in a linear equaliser, or the centre tap of forward filter in a decision-feedback equaliser. A delay can be taken into account by padding or truncating data appropriately. The DFE is particularly useful for channels with severe amplitude distortions and has been widely used in wireless communications.

A DFE yields better performance in the presence of moderate to severe ISI as experienced in a wireless channel. Given the same number of taps, whether a DFE can perform better than a linear equaliser or not in combating ISI depends on the channel impulse response. To have an idea about the extent of improvement with DFE, Fig. 7.12 gives a comparison of BER performance as a function of E_b/N_o, using accurate equalisation error in the adaptive algorithms.

Here, a 2-path wireless propagation channel is considered where the first path experiences Rician fading and the second path experiences Rayleigh fading. BPSK with coherent demodulation is used with perfect carrier-phase synchronisation. The adaptive linear equaliser uses a 7-tap whereas an adaptive DFE uses a 3-tap feedforward filter and a 2-tap feedback filter. It is observed that both the linear equaliser and the DFE improve the transmission performance over that with no equalisation. The DFE performs much better than the linear equaliser due to the limited number of taps in the linear equaliser and the noise enhancement effect at low E_b/N_o values.

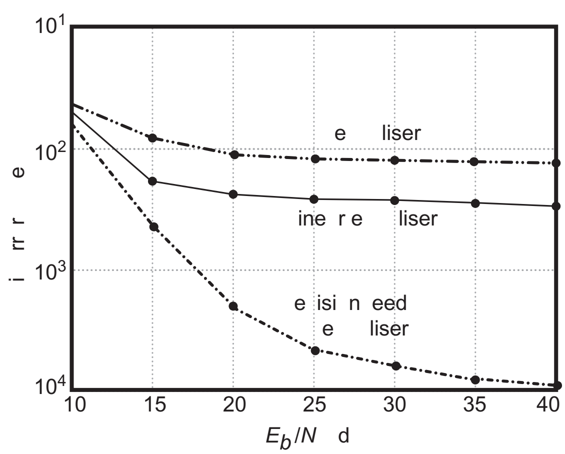

Fig. 7.12 *Comparison of BER performance with and without equalisation*

Facts to Know!

For coherent demodulation, the carrier frequency and phase of the received modulated signal must be precisely established.

7.7.3 MLSE Equaliser

Theoretically, an MLSE equaliser yields the best possible performance but is computationally intensive. In this equaliser, the training sequence is used to estimate the channel multipath characteristics and then use the estimate of the channel to estimate the effects of ISI. Given the estimates of the discrete channel impulse response, an MLSE receiver uses a trellis diagram with the Viterbi algorithm to obtain maximum-likelihood estimates of the transmitted symbols. It receives a baseband linearly modulated input signal and outputs the maximum likelihood sequence estimate of the signal, using an estimate of the channel modeled as a finite input response (FIR) filter. The equaliser decodes the received signal firstly by applying the FIR filter, corresponding to the channel estimate, to the symbols in the input signal. Then it uses the Viterbi algorithm to compute the trace-back paths and the state metric, which are the numbers assigned to the symbols at each step of the Viterbi algorithm. The metrics are based on Euclidean distance. The output is the maximum-likelihood sequence estimate of the signal, as a sequence of complex numbers corresponding to the constellation points of the modulated signal.

A functional block schematic of the adaptive MLSE receiver is shown in Fig. 7.13. It consists of two main parts—the adaptive channel estimator and the MLSE algorithm.

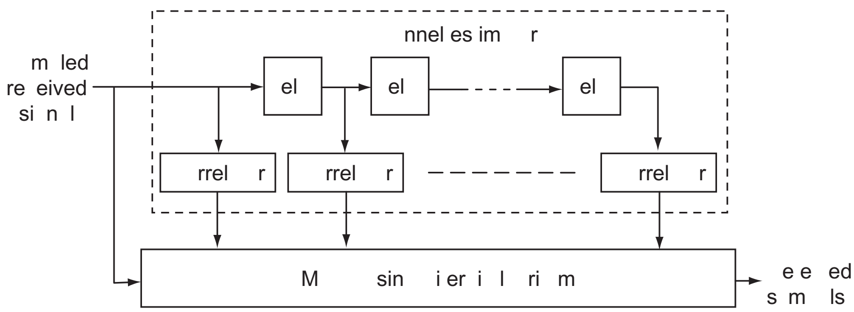

Fig. 7.13 *Functional schematic of adaptive MLSE receiver*

The sampled impulse response of the channel, taken at symbol intervals, is measured with the adaptive channel estimator. It is then compared with the sequence of the sampled received signal with all possible received sequences and determines the most likely transmitted sequence of symbols. The MLSE is the optimal method of canceling the ISI. However, the complexity of MLSE receiver grows exponentially with the length of the channel impulse response.

7.8 DIVERSITY

The wireless channel exhibits frequency-selective fading in high data rate transmission when the signal bandwidth is larger than the channel coherence bandwidth. To combat the effect of multipath fading, diversity techniques are used. Diversity improves transmission performance by making use of more than one independently faded version of the transmitted signal with comparable signal strengths. When the instantaneous SNR is low during a severe fading period producing more transmission errors, the diversity improves transmission accuracy considerably.

7.8.1 Diversity Techniques

Diversity techniques are widely used to improve wireless link performance and received signal quality. In wireless systems, the most effective method of counteracting the effects of channel fading is to use diversity schemes in the transmission and reception of the signal. Diversity is based on the fact that individual channels experience independent fading patterns. The short-term multipath fading can severely reduce transmission accuracy. For example, to achieve a BER of 10^{-4} for BPSK with coherent demodulation over a flat Rayleigh fading channel, the required average received SNR per bit is approximately 26 dB higher than that required for an AWGN channel. The degradation of transmission quality due to channel fading cannot be simply overcome by increasing the transmitted signal power, because in a severe fading channel, the instantaneously received SNR per bit can still be very low, resulting in a high probability of transmission error. In wireless mobile communications, the power available on the reverse link is very much constrained by limited battery capacity in hand-held subscriber units. With the use of diversity, the probability that all the received signals are in a fade at the same time reduces considerably, yielding a significant reduction in the average error rate.

The errors introduced by multipath fading can be compensated by providing multiple logical channels between the transmitter and receiver in some sense and sending a part of the signal over each logical channel. In other words, if one radio signal path experiences a deep fade, another radio signal path may have a strong signal. In this way, the transmission is spread out to avoid being subjected to possible errors, thereby reducing errors if not eliminating them completely. Diversity has provisions to select the best radio signal path; the improvement in instantaneous and average SNR at the receiver could be as high as 20 dB to 30 dB. Diversity exploits the random behaviour of radio propagation by finding highly uncorrelated signal paths for communication. Since diversity is implemented at the receiver without any involvement of the transmitter, it does not require any overhead as in case of training bits overhead in equalisation.

> **Facts to Know!**
>
> Cellular communication systems are mostly interference limited, and mitigation of channel fading by diversity reception results into improved interference tolerance and system capacity.

There are various ways of obtaining independently faded signals, resulting in various types of diversity mechanisms such as frequency diversity, time diversity, space or antenna diversity, angle diversity, path diversity, and polarisation diversity.

- In ***eq e i e it***, the desired message is transmitted over several frequency slots having separation between adjacent frequency slots larger than the channel coherence bandwidth. It improves wireless-link transmission quality using additional bandwidth. Frequency diversity is often employed in line-of-sight microwave communications which carry several channels in FDM mode. It is also used in OFDM modulation and access techniques by providing simultaneous modulation signals with error-control coding across a large bandwidth.
- In ***ti e i e it***, the desired message is transmitted repeatedly over several time periods having separation between adjacent time slots larger than the channel coherence time. It is more frequently exploited through interleaving, FEC, and ARQ techniques, although it introduces signal-processing delay. Time diversity is implemented in a Rake receiver for spread spectrum CDMA systems where the multiple channel provides redundancy in the transmitted message.
- ***Sp e i e it***, also known as ***te i e it***, is one of the most popular forms of receiver diversity schemes widely used in wireless systems. In space diversity, the desired message is transmitted by using multiple *Tx* and/or *Rx* antennas, having physical separation between adjacent antennas as multiples of operating wavelength. The signals received from horizontally spatially separated multiple receiver antennas at the base station (usually of the order of 10λ), and the mobile unit provide diversity reception. It does not require additional system resources; however, it incurs cost of additional antennas.

> **Facts to Know!**
>
> Forward Error Correction (FEC) coding can be regarded as a certain kind of time diversity. Although it introduces some redundancy into the transmitted data stream and requires an increase of the transmission bandwidth, yet it is an effective means of mitigating the multipath fading effect.

- In *e i e it*, the desired message is received simultaneously by several directional antennas installed in widely different directions in order to receive all the nonoverlapped uncorrelated scattered signal paths. It can be viewed as a special case of space diversity since it also requires multiple antennas.
- In *p t i e it*, a Rake receiver can separate the received signal components from different propagation paths by using code correlation and then combining the signal components constructively as in DSSS based CDMA cellular systems. It reduces the transmitted power and increases the system capacity; however at the cost of increased complexity.
- In *p i ti i e it*, horizontally as well vertically polarised signals or left- and right- circular polarised waves are transmitted simultaneously. Since these are uncorrelated in the mobile radio path, one of these signals will provide strong received level after fading. Polarisation diversity is used in line-of-sight microwave communication links to support two simultaneous users on the same radio channel, thereby improving link margin and system capacity.

Facts to Know!

In principle, there exists no limit to the number of diversity branches except in case of polarisation diversity. For example, several practical wireless applications in the 2.4 GHz band have used up to 5 receiver antennas to achieve space diversity.

7.8.2 Macroscopic and Microscopic Diversity

The mobile units are capable of receiving signals from more than one serving base stations at the same time, the base stations being separated in space by large distance, as in a CDMA cellular system. Due to variations in the terrain conditions and the nature of the surroundings, signals from different base stations may experience large-scale fading caused by shadowing effect. The received signal strength at a mobile can drop below that of free space conditions in deeply shadowed conditions. The average SNR value can be improved by the mobile on the forward link considerably by selecting a base station which is less shadowed than other base stations. Similarly, the base-station receiver can use macroscopic diversity in the reverse link by employing a number of receiving antennas sufficiently separated in space.

Microscopic diversity is similar to space diversity, or antenna diversity, in which more than one receiver antennas, horizontally separated by $10\lambda - 20\lambda$ in space, receive the rapidly changing signal due to small-signal fading in Rayleigh fading wireless channel and select the best signal to mitigate the effects of small-signal fading.

7.8.3 Space-Diversity Combining Techniques

The use of diversity at receiver needs combining the outputs of statistically independent fading channels in accordance with a criterion that leads to improved receiver performance. It is assumed that the wireless communication channel is described by a frequency-flat, slow-fading Rayleigh channel. It implies that all the frequency components constituting the transmitted signal are characterised by the same random attenuation and phase shift. The fading remains essentially unchanged during the transmission of each symbol and the fading phenomenon is decribed as Rayleigh distribution. A variety of combining techniques are available for reception of the space or antenna diversity signals. Fig. 7.14 shows an equivalent model of a diversity system with coherent demodulation and M-th order diversity.

There are a number of techniques for combining statistically independent faded signal components available at the output of the coherent demodulators for transmitted symbol detection by the decision device. The diversity branches are summed up linearly in the linear combining method. Various linear combining techniques differ in the extent of transmission performance improvement versus receiver complexity. Linear combining techniques include selective combining, equal gain, and maximal ratio combining.

Selective-Diversity Combining Technique With selective diversity, one best signal is chosen based on the received signal strengths from the set of diversity branches. The receiver monitors the SNR value of each

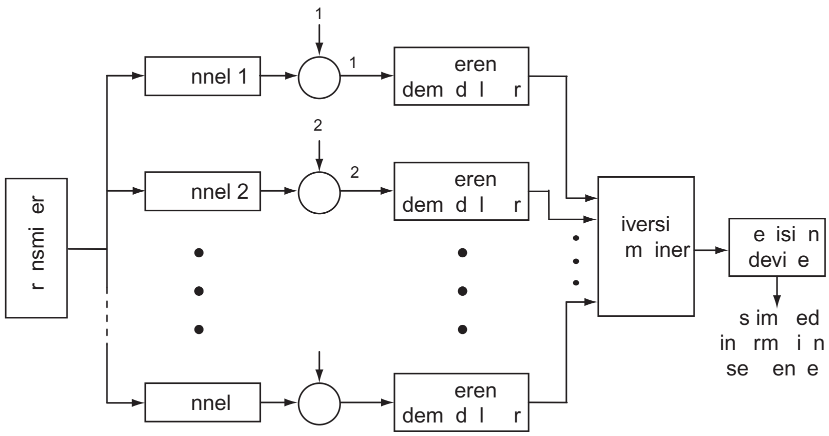

Fig. 7.14 | *A model of diversity system with coherent demodulation*

diversity branch and selects the one with the maximum SNR value for signal detection. It is much easier to implement without much performance degradation. It is generally employed for the reverse link transmission where the diversity branches can be physically located in different base stations. Figure 7.15 depicts a simple block diagram of the selective-diversity combining technique.

Consider M number of independent fading signals received by multiple receiver antennas. There are M-branch receivers comprising of coherent demodulators. The output of demodulators is presented to a logic circuit which selects the particular branch receiver output having the largest SNR value of the received signal. Conceptually, selection diversity combining is the simplest form of 'space diversity on receive' technique. The selection-combining procedure requires that the receiver outputs are monitored continuously and, at each instant of time, the receiver with the largest instantaneous SNR value is selected.

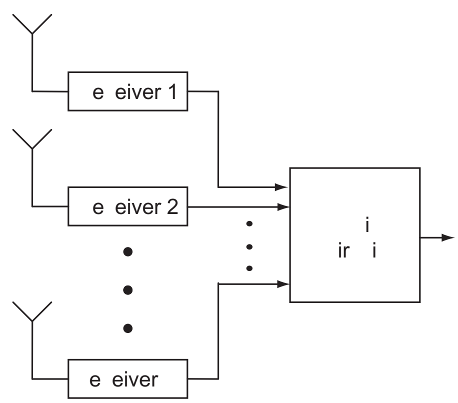

Fig. 7.15 | *Selective diversity combining technique*

From a practical implementation point of view, such a selective combining procedure is cumbersome. There is an alternate method by adopting a scanning version of the selective combining procedure. Firstly, the receiver with the strongest output signal is selected and maintained till its instantaneous SNR value does not drop below a pre-defined threshold SNR value. Then, a new receiver that offers the strongest output signal is selected and the selection procedure is repeated. This method has an advantage that it requires only one receiver and is very simple to implement.

The performance of the selective combining technique is not optimum because it ignores the information available from all the diversity branches except for the particular selected receiver branch that produces the instantaneous SNR value greater than the pre-defined threshold SNR value. The probability, $P_M(S_i)$, that all independent diversity branches receive signals which are simultaneously less than specified threshold SNR, S_i, is given by

$$P_M(S_i) = (1 - e^{-Si/Sr})^M \tag{7.38}$$

where S_r is the average received SNR value, and M is the number of independent diversity branches in selective diversity combining technique.

It can be seen that selective diversity can greatly improve the BER performance. The performance improvement is more significant when M is increased from 1 to 2 than it is when M is further increased from 2 to 4, and to 8. Comparing the BER for an AWGN channel with that for the fading channel without diversity, the transmission performance is severely degraded by the Rayleigh fading. Coherent BPSK performs better than differential BPSK in AWGN channel. Since DBPSK and π/4-DQPSK does not require accurate carrier phase synchronization and is robust to carrier phase distortion introduced by the fading channel as long as the phase distortion does not change much over a time duration of two symbol intervals.

Facts to Know!

For the coherently demodulated receiver, there is no difference whether the combining is carried out in the pre-detection or in the post-detection stage.

Maximal-Ratio Diversity Combining Technique. It is considered to be the optimum technique of combining in which the diversity branches are weighted prior to summing them, each weight being proportional to the signal strength of the received branch. This technique assumes that the receiver is able to accurately estimate the amplitude fading and carrier-phase distortion for each diversity channel. With knowledge of the complex channel gains, the receiver coherently demodulates the received signal from each branch. After removing the phase distortion, the coherently detected signal is then weighted by the corresponding amplitude gain. The weighted received signals from all the M branches are then summed together and applied to the decision device. Figure 7.16 describes the block diagram of a maximal-ratio combiner.

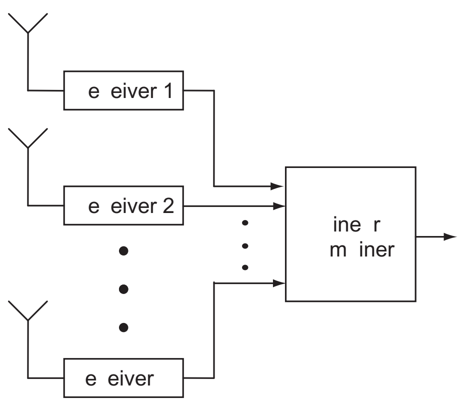

Fig. 7.16 *Maximal-ratio diversity combining technique*

Consider the transmission of a digital modulated signal over flat slow Rayleigh fading channels using coherent demodulation with Mth order diversity. It is assumed that the channel-fading processes and the additive white Gaussian noise processes are mutually statistically independent, and these are independent of each other. For a slow fading channel, the complex gain can be assumed to be a complex constant over each symbol interval. The demodulator in each channel is optimum for an AWGN channel using filters matched to the orthonormal functions.

Theoretically, the maximal-ratio combiner is the optimum among linear diversity combining techniques in the context that it produces the largest possible value of instantaneous SNR value. In fact, the instantaneous output SNR value of the maximal-ratio combiner can be large even when the SNR values of the individual branches are small. Practically, it is difficult to achieve the exact setting and significant instrumentation is needed to adjust the complex weighting parameters to their exact values. The BER transmission performance improvement is achieved by diversity with a maximal-ratio combiner for coherent BPSK, QPSK, and MSK over Rayleigh fading channels. In practice, the performance difference between DBPSK with selective diversity and coherent BPSK with a maximal ratio combiner may not be quite significant. Diversity can significantly improve the transmission performance. As the number of branches increases, the BER performance approaches to that of an AWGN channel.

Equal-Gain Diversity Combining Technique In this, all the signals are weighted equally after coherent demodulation which removes the phase distortion. All the weighting parameters have their phase angles set opposite to those of their respective multipath branches, and their magnitudes are set equal to some constant value. The coherently detected signals from all the M branches are simply added and applied to the decision device. As the receiver does not need to estimate the amplitude fading, the receiver design is not complex. Equal-gain diversity combining also relies on the ability to estimate the phase of the different diversity branches and to combine the signals coherently. Due to hardware limitations or physical separation of the diversity receivers, it is difficult to implement it practically. The performance of an equal-gain combiner is only marginally inferior to a maximal-ratio combiner and superior to a selection diversity combiner. Among the three linear combining techniques, maximal-ratio combining offers the best performance, followed by equal gain combining.

Square-Law Nonlinear Space Diversity Combining Technique Square-law combining offers the opportunity to obtain an advantage in diversity without requiring phase estimation. Unlike maximum-ratio combining, square-law combining is applicable to certain modulation schemes such as orthogonal modulations including direct-sequence CDMA signals or FSK in which different but approximately orthogonal sequences or frequencies are used to represent different data symbols.

Facts to Know!

For VHF, UHF, and microwave wireless/mobile radio communication applications, both the maximal-ratio and the equal-gain combining methods are not suitable due to a rapidly changing, random-phase, multipath fading environment. The selection method is more suitable for mobile radio application because of its simple implementation.

7.9 CHANNEL CODING

Channel coding is effective in combating independent random transmission errors over a noisy channel. Channel coding adds redundancy information to the information data at the transmitter in a deterministic manner, following some logical relation with the original information. The receiver receives the encoded data with transmission errors. At the receiver, the original information can be decoded with the controlled redundancy based on the same logical relationship between the original information and redundant information. The redundancy causes channel coding to consume additional frequency bandwidth during transmission and may seem to be a waste of system resources. However, if an error-correction code is designed properly using the frequency bandwidth and transmission power, the coded sequences can be transmitted at a faster rate.

When the transmission rate for the information bits remain the same as in the uncoded system, the transmission accuracy for the coded system is higher. This results in the coding gain which translates to higher transmission accuracy, higher power and spectral efficiency. In cellular systems, the traffic consists of compressed data and is very sensitive to transmission errors. Therefore, channel coding can be defined as the processing of coding discrete digital information in a form suitable for transmission with an objective of enhanced reliability. Channel coding is applied to ensure adequacy of transmission quality in terms of bit-error rate and frame-error rate. Thus, channel coding provides excellent BER performance at low SNR values at the expense of reduction in the bandwidth efficiency of the wireless link in high SNR conditions.

Facts to Know!

In typical applications, channel coding is used in mobile environments in which transmitter power is limited. In a cellular interference-limited environment increase of transmitter power may not lead to performance improvement. Channel coding improves system performance by increasing symbol energy.

To improve the reliability of a digital transmission system further without increasing the required bandwidth or transmit power, channel coding can be combined with trellis-coded modulation. This is achieved by combining convolution code with a higher-order modulation scheme at the transmitter and by combining the channel decoder and demodulator together at the receiver. The channel coding can also be combined with an antenna diversity scheme, referred to space-time coded modulation, which is a bandwidth as well as power efficient method for wireless communications. Using multiple *Tx* and/or *Rx* antennas, and based on the channel state information, the spatial properties of the state-time codes can ensure that the diversity is used at the transmitter while maintaining optional receiver diversity which is particularly important for limited battery-operated mobile phones.

Typically, a code may have a larger or at least an equal error-detection capability than the error-correction capability. However, in wireless communications, if the error can only be detected and not corrected, then the transmission is not successful. In such case, techniques such as retransmission and ARQ need to be employed. In multilevel modulation schemes, it is possible to encode symbols. Bose–Chaudhuri–Hocquenghem (BCH) codes are a popular class of nonbinary cyclic block codes. Reed-Solomon (RS) codes are a subset of the BCH codes and are well suited for burst-error correction. RS codes in association with efficient coding techniques make highly efficient use of redundancy, and symbol sizes and block lengths which can be easily adjusted to accommodate a wide range of information data sizes. The three most commonly used channel coding or error-control (error correction and detection) coding techniques are block codes, convolutional codes, and Turbo codes. Block codes and convolutional codes are part of forward error correction (FEC) type of channel-coding techniques. Turbo code is a new class of block codes.

7.9.1 Block Codes

Block coding involves encoding a block of source information bits into another block of bits with addition of some redundancy bits to combat channel errors induced during wireless transmission. Error-control coding techniques detect and possibly correct errors that occur when messages are transmitted in a wireless communication system. To accomplish this, the encoder transmits not only the information symbols, but also one or more redundant symbols. The decoder uses the redundant symbols to detect and possibly correct whatever errors occurred during transmission. For a given data rate, error-control coding can reduce the probability of error, or reduce the required SNR to achieve a desired probability of error at the expense of transmission bandwidth or decoder complexity.

To generate an (n, k) block code, the channel encoder accepts information data in successive k-bit blocks and adds (n k) redundant bits to each block that are algebraically related to the k message bits, where $k < n$, thereby producing an overall encoded block of n bits. The data rate at which the block encoder produces bits is given by

$$R_o = (n/k)\, R_s \tag{7.39}$$

where R_o is the channel data rate at the output of block encoder; and

R_s is the source-information data rate.

The code rate is defined as the ratio of source information data rate, k bps and channel data rate, n bps, that is

$$\text{Code rate} \quad = \quad / \tag{7.40}$$

The code rate is a dimensionless ratio. The ability of a block code to correct errors is a function of the code distance, defined as the difference in the number of elements between two code words. In case of a binary code, the code distance is known as the *Hamming distance*.

EXAMPLE 7.10 | Code rate for GSM control channel

In GSM cellular communication system, a data block of 184 bits is encoded into 224 bits of code word on the control channel before sending it to a convolution encoder. Determine the number of parity check bits added and the code rate of the block encoder used.

Solution

Number of information bits, $k = 184$ bits (given)

Number of encoded bits, $n = 224$ bits (given)

Step 1: *To determine the number of parity check bits added*

Number of parity check bits $= n - k$

$= 224 - 184 =$ **40 bits**

Step 2: *To determine the code rate of the block encoder used*

The code rate of block encoder, $r = k/n$

$= 184 / 224 =$ **0.82**

Block codes use algebraic techniques properties to encode and decode blocks of data bits or symbols. The code words are generated in a systematic form in such a way that the original k number of source information data bits are retained as it is and the $(n \quad k)$ parity check bits or redundant bits are either appended or prepended to information data bits. A simple operation of a block code is depicted in Fig. 7.17.

Mostly, operation in a block encoder are based on linear feedback shift registers that are easy to implement and inexpensive. The parity check bits are generated using a generator polynomial or matrix. Codes generated by a polynomial are called *cyclic codes*, and the resultant block codes are called *cyclic redundancy check* (CRC) codes. The code word comprising of n bits of encoded block of data is a transmitter over the channel. The received code word may be error-free or modified due to channel error. Sometimes it may result in another valid code word, which cannot be detected as error, with a probability given by

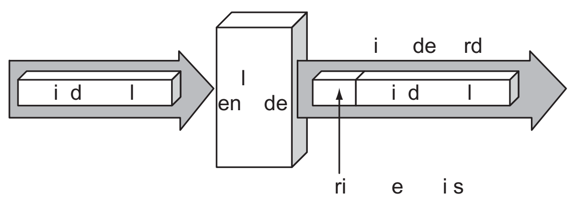

Fig. 7.17 *A simple block code operation*

$$P_D \leq 2^{-(n-k)} \tag{7.41}$$

where P_D is termed as probability of false detection. It can be minimised by designing block codes in such a way so that different code words have a large code distance.

Facts to Know!

The main reason that cyclic codes are of major importance is that encoding and decoding can be implemented using simple shift-register and a small amount of additional logic.

Block codes have certain limitations such as the following:

– Block codes have limited ability to handle large numbers of distributed errors in a wireless channel.
– Block codes use hard decisions that tend to destroy information. They do not achieve the performance as obtained through the use of soft decisions.
– The block encoder accepts a k-bit information data block and generates an n-bit code word. Thus, code words are generated on a block-by-block data basis. Clearly, a provision must be made in the block encoder to buffer an entire information data block before generating the associated code word.

7.9.2 Convolutional Codes

Error-control coding deals with the techniques used to enhance digital signals so that they are less vulnerable to channel impairments such as noise, fading and jamming. Convolutional coding is a special case of error-control coding. There are applications in which the information bits come in serially rather in large blocks, in which case the use of a buffer as in case of a block encoder may be undesirable. In such situations, the use of convolution coding which generates redundant bits by using modulo-2 convolutions may be a preferred technique. A convolutional encoder is not a memoryless device. Even though a convolutional coder accepts a fixed number of message symbols and produces a fixed number of code symbols, its computations depend not only on the current set of input symbols but on some of the previous input symbols. A channel that exhibits multipath fading is an example of a channel with memory. Most convolutional codes are designed to combat random independent errors which cause degradation in error performance over a wireless channel having memory.

Facts to Know!

The convolutional encoder is a different type of encoder, in which the information symbols are not grouped together in blocks for encoding.

Unlike block code technique, the convolutional-coding technique does not map individual blocks of input information data bits into blocks of code words. Convolution encoders accept a continuous stream of data bits and map them into an output data stream adding redundancies in the convolution process. The redundancies depend on not only the current k bits but also several of the preceding k bits. Thus, the code rate still remains the same, that is, k/n. The number of the preceding bits is called the constraint length m that is similar to the memory in the convolution encoder.

Generally, convolution coders are more powerful than block codes in terms of providing FEC, but are not useful for detection or ARQ schemes. At the receiver, FEC is performed using a maximum-likelihood decoding algorithm that determines what data sequence would have been most likely transmitted, given the received sequence of bits. VLSI implementation of the Viterbi algorithm is the most common algorithm used for this purpose.

Facts to Know!

An important parameter in the design of convolutional encoding is the constraint length, which is defined as the number of output symbols that are affected by a given input symbol.

A convolutional encoder can assume one of the following two forms:

Recursive Systematic The encoder uses feedforward as well as feedback paths and the k-tuple of information data bits appears as a subset of the n-tuple of output bits. This scheme is used in Turbo codes also.

Non-Recursive Non-Systematic The encoder distinguishes itself by the use of feed forward paths only and the information bits lose their distinct identity as a result of the convolution process.

Feed forward or feedback convolutional codes can be described by a trellis structure or a set of generator polynomials. It uses the Viterbi algorithm to implement hard-decision and soft-decision decoding. A polynomial description of a convolutional encoder describes the connections among shift registers and modulo-2 adders. For example, Fig. 7.18 depicts a feedforward convolutional encoder that has one input, two outputs, and two shift registers.

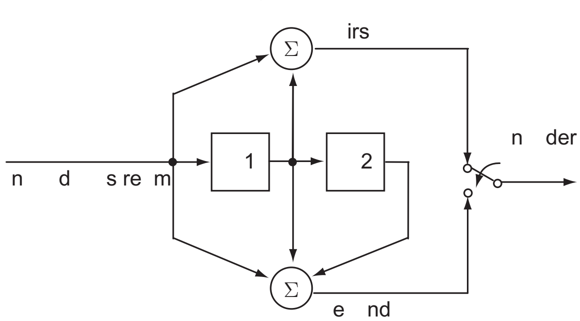

Fig. 7.18 *A feed forward convolutional encoder*

A binary convolutional encoder may be viewed as a finite-state machine that consists of m-stage shift register with prescribed connections to modulo-2 adders and a multiplexer that serialises the outputs of the adders. A k-bit information data sequence produces a coded output sequence of length $n\,(k + m)$ bits. Therefore,

$$\text{Code rate of the convolutional code, } r = k / n\,(k + m) \tag{7.42}$$

Typically, $k >> m$. Hence, code rate reduces to $1/n$ bits/symbol.

A polynomial description of a convolutional encoder has either two or three components, depending on whether the encoder is a feedforward or feedback type:

– Constraint lengths
– Generator polynomials
– Feedback connection polynomials (for feedback encoders only)

Constraint Lengths The constraint length of a convolutional code is defined as the number of shifts over which a single information bit can influence the encoder output. It is expressed in terms of information bits. The constraint lengths of the encoder form a vector whose length is the number of inputs in the encoder diagram. The elements of this vector indicate the number of bits stored in each shift register, including the current input bits. In an encoder with an m-stage shift register, the memory of the encoder equals m information bits and $k = m+1$ shifts are required for an information bit to enter the shift register and finally come out. Hence, the constraint length of the encoder is k. In the above figure, the constraint length is three, $n = 2$, and code rate . It is a scalar because the encoder has one input stream, and its value is one plus the number of shift registers for that input.

Generator Polynomials If the encoder diagram has k inputs and n outputs, then the code generator matrix is a k-by-n matrix. The element in the ith row and jth column indicates how the ith input contributes to the jth output. For systematic bits of a systematic feedback encoder, match the entry in the code generator matrix with the corresponding element of the feedback connection vector.

Feedback Connection Polynomials If a feedback encoder is required to be represented, then a vector of feedback connection polynomials is needed. The length of this vector is the number of inputs in the encoder diagram. The elements of this vector indicate the feedback connection for each input, using an octal format.

Trellis Description of a Convolutional Encoder A trellis description of a convolutional encoder shows how each possible input to the encoder influences both the output and the state transitions of the encoder. Figure 7.19 depicts a trellis for the convolutional encoder shown in Fig. 7.18. The encoder has four states (numbered in binary from 00 to 11), a one-bit input, and a two-bit output. (The ratio of input bits to output bits makes this encoder a rate-1/2 encoder.) Each solid arrow shows how the encoder changes its state if the current input is zero, and each dashed arrow shows how the encoder changes its state if the current input is one. The octal numbers above each arrow indicate the current output of the encoder.

As an example of interpreting this trellis diagram, if the encoder is in the 10 state and receives an input of zero then it outputs the code symbol 3 and changes to the 01 state. If it is in the 10 state and receives an input of one, then it outputs the code symbol 0 and changes to the 11 state. Note that any polynomial description of a convolutional encoder is equivalent to some trellis description, although some trellises have no corresponding polynomial descriptions.

If the encoder has a feedback configuration and is also systematic then the code generator and feedback connection parameters corresponding to the systematic bits must have the same values. For example, Fig. 7.20 shows a rate 1/2 systematic encoder with feedback.

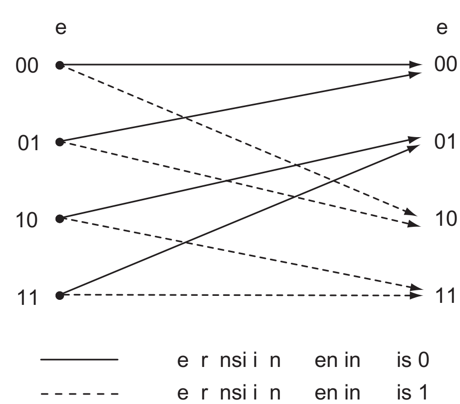

Fig. 7.19 *A trellis for the convolutional encoder*

This encoder has a constraint length of 5, a generator polynomial matrix of 37 33 , and a feedback connection polynomial of 37. The first-generator polynomial

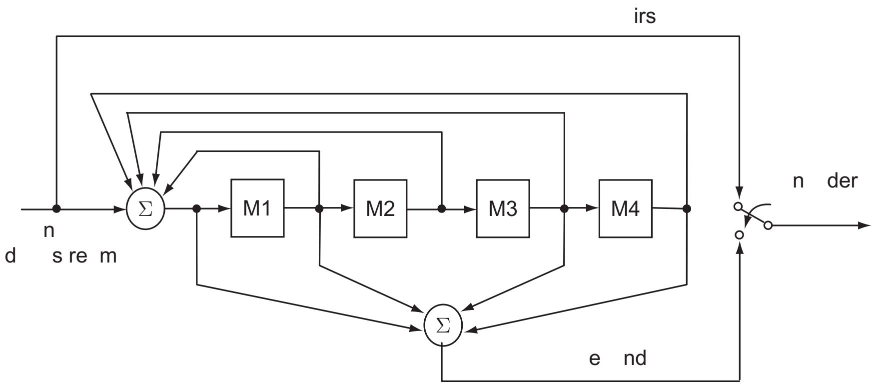

Fig. 7.20 *A rate 1/2 systematic encoder with feedback*

matches the feedback connection polynomial because the first output corresponds to the systematic bits. The feedback polynomial is represented by the binary vector [1 1 1 1 1], corresponding to the upper row of binary digits in the diagram. These digits indicate connections from the outputs of the registers to the adder. Note that the initial 1 corresponds to the input bit. The octal representation of the binary number 11111 is 37. The second generator polynomial is represented by the binary vector [1 1 0 1 1], corresponding to the lower row of binary digits in the diagram. The octal number corresponding to the binary number 11011 is 33.

Convolution coding is widely used in digital cellular systems. For example, GSM employs a rate convolution encoder. In GSM speech processing, a digitised voice of 260-bits block data is broken up into 182 class-I bits and 78 class-II bits. The most significant 50 bits of the class-I bits are enhanced with a block code that adds three parity bits. The sum of 182 class-I bits, three parity bits, and four tail bits, that is, (182+3+4) = 189) bits are passed through a rate convolution encoder to produce 378 bits. These are added to 78 class-II bits to produce 456 bits of encoded data. Similarly, in the IS-95 CDMA standard, a convolution encoder is employed in both the forward and reverse links, with a rate convolution encoder and a rate 1/3 convolution encoder respectively that has a constraint length of $m = 9$.

7.9.3 Turbo Codes

The operation of Turbo encoding is based on the use of a pair of encoders, separated by the interleaver, and iterative detection involving the use of feedback around a pair of decoders separated by a deinterleaver and an interleaver. Turbo codes provide significant improvements in the quality of data transmission over a noisy channel. The encoder uses a parallel FEC encoding scheme in which the information is systematically encoded by two separate identical encoders. Figure 7.21 illustrates the functional block diagram of a Turbo encoder.

The *pseudorandom interleaver,* also called *turbo interleaver*, enables the second encoder to reorder the information bits prior to encoding. The information bits and the parity bits generated by the two encoders are then multiplexed and punctured in a repeating pattern to increase the code rate for transmission. The function

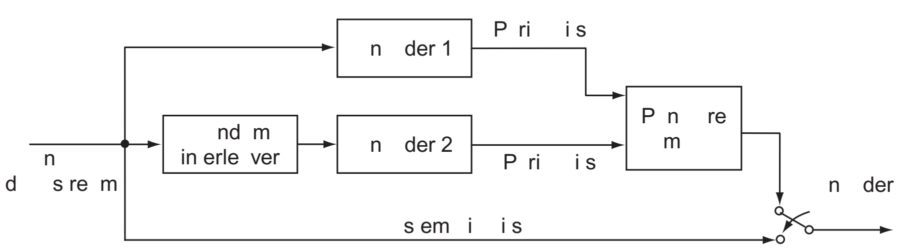

Fig. 7.21 *Block diagram of a Turbo encoder*

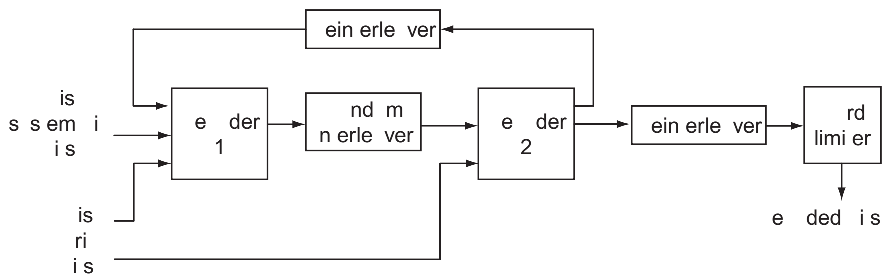

Fig. 7.22 *Block diagram of a Turbo decoder*

of a Turbo decoder is just the inverse of Turbo encoder functions. Figure 7.22 illustrates the functional block diagram of Turbo decoder.

The first decoder, the interleaver, the second decoder, and the deinterleaver constitute a single-hop feedback system. This arrangement makes it possible to iterate the decoding process in the receiver many times so as to achieve satisfactory performance. The inputs to the first decoder are the channel samples corresponding to the systematic information bits, the channel samples corresponding to the parity bits of the first encoder, and the extrinsic information about the systematic bits that were determined from the second decoder.

7.9.4 Comparison of Coding Techniques

- A Turbo code is closer to the random code because it uses a pseudorandom interleaver to separate its own two convolutional encoders.
- In higher code rates or low SNR value conditions, Turbo codes exhibit better performance than traditional convolutional codes.
- Convolutional codes and Turbo codes perform better with soft decisions. However, convolutional codes can work with hard decisions also.
- Convolutional codes do not have an error floor, whereas Turbo codes do have an error floor. It means in Turbo codes, the BER drops very quickly in the beginning, but eventually settles down and decreases at a much slower rate.
- Both convolutional codes and Turbo codes require the use of flush bits to initialise them to state 0 at the end of the incoming source information bits sequence. However, due to parallel encoding structure of Turbo codes, it is not straightforward to flush the second encoder.
- Turbo codes are decodable, and hence they are of more practical importance.

Turbo codes are inherently block codes with the block size determined by the size of the turbo interleaver. These codes are used in 3G cellular technology for high-speed-data rate applications.

7.10 INTERLEAVING

The error-correcting codes are generally capable of correcting individual data-bit errors, but not a burst error involving a group of adjacent data bits. However, in the wireless and mobile channel environment, the burst error occurs quite frequently. In order to correct the burst error, *i te e i* exploits time diversity without adding any overhead in wireless digital cellular communication systems such as GSM, IS-95 CDMA, 3G cellular systems. Interleavers disperse the burst error into multiple individual errors which can then be handled by an error-correcting code. Furthermore, interleaving does not have error-correcting capability. Therefore, interleaving is always used in conjunction with an error-correcting code. In other words, interleaving does not introduce any redundancy into the information sequence, so it does not add to extra bandwidth requirement. The disadvantage of interleaving is additional delay as the information data sequence needs to be processed block by block. Therefore, small memory size interleaving is preferred in delay-sensitive applications.

Facts to Know!

Interleaving is an effective method to combat error burst occurring on a wireless fading channel. Here the code word is spread, positioning the bits one away from another so that they experience independent fading.

Correlative channel fading usually creates bursty errors during fading periods, which reduces the forward error-correction capability of channel coding. Interleaving is usually followed by channel coding and is quite effective in combating bursty errors. The coded symbols are first interleaved before being mapped to modulated waveforms in the transmitter. At the receiver, the demodulator output symbols are then de-interleaved before being applied to the decoder. The interleaver length is kept sufficient enough so as to eliminate the negative effect of the channel-fading correlation on the coding gain. When the normalised channel fading rate is low, interleaving becomes unnecessary and adds extra processing delay and hardware memory complexity which may not be acceptable in real-time communications applications.

The basic function of an interleaver is to protect the transmitted data from burst errors. Due to the use of speech coders, many important bits are produced together. The interleaver spreads these bits out in time so that all these bits are not corrupted at the same time by deep fade or noise burst. There are many types of interleavers such as block interleaver, semi random and random interleaver, circular interleaver, odd-even interleaver, and near-optimal interleaver. Each one has its advantages and drawbacks in the context of noise. Block interleaver is the most commonly used interleaver in wireless communication systems.

Interleaving the coded message before transmission and deinterleaving after reception causes bursts of channel errors to be spread out in time. An interleaver permutes symbols according to a mapping. A corresponding deinterleaver uses the inverse mapping to restore the original sequence of symbols. Interleaving is a technique, which only requires knowledge of the duration of the channel memory, not its exact statistical characterisation.

A convolutional interleaver has memory; that is, its operation depends not only on current symbols but also on previous symbols. If a convolutional interleaver is used followed by a corresponding convolutional deinterleaver, then a nonzero delay means that the recovered data (output from the deinterleaver) is not the same as the original data (input to the interleaver). If the two data sets are compared directly then the delay must be taken into account by using appropriate truncating or padding operations.

7.10.1 Block Interleavers

A block interleaver accepts a set of symbols and rearranges them, without repeating or omitting any of the symbols in the set. The number of symbols in each set is fixed for a given interleaver. Table 7.5 summarises various types of block interleavers along with their functions. Each special-case interleaver function uses the same computational code that the general block interleaver function uses. Pseudorandom block interleavers are used in Turbo codes.

Table 7.5 *Types of block interleavers and their functions*

Block interleaver type	*Description*
General block interleaver	Uses the permutation table given explicitly as an input argument
Algebraic interleaver	Derives a permutation table algebraically
Helical scan interleaver	Fills a matrix with data row by row and then sends the matrix contents to the output in a helical fashion
Matrix interleaver	Fills a matrix with data row by row and then sends the matrix contents to the output column by column
Random interleaver	Chooses a permutation table randomly using the initial state input provided

EXAMPLE 7.11 | Block interleaving in GSM

In GSM, the output of the convolutional encoder consists of 456 bits for each input of 228 bits. How is block interleaving of encoded data implemented? Determine the delay in reconstructing the code words corresponding to the reception of 8 TDMA frames. Comment on the results obtained.

Solution

Step 1. *To find code rate of the convolutional encoder*

Number of data bits at the input of convolutional encoder = 228 bits (given)

Number of data bits at the output of convolutional encoder = 456 bits (given)

Hence, code rate of the convolutional encoder = 456/228 =

Step 2. *To find number of encoded data bits per frame*

Number of TDMA frames = 8 (given)

Number of encoded data bits in each TDMA frame = 456/8 = 57

Step 3. *Block interleaving procedure of encoded data*

Therefore, the encoded 456 data bits are split into 8 uniform blocks of 57 bits each. These 57 bits in each block are spread over eight TDMA frames so that even if one frame out of five is lost due to channel burst error, the voice quality is not affected.

Step 4. *To determine delay in reconstructing the code words*

One TDMA frame time duration = 4.6 ms

Time taken to transmit 8 TDMA frames = 8 × 4.6 = 36.8 ms

Therefore, the delay in reconstructing the code words corresponding to the reception of 8 TDMA frames is 36.8 ms.

Comment on the results Usually, a delay of 50 ms is acceptable for voice conversation. Hence, the delay introduced due to block interleaving in a GSM system does not degrade the voice quality performance while enhancing BER performance to combat channel errors.

7.10.2 Convolutional Interleaving

The sequence of encoded bits to be interleaved is arranged in blocks of L bits. For each block, the encoded bits are sequentially shifted into and out of a bank of N registers by means of two synchronised input and output commutators. Figure 7.23 shows the functional block diagram of a convolutional interleaver.

As shown in the figure, the zeroth shift register does not have any memory element to store bits. It implies that the incoming encoded symbol is transmitted immediately. Each successive shift register provides a memory capacity of L symbols more than the preceding shift register. Each shift register is scanned regularly

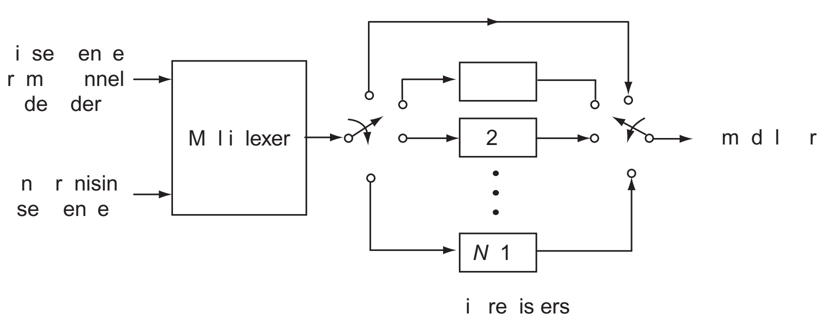

Fig. 7.23 | *Block diagram of a convolutional interleaver*

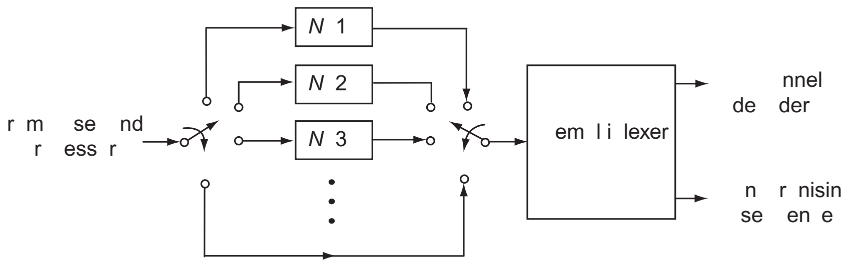

Fig. 7.24 | *Block diagram of a convolutional deinterleaver*

on a periodic basis. With each new encoded symbol, the commutators switch to a new shift register. The new symbol replaces the oldest symbol in that register. After completing with the last ($N-1$)th shift register, the commutators return to the zeroth shift register, repeating the whole process again and again. Figure 7.24 shows the functional block diagram of a convolutional deinterleaver.

The convolution deinterleaver in the receiver also uses N shift registers and a pair of input/output commutators synchronised with those input/output commutators in the interleaver. However, the shift registers are stacked in the reverse order of those in the interleaver to perform the inverse operation of that in the interleaver.

In practice, RAM is used instead of shift registers in the design and implementation of a convolutional interleaver and deinterleaver. Total end-to-end delay and the memory requirement in case of both convolutional interleaver and deinterleaver is $L(N-1)$ symbols and $L(N-1)/2$ only which is one-half that of a block interleaver and deinterleaver.

7.11 SPEECH CODING

The efficient utilisation of the allocated spectrum is the prime objective in the design of digital wireless communication systems. The systems rely on the use of speech coding to remove almost all the natural redundancy inherent in an analog speech signal, while ensuring a high-quality reproduction of the original speech signal at the receiver. The speech coding, also called *source coding*, makes the information signal compatible with digital processing. Encoding analog voice into digital format enhances the quality of voice, improves the overall performance of the system in terms of spectral efficiency, and increases the system capacity. Digital-modulation techniques combined with robust speech-coding schemes, which can inherently withstand higher bit-error rates needing less channel coding, will be the ultimate choice for a spectrum-efficient mobile communication system.

The carrier modulation and speech-coding techniques are logically independent processes but are strongly interrelated. The improvement in either of them is towards achievement of a common goal of higher spectral efficiency. Higher level modulation offers higher spectral efficiency but higher signal-to-noise (S/N) ratio is required to achieve a given BER that is difficult to achieve in a mobile environment. The required S/N can be reduced by using low bit-rate speech coding and efficient error-correction techniques prior to modulation. The important parameters of a speech coder are the transmitted bit rate, the speech quality, the robustness in the presence of fading and interference, and the complexity of implementation. Various speech-coding techniques are mainly based on linear predictive coding (LPC) strategy.

7.11.1 Characteristics of Speech Signals

The speech signals can be bandlimited to 10 kHz without affecting their perception. The standard telephone speech band is 300 Hz – 3400 Hz, with a sampling frequency of 8000 samples per second. The wideband speech band is specified as 50 Hz – 7000 Hz, with 16000 samples per second having AM broadcast quality. The standard FM broadcast uses an audio range of 50 Hz – 15000 Hz at 32000 samples per second.

The CD audio frequency range is 20 Hz – 20000 Hz at 44100 samples per second. The excitation modes in human speech can be categorised as follows:

Voiced Air forced through the glottis, vibrating the vocal cords to produce quasi-periodic pulses which excite the vocal tract (vowels, diphthongs, semivowels, ...).

Fricatives Forming a constriction and forcing air at high enough velocity to produce turbulence (broad-spectrum noise source of excitation). For example, *Voiced*: v, the, z, zh; *Unvoiced*: f, th, s, sh.

Plosives Making a complete closure, building up pressure behind closure, and abruptly releasing it. For example, *Voiced*: b, d, g, dzh; *Unvoiced*: p, t, k, tsh.

The speech signal has characteristics such as generally non-stationary; can be assumed to be stationary over short segments of 5–30 ms; has strong correlation among slowly varying spectrum, energy, pitch, and voicing; has short-term and long-term correlations; and few resonances (formants) and harmonic structure. Voiced speech has mostly a frequency range as 60–200 Hz (male), 150–450 Hz (female), and 200–600 Hz (child). It has a large dynamic range and most of the voice energy is contained in the low frequency range which forms 65% of speech. Unvoiced speech has no pitch periodicity, smaller dynamic range, and most of the energy is contained in the high-frequency range.

7.11.2 Quantisation of Speech Signal

Quantisation is the process of converting an infinite number of possible values or levels to a finite number of conditions. Speech signals contain an infinite number of amplitude values in time domain. Analog-to-digital conversion and data compression are essential requirements of speech coding. The input information signal is converted to binary digits; the bits are then grouped to form message symbols. The digital messages in the logical format are then transformed into baseband waveforms, the process called pulse code modulation which includes sampling, quantisation and A/D conversion. The term baseband refers to a signal whose spectrum extends from dc up to some finite value, usually less than a few MHz. Thus, converting an analog speech signal to a pulse-code modulation code with a limited number of combinations require quantisation. Scalar quantisation is a process that maps all inputs within a specified range to a common value. It maps inputs in a different range of values to a different common value. It digitises an analog signal.

In essence, quantisation is the process of rounding off the amplitude levels of flat-top samples of the analog speech signal to a manageable number of levels. A quantisation partition defines several contiguous, non-overlapping ranges of values within the set of real numbers. Quantisation distorts the original analog signal. The quantisation error is inversely proportional to the number of levels for an amplitude range. Two parameters determine a quantisation: a partition and a code book. Distortion can be reduced by choosing appropriate partition and code book parameters. However, testing and selecting parameters for large signal sets with a fine quantisation scheme can be tedious. One way to produce partition and code book parameters easily is to optimise them according to a set of so-called training data. The training data should be typical of the kinds of signals that will be actually quantised. The performance of a quantiser is measured as the output signal-to-quantisation noise ratio.

Thus, the process of transforming sampled amplitude values of an analog speech signal into a discrete amplitude value is referred to as quantisation. The peak-to-peak range of the input sample values is subdivided into a finite set of decision levels or decision thresholds, and the output is assigned a discrete value selected from a finite set of representation levels.

Facts to Know!

The amplitude difference between the sampled value and the quantised level is called the quantisation error. This error is directly proportional to the difference between consecutive quantisation levels. With a higher number of quantisation levels, a lower quantization error is obtained.

A quantiser is memoryless in the sense that the quantiser output is determined only by the value of a corresponding input sample, independently of earlier analog samples applied to the input.

The quantisation process introduces an error defined as the difference between the input signal and the output signal. This error is called the *quantisation noise*. It is produced due to rounding off sample values of an analog baseband signal to the nearest permissible representation levels of the quantiser. As such, quantisation noise differs from channel noise in that it is signal dependent.

7.11.3 Uniform and Nonuniform Quantisation

Depending upon the process of quantisation to distribute quantisation levels to be uniformly spaced or not, the quantisation can be categorised as uniform quantisation or nonuniform quantisation. In ***i q ti ti***, the quantisation levels are uniformly spaced, and the distortion introduced is quite often modeled as additive quantisation noise. A PCM coding technique, using 8 bits per sample at a sample frequency of 8 KHz adopted for commercial telephone application at 64 kbps data rate, is an example of uniform quantisation.

> **Facts to Know!**
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> Experimentally it has been found that 256 quantisation levels are necessary to achieve an acceptable signal-to-noise ratio of a telephone toll-grade quality voice. This requires 8-bits of information per quantised sample.

A quantiser whose SNR remains essentially constant for a wide range of input power levels is said to be robust. The provision for such robust performance necessitates the use of a non-uniform quantiser. In ***i q ti ti***, the spacing between the quantisation levels is unequal and mostly the relation is logarithmic. In a non-uniform quantiser, the step size varies. For smaller amplitude ranges the step size is small and for larger amplitude ranges, the step size is large. The nonuniform quantisation technique employs an additional logarithmic amplifier before processing the sampled speech signals by a uniform quantiser. The operation of a nonuniform quantiser is equivalent to passing the baseband signal through a compressor and then applying the compressed signal to a uniform quantiser. The resultant signal is then transmitted. At the receiver, a device with a characteristic complementary to the compressor, called an *expander* is used to restore the signal samples to their correct relative level. The compressor and expander taken together constitute a compander. In this technique, strong speech signals are compressed and weak speech signals are amplified in accordance with US μ-law companding or European A-law companding rules.

Nonuniform quantisation has higher average signal-to-quantisation noise power ratio value than that of in the uniform quantiser. RMS value of the quantiser noise power of a nonuniform quantiser is substantially proportional to the sampled value and hence the effect of the quantiser noise is also reduced. Hence, the advantages of nonuniform quantisation are reduced quantisation noise and high average SNR value as compared to uniform quantisation.

7.11.4 Adaptive Quantisation

The term 'adaptive' means being responsive to changing level and spectrum of the input speech signal. The variation of performance with speakers and speech material together with variations in signal level inherent in the speech communication process, make the combined use of adaptive quantisation and adaptive prediction necessary to achieve the best performance.

An adaptive quantiser varies its step size in accordance to the input speech signal power. The quantised version of the original input is reconstructed from the decoder output using the same predictor as used in the transmitter. The prediction gain is maximised by minimising the variance of the prediction error. In the absence of noise, the encoded signal at the receiver input is identical to the encoded signal at the transmitter output.

For the speech signals which do not change rapidly from one sample to the next sample, adaptive quantisation is preferred. With traditional quantisation, when such highly correlated samples are encoded, the resulting encoded signal contains redundant information. By removing this redundancy before encoding, an efficient coded signal can be obtained. By knowing the past behaviour of a signal up to a certain point in time, it is possible to make some inference about the future values. A digital coding scheme that uses both adaptive quantisation and adaptive prediction is called *adaptive differential pulse code modulation* (ADPCM). The use of adaptive prediction in ADPCM is required because speech signals are inherently non-stationary, a phenomenon that manifests itself in the fact that autocorrection function and power spectral density of speech signals are time-varying functions of their respective variables. This implies that the design of predictors for such inputs should likewise be time-varying, that is, adaptive. As with adaptive quantisation, there are two schemes for performing adaptive prediction — adaptive prediction with forward estimation (APF), in which unquantised samples of the input signal are used to derive estimates of the predictor coefficients; and adaptive prediction with backward estimation (APB), in which samples of the quantiser output and the prediction error are used to derive estimates of the prediction error and predictor coefficients. ADPCM speech coding technique is used in the cordless telephone standards such as CT2, DECT, and PHS at 32 kbps data rate.

Facts to Know!

The term 'adaptive quantisation' refers to a quantiser that operates with a time-varying step size. It is an efficient quantisation technique because usually the speech signals have a large dynamic range, as large as 40 dB or more.

PCM and ADPCM are both time-domain coders in that the speech signal is processed in the time-domain as a single full-band signal. *pti e* *i* (ASBC) is a frequency domain coder, in which the speech signal is divided into a number of sub-bands and each one is encoded separately. The noise shaping is accomplished by adaptive bit assignment. In particular, the number of bits used to encode each sub-band is varied dynamically and shared with other sub-bands, such that the encoding accuracy is always placed where it is needed in the frequency domain characterisation of the signal. Indeed, sub-bands with little or no energy may not be encoded at all.

The adaptive sub-band coder is capable of digitising speech at a data rate of 16 kbps with a speech quality comparable to that of 64 kbps PCM. To accomplish this performance, it exploits the quasi-periodic nature of voiced speech and a characteristic of the hearing mechanism known as *noise masking*. Periodicity of voiced speech manifests itself in the fact that people speak with a characteristic pitch frequency. This periodicity permits pitch prediction, and therefore a further reduction in the level of the prediction error that requires quantisation, compared to differential pulse-code modulation without pitch prediction. The number of bits per sample that needs to be transmitted is thereby greatly reduced, without a serious degradation in speech quality.

Facts to Know!

The CD-900 cellular system uses adaptive sub-band coding for speech coding.

7.11.5 Vector Quantisation

The difference between vector quantisation and scalar quantisation is that vector quantisation uses the data block as a vector and each of the vectors is compared with a pre-designed code book and chooses a vector with the best match. With scalar quantisation, there is no blocking and the quantisation is applied to each coefficient. Scalar quantisation or non-uniform quantisation is used more because it is less complex.

The method of vector quantisation is used as a coding mechanism in many application areas such as image coding, speech coding, and speech recognition. The vector quantisation coding technique estimates a set of vectors represented by groups of parameters that portray the block of data under consideration. These vectors are indexed as reference vectors and termed as a *code book*. The code book symbolises reference features with these being quantised and having minimum average distortion.

The majority of coding systems are refined using small records of data with uncorrelated features, and given suitable distance, or a distortion measure. One can therefore accomplish good results in many applications terms such as speaker identification and recognition. Generally, the distortion measure that is used for the vector quantisation coding process is the *Euclidean distance measure*. The vector quantization coding technique is implemented using field programmable gate arrays (FPGAs) devices. FPGA based vector quantisation coding systems can eliminate the need for simulation in an onboard level design, requiring only simulation to be done on chip level designs.

7.11.6 Types of Vocoder

A voice coder (vocoder) analyses the voice signal and tries to reduce the amount of data that needs to be transmitted by constructing a model for the human vocal system. A vocoder can imitate human voice with an electronic system, comprising of basically excitation and filter functions.

Vocoders can be classified as channel vocoders, formant vocoders, cepstrum vocoders, voice-excited vocoders, and linear predictive vocoders. *Channel vocoders* are primarily frequency domain vocoders that determine the envelope of the speech signal for a number of frequency sub-bands in the audio frequency range. Various parameters of the synthesised speech such as pitch frequency, the voice/unvoiced decision, and sub-band energy levels are transmitted. The decoder operation is just the inverse of vocoder operations. The *formant vocoder* transmits the positions of the formants (peak) of the spectral envelope of the voice signal instead of sending all samples of the power spectrum envelope. Ideally, a formant vocoder can operate at 1200 bps but they are difficult to implement practically.

> **Facts to Know!**
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> Vocoders model the speech-generation process. The basic model consists of an excitation signal typical of the air pressure modulated by the vocal cords and a filter characterising the vocal track (mouth and nose) at a competitively low rate (typically 50 times a second) in order to simulate the speed of movement of the mouth and tongue.

The *cepstrum vocoders* distinguish between the low frequency coefficients in the vocal tract spectral envelope and high-frequency coefficient of vocal tract excitation. *Voice-excited vocoders* generate a spectrally flat signal with energy at pitch harmonics. They operate at 7.2 kbps – 9.6 kbps with a reasonable good voice quality. *Linear predictive vocoders* are time-domain vocoders which extract the prominent features of speech from the time waveform of the speech signal. LPCs are widely used in wireless communication systems operating at low bit rate with acceptable voice quality.

7.11.7 Linear Predictive Vocoders

Prediction simply means predicting or estimating the present or future value of a discrete-time signal, given a set of past samples of the signal. The prediction error is defined as the difference between the actual future value of the signal and the predicted value produced by the predictive vocoders. The smaller the prediction error, the more reliable the vocoder will be. Linear predictive vocoders are based on the principle of analysis by synthesis, which implies that the encoder includes a replica of the decoder in its design. Figure 7.25 shows the functional block diagram of a typical linear predictive vocoder and decoder.

A linear predictive vocoder consists of three main parts: a synthesis filter, an excitation generator, and error minimisation which form a closed-loop optimisation process to operate at a bit rate below 16 kbps. The function of the synthesis filter is to produce a synthetic version of the original speech that is configured to be of high quality. It may consist of a linear filter connected in cascade with a long-term predictor. The linear filter is designed to model the short-term spectral envelope of speech, whose parameters are computed on the basis of predicting the present samples of the speech signal from 8–16 previous samples. The speech samples are divided into 10–30 ms long speech frames for computing the filter parameters, and each frame is subdivided further into 5–15 ms long sub-frames to optimise the excitation. The long-term predictor is used for modeling the fine structure of the speech spectrum. The function of an excitation generator is to produce

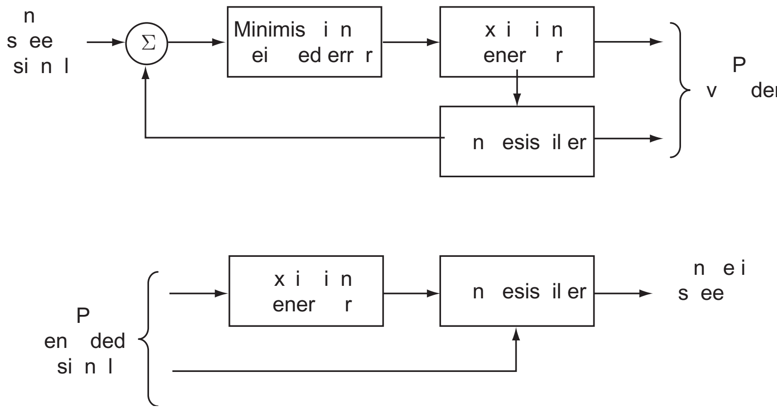

Fig. 7.25 *Block diagram of linear predictive vocoder and decoder*

the excitation applied to the synthesis filter. The excitation means adjustable amplitudes and positions of the individual pulses, consisting of a definite number of pulses every 5–15 ms. For optimising the weighted error between the original speech and the synthesised speech, the mean-square error minimisation technique is employed. The quantised filter parameters and quantised excitation constitute the transmitted signal. The LPC transmit the selected characteristics of the error signal such as the gain factor, the information about the pitch, and voiced/unvoiced decision.

At the receiver, the speech decoder consists of the excitation generator and the synthesis filter, identical to the ones used in the vocoder. The decoder uses the received signal to produce a synthetic version of the original speech signal. The decoded excitation is passed through the synthesis filter having identical parameters as those used in the vocoder. To compress a speech signal using linear prediction, the excitation signal and prediction coefficients must be quantised using as few bits as possible without sacrificing quality. However, there is always a trade off, as lower bit rates are achieved, the quality of the reproduced speech is reduced. When the LPC uses more than one pulse, it is called multipulse excited LPC (MPE-LPC). The MPE-LPC technique offers excellent speech quality at high complexity. Moreover, it is not much affected by channel errors.

> **Facts to Know!**
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> LPC vocoders model the vocal-tract filter by means of a single, linear all-pole (between 6 and 12) filter.

Residual Excited LPC (RELP) In residual excited LPC, the speech signal is synthesised and subtracted from the original speech signal to form a residual signal. The filter coefficients and excitation parameters from a speech frame are transmitted along with the residual signal after quantisation and encoding. At the receiver, firstly the signal is generated using the LPC model parameters. Then the residual signal is added to synthesise an approximation of the original speech signal, thereby improving the speech signal quality at low complexity.

Regular Pulse Excited LPC (RPE-LPC) The intervals between the individual pulses in the excitation are constrained to assume a common value. The resultant analysis-by-synthesis vocoder is said to have regular-pulse excitation. It reduces the computational complexity of the vocoder and decoder design. RPE-LPC is also called regular pulse excited long-term predictor (RPE-LTP) because a long-term prediction loop is added to the RELP vocoder. This configuration has resulted in reduction of the net bit rate from 14.77 kbps to 13 kbps without any degradation in speech quality. This type of vocoder is used in GSM and DCS-1800 cellular systems standards.

Code Excited LPC (CELP) It uses a predetermined code book of zero-mean white Gaussian vectors as the source of excitation for the synthesis filter which performs short-term as well as long-term predictions. Figure 7.26 shows the functional block diagram of a CELP vocoder.

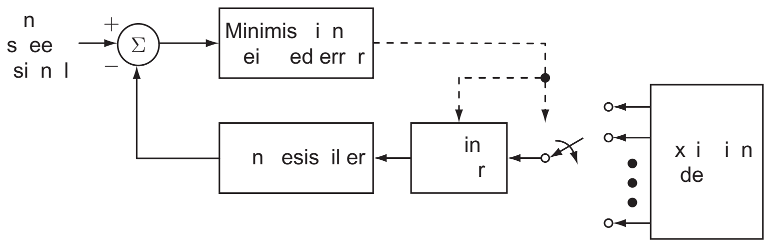

Fig. 7.26 *Block diagram of CELP vocoder*

Firstly, the synthesis filter parameters are computed from the original speech signal samples. The average power of the weighted error between the original speech signal and the synthesised speech signal at the output of the synthesis filter is minimised. Then the gain factor and the selection of a particular code stored in the excitation code book is optimised. At the receiver, the parameters of the synthesis filter and the code book are known. Using the received signal, a CELP decoder computes its own synthesis filter and the appropriate excitation for the filter. The output is a synthetic version of the original speech signal with a good quality speech at bit rates below 8 kbps. CELP is implemented using digital signal processing and VLSI technology because of intensive computational complexity. The IS-95 CDMA cellular system used CELP coders at 1.2 kbps to 14.4 kbps.

Facts to Know!

The implementation of the VSELP speech codec algorithm utilises less than 60 mA current from a 5-V battery when operated with a 2-MHz clock frequency.

Vector-Sum Excited LPC (VSELP) It is a modification of the CELP vocoders in the sense that the code books are organised with a pre-defined structure with an objective of reduction in the time required for finding out the optimum code word. These code books impart high speech quality, modest computational complexity, and increased robustness to channel errors. It is used in IS-136 standard for a US digital cellular system at a raw data rate of 8 kbps and a channel data rate of 13 kbps.

7.11.8 Comparison of Speech Coders

The selection of a coding scheme is mainly driven by achieving the desired voice quality by adding minimum overhead while protecting the information bits to counter the impact of the channel noise. Treating those bits within a message that decides the speech quality separately from those that are less critical, assists this strategy. There are two types of bit rates associated with a speech coder, that is, the raw or uncoded bit rate and the encoded bit rate to account for error correction. A low-rate speech coder will essentially require less bandwidth for transmission which is useful in wireless environments. But usually, this is achieved at the expense of the quality of speech. For example, traditional linear predictive coders (LPC) are capable of providing good voice quality at very low bit rates (1.2 kbps – 4.8 kbps) as compared with the 64 kbps rate of PCM, but tend to be extremely vulnerable to errors. RELP coders operating at 8 kbps – 16 kbps and sub-band coders operating at 16 kbps have also proved to be fairly robust.

The speech-coding scheme adopted for USDC system is Vector-Sum Excitation Linear Prediction (VSELP) operating at around 8 kbps. The channel coding used is a convolutional code with a constraint length of 6.

The overall coded bit rate is 13 kbps only. With overhead bits and slow-associated control channel, the overall bit rate is 16.2 kbps per channel and 48.6 kbps for 3 channels. The spectral efficiency (1.6 bps/Hz) is better than that of a GSM system. The USDC system uses $\pi/4$-QPSK modulation with 30 kHz channel spacing.

Regular-pulse excited linear predictive coder with (RPE-LPC) speech coder used in GSM cellular system operates at 13 kbps data rate and utilises a speech frame of 20 ms duration. The most sensitive bits are protected by a convolutional code with 1/2 rate and a constraint length of 5. The overall coded bit rate per speech signal is 22.8 kbps. To combat the channel noise, 30% overhead (10.1 kbps) as guard time and synchronisation, and 0.95 kbps as slow-associated control channel is added before transmission. Thus, the gross bit rate is 33.85 kbps per channel and 270.8 kbps for 8 channels. The spectral efficiency of GSM system employing GMSK modulation ($B \times T_b = 0.3$) technique operating with 200 kHz channel spacing is 1.35 bps/Hz.

Table 7.6 summarises the bit rate comparison of various types of coders used in different cellular standards across the world.

Table 7.6 *Bit-rate comparison of various coders*

S. No.	*Voice coder*	*Cellular standard*	*Uncoded bit rate*	*Encoded bit rate*
1.	VSELP	IS-136 (2G US digital cellular); JDC (2G Japanese digital cellular)	8 kbps	13 kbps
2.	RPE-LPC	GSM (2G European digital cellular); DCS-1800 PCS in US	13 kbps	22.8 kbps
3.	QCELP	IS-95 (2G CDMA digital cellular)	9.6, 4.8, 2.4, and 1.2 kbps	22.8 kbps or 19.2 kbps

A higher rate encoder is employed for speech coding in those system applications where the voice quality is the most important criterion rather than the spectral efficiency. An adaptive differential pulse code modulation (ADPCM) scheme is used in cordless systems such as CT-2 and DECT which operate at uncoded as well as overall bit rate of 32 kbps. Speech coders may give a poor performance when the error rates are as high as 1 in 100. In such cases, error control coding along with voice-compression schemes and block interleaving techniques are usually employed with low-rate speech coders. The voice-compression scheme removes redundancy in a digitised voice; error-control coding and interleaving techniques introduce controlled redundancy to provide better performance in a mobile wireless environment.

Facts to Know!

Voice quality is subjective. Statistical opinion scores of several hundred listeners are frequently required. Therefore, rigorous and fair performance comparisons of various codec methods are highly complex and elaborative tasks.

7.12 HAND-OFF MECHANISMS

In cellular mobile communication system, continuation of an active call is one of the most important performance determining key parameter. The hand-off process enables a cellular system to provide such a facility by transferring an active call from one cell to another cell automatically as and when the need arises. The transfer of current communication channel could be in terms of frequency band, time slot, or code word to a new cell-site. Hand-off occurs when a received signal from its serving cell becomes weak and another cell-site can provide a stronger signal to the mobile subscriber. If the new cell-site has some free voice channels then it assigns one of them to the handed-off call. If all of the voice channels are busy at the hand-off time there are two possibilities: to drop the call or to delay it for a while till the voice channel becomes available.

Thus, hand-off refers to the process of transferring an ongoing call from one channel connected to one cell to another channel connected to an adjacent cell without interruption when the mobile subscriber is moving from one cell to another cell. In a cellular system, all mobile calls may not be completed within the boundary of a relatively small cell.

To provide continuation of a call between cells, the cellular system is equipped with its own system-level switching and control capability. Through continuous monitoring of received signal strength or other digital parameters received from individual cell-sites, the cellular system can sense when an active mobile subscriber with a call in progress crosses the radio coverage boundary of its serving cell to another cell and can switch the call to a new cell without interruption. This is termed hand-off.

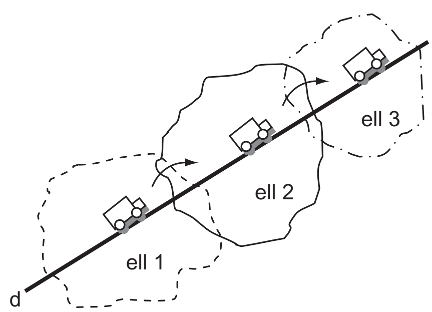

Fig. 7.27 *The occurrence of hand-off*

Figure 7.27 depicts the occurrence of hand-off at the joint boundaries of two adjacent cells. Here, a mobile subscriber is moving away from the area covered by one cell and entering the area covered by another cell, the call is transferred to the second cell in order to avoid undesired call termination when the mobile subscriber goes outside the range of the first cell.

There may be different reasons why a hand-off might be needed. Hand-off may be needed in the following situations where the cell-site receives weak signals from the mobile unit.

Scenario 1 Hand-off may be needed at the cell boundary, based on the threshold level of received signal strength, typically –100 dBm in a noise-limited environment and –95 dBm in interference-limited systems.

Scenario 2 Hand-off may be needed at the cell boundary, based on the carrier-to-interference ratio (C/I), typically 18 dB or so, in order to have telephone voice quality signal.

Scenario 3 Hand-off may be needed when the mobile unit is reaching the signal-strength holes (also called weak spots or signal gaps or coverage holes) within the cell coverage area. In this particular case, the hand-off is necessary in order to maintain the continuity of the on-going call but cannot be made because of non-availability of received signal from any other cell since this small area is not covered by any other adjacent cell. This situation is depicted in Fig. 7.28.

There are a number of other reasons for hand-off to occur such as

– when the capacity for connecting new calls of a given cell is exhausted, an existing or new call from a mobile subscriber, which is located in an area overlapped by another cell, is transferred to that cell in order to free-up some capacity in the first cell for other mobile subscribers, who can only be connected to that cell
– when the channel used by the mobile subscriber is interfered by another mobile subscriber using the same channel in a cochannel cell, the call is transferred to a different channel in the same cell or to a different channel in another cell in order to avoid the cochannel interference

Similarly, in CDMA networks a soft hand-off may be induced in order to reduce the interference to a smaller neighbouring cell due to the near–far effect even when the mobile subscriber still has an excellent communication link to its current cell.

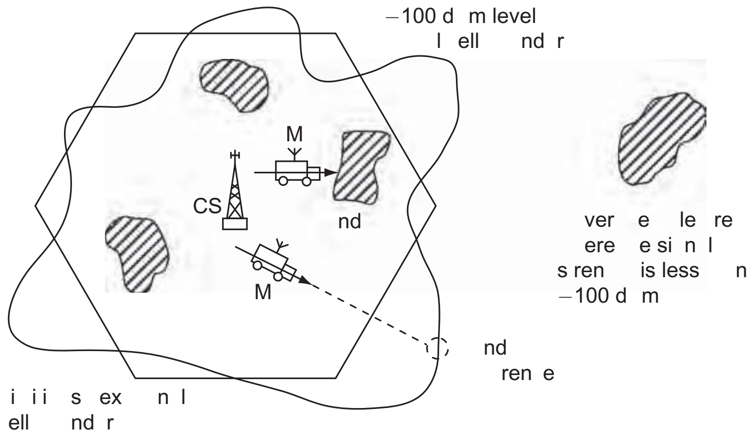

Fig. 7.28 *The need of hand-off*

7.12.1 Types of Hand-off

Two of the most important metrics for evaluating the performance of a hand-off procedure are call-blocking probability and forced-call termination probability. The *call-blocking probability* is the probability of blocking a new call request. The *forced-call termination probability* is the probability of dropping an active call due to hand-off failure. The primary objective of a hand-off procedure is to decrease forced termination probability while not increasing the call-blocking probability significantly.

Depending upon the nature of occurrence of hand-off, hand-off procedures can be classified in the following ways :

- Inter-cell hand-off
- Intra-cell hand-off
- Hard hand-off
- Soft and softer hand-off

The most basic form of hand-off is when a mobile call in progress is redirected from its present serving cell to a new target cell using a different channel. In cellular networks, the present serving and the new target cells may be served from two different cell-sites or from one and the same cell-site using two sectors. Such a hand-off, in which the present serving and the new target cells are different cells (even if they are on the same cell-site) is called *i te e* .

The purpose of inter-cell hand-off is to maintain the call as the mobile subscriber is moving out of the area covered by the present serving cell and entering the area of the new target cell. Sometimes a mobile call may be initiated in one cellular system controlled by one Mobile Telephone Switching Centre or Mobile Switching Centre and the mobile enters another cellular system controlled by different MTSO or MSC before termination of the call. In such situations, intersystem hand-off take place so that the call is continued, as shown in Fig. 7.29.

Hand-off request by the first MTSO (MTSO A) is sent to the second MTSO (MTSO B) through a dedicated link between them and the second MTSO makes a complete hand-off during the call communication. A special case is possible, in which the present serving cell and the new target cell are one and the same cell and only the used channel is changed during the hand-off. Such a hand-off, in which the cell is not changed, is called *i t e* . The purpose of intra-cell hand-off is to change one channel, which may be interfered or affected by fading, with a new clearer or least fading channel.

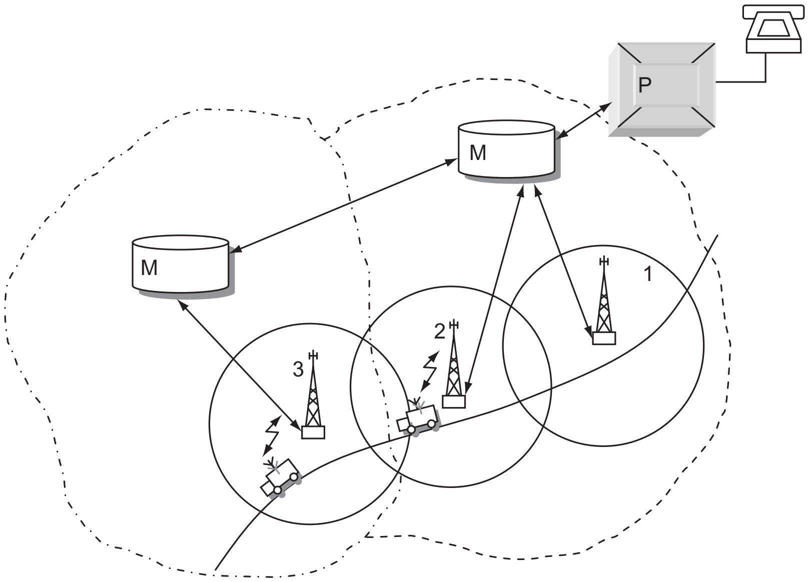

Fig. 7.29 *Intersystem hand-off*

A is one in which the channel in the present serving cell is released and only then the channel in the new target cell is engaged. Thus, the connection to the present serving cell is broken before the connection to the new target cell is made. That is the reason such hand-offs are also known as *break-before-make hand-off*. Hard hand-offs are intended to be instantaneous in order to minimise the interruption to the call. An advantage of the hard hand-off is that at any moment in time, one call uses only one channel. The hard hand-off event is indeed very short and usually is not perceptible by the subscriber. In the analog cellular systems, it could be heard as a click or a very short beep. In digital cellular systems it is unnoticeable. Another advantage of the hard hand-off is that the mobile equipment need not be capable of receiving two or more channels in parallel, which makes its design simpler and cost-effective. A disadvantage of hard hand-off is that if a hand-off fails the call may be temporarily interrupted or even terminated abnormally.

A ***t*** procedure can establish multiple connections with neighbouring cells. A soft hand-off is the one in which the channel in the present serving cell is retained and used for a while in parallel with the channel in the new adjacent target cell. In this case, the connection to the new adjacent target cell is established before the connection to the present serving cell is broken, hence this hand-off is also called *make-before-break hand-off*. The interval, during which the two connections are used in parallel, may be very small or substantial. A soft hand-off may involve using connections to more than two cells by one mobile subscriber at the same time. When a call is in a state of soft hand-off, the signal out of the best of all used channels can be utilised for the call at a given moment. Alternatively, all the signals can be combined to produce a clearer copy of the signal. This type of hand-off is then termed as *softer hand-off*. Softer hand-offs are possible when cells involved in the hand-off have a single cell-site but different sectors.

One advantage of the soft hand-offs is that the connection to the present serving cell is broken only when a reliable connection to the new target cell has been established. Therefore, the chances that the call will be terminated abnormally due to a failed hand-off are extremely lower. Moreover, simultaneously channels in multiple cells are maintained and the call could only fail if all of the channels are interfered or faded equally at the same time. Interference and fading in different channels are not related and therefore the probability of them taking place at the same moment in all channels is very low. Thus, the reliability of the connection becomes higher even when the hand-offs occur in locations of poor radio coverage. The disadvantage of soft hand-off is more complex and expensive mobile equipment, which must be capable of processing several channels in parallel. And there is need of several channels in the network to support just a single call. This reduces the number of remaining free channels and thus reduces the capacity of the network.

Soft hand-off is used by the CDMA cellular systems where the cells use same frequency band in all the channels using different code words. Each mobile subscriber maintains an active set where cell-sites are added when the RSSI exceeds a given threshold level and removed when RSSI drops below another threshold value for a given amount of time specified by a timer. When a presence or absence of a cell-site to the active set is encountered, soft hand-off occurs.

7.12.2 Initiation of Hand-off

In cellular systems, once a call is established, the set-up or control channel is not used during the call duration. Therefore, hand-off is always implemented on the voice channel. The value of implementing hand-offs as well as the number of hand-offs per call depends upon the size of the cell. For example, the smaller the cell size, the greater the number of occurrence of hand-offs and the value of implementing hand-offs.

For the practical realisation of hand-offs in a cellular network, each cell is assigned a list of potential target cells, which can be used for handing-off calls from this serving cell to them. These potential target cells are called *neighbours* and the list of such potential target cells is called *neighbour list*. Creating a reliable neighbour list for a given cell is possible with the use of customised special computer tools. These software tools implement different hand-off algorithms and may use input data from field measurements or computer predictions of radio wave propagation in the areas covered by the cells.

During a call, one or more parameters of the signal in the voice channel in the present serving cell are monitored and assessed in order to decide when a hand-off may be necessary. The system parameters may include the received signal power, the received signal-to-noise ratio, bit error rate, frame error rate, received speech quality level, distance between the mobile subscriber and the cell-site (estimated from the radio signal propagation delay) and many other parameters.

The system parameters on the downlink (forward link) as well as on the uplink (reverse link) may be monitored. The hand-off may be requested by the mobile subscriber unit or by the cell-site of its present serving cell and, in some systems, by a cell-site of a neighbouring cell. The mobile subscriber and the cell-sites of the neighbouring cells monitor each others' received signal levels and the best target cell candidates are selected among the neighbouring cells.

In cellular wireless networks, it is very important to deal with mobile hand-off between cells so that they can maintain a continuous and QoS (Quality of service)-guaranteed service. Hand-off initiation is the process of deciding when to request a hand-off. Usually, hand-off decision is based on received signal strength indicator (RSSI) from present cell-site and neighboring cell sites. Figure 7.30 shows RSSI level of current sell-site (CS1) and one neighboring cell-site (CS2).

The RSSI gets weaker as mobile goes away from CS1 and gets stronger as it gets closer to the CS2 as a result of signal propagation. The received signal is averaged over time using an averaging window to remove momentary fadings due to geographical and environmental factors.

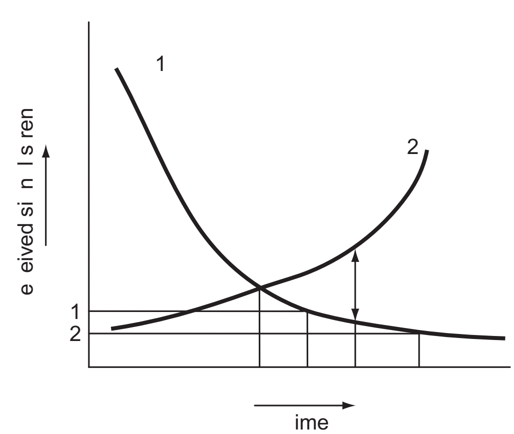

Fig. 7.30 *Movement of a mobile in the hand-off zone*

Major hand-off initiation techniques include relative RSSI, relative RSSI with threshold value, relative RSSI with hysteresis, and relative RSSI with hysteresis and threshold.

(a) In the *relative RSSI technique*, the RSSI levels are measured over time and the cell-site with strongest signal is chosen to hand-off. For example, as seen in Fig 7.30, RSSI of CS2 exceeds RSSI of CS1 at the point *A* and hand-off is requested. However, due to signal fluctuations, several hand-offs can be requested while CS1's RSSI is still enough to serve mobile. These unnecessary hand-offs are known as *ping-pong effect.* As the number of hand-offs increase, the forced termination probability also increases. So, hand-off techniques should avoid unnecessary hand-offs.

(b) In *relative RSSI* with threshold value technique of hand-off initiation, a threshold value (say T1 in Fig 7.30) is introduced to overcome the ping-pong effect. The hand-off is initiated if CS1's RSSI is lower than the threshold value and CS2's RSSI is stronger than that of CS1. The hand-off request is issued at the point *B* as shown in Fig. 7.30. The relative RSSI with hysteresis technique uses a hysteresis value ('*h*' in Fig 7.30) to initiate hand-off. Hand-off is requested when the CS2's RSSI exceeds the CS1's RSSI by the hysteresis value '*h*' (the point *C* in Fig. 7.30).

(c) The *relative RSSI with Hysteresis and Threshold value technique* combines both the threshold and hysteresis value concepts to come with a technique with minimum number of hand-offs. The hand-off is requested when the CS1's RSSI is below the threshold (T1) and CS2's RSSI is stronger than CS1's by the hysteresis value *h* (the point *C*). If a lower threshold value is chosen than T1 (but higher than T2) then the hand-off initiation would be somewhere at the right of the point *C*.

In all these techniques, hand-off initiation takes place before the receiver threshold point *D*. Receiver threshold is the minimum acceptable RSSI for call continuation (T2). If RSSI drops below the receiver threshold, the ongoing call is then dropped. The time interval between hand-off request and receiver threshold enable cellular systems to delay the hand-off request until the receiver threshold time is reached when the neighboring cell does not have any free channels.

7.12.3 Delaying of Hand-off

A hand-off could be delayed if no available cell could take the call. If no neighboring cells are available even after the second hand-off level is reached, the call is dropped below the threshold level. The MTSO always handles the hand-off call on priority, followed by the new originating calls. One of the advantages of delayed hand-offs is to make the hand-off occur at the proper location and eliminate possible interference in the system. If the neighboring cells are busy, delayed hand-off may help the switching processor to handle call processing adequately. Delayed hand-off can even circumvent the need for a hand-off if the mobile unit happens to be in a coverage hole for a short duration.

Queuing of hand-offs is more effective than two-threshold-level delayed hand-offs. If the neighboring cells are busy, MTSO will queue the requests of hand-off calls. Queuing of hand-offs is possible because there is finite time interval between the time the received signal level drops below the hand-off threshold level and the time the call is dropped due to inadequate signal level. Queuing for the hand-off is more important because call drops upset mobile subscribers more than call blockings of initiating calls. The queuing hand-off calls-prioritisation scheme queues the hand-off calls. When all the available channels are occupied in a cell-site. When a channel is released, it is assigned to one of the hand-off calls in the queue. A new call request is assigned a channel if the queue is empty and if there is at least one free channel in the cell-site. Also, some systems queue new calls to decrease call-blocking probability. Normally, adding queues in hand-off calls does not affect the blocking probability of originating calls. There is a trade off between the total carried traffic and the reduction in probability of forced termination of calls.

The time interval between hand-off initiation and receiver threshold makes it possible to use queuing hand-off calls. When a channel is released at cell-site, a timer is started. If a hand-off request is done in that time interval, it is assigned to it. Otherwise, when the timer expires, the channel can be assigned to new or hand-off calls depending on the arrival order. In a prioritisation scheme called *Measurement Based Prioritisation Scheme* (MBSP), the hand-off calls are added to the queue and priorities of the calls changes dynamically based on the power level they have. The calls with power level close to the receiver threshold have the highest priorities. This scheme provides better results from the first-in first-out (FIFO) queuing scheme where the hand-off calls are served.

7.12.4 Hand-off Control Algorithms

At the cell-site, the received signal strength is always monitored on a reverse voice channel. When the received signal strength reaches the defined hand-off level (usually higher than the threshold level for the minimum required received signal level for desired voice quality) for a predefined period then the cell-site sends a request to MTSO for a hand-off on the call. Unnecessary hand-offs are avoided as a result of the direction and the speed of the vehicle.

There are two situations where hand-offs are necessary but cannot be made! Firstly, when the mobile subscriber is located at a signal-strength hole within a cell but not at the boundary—the call must be kept in the old frequency channel until it is dropped as the result of an unacceptable received signal level. Secondly, when the mobile subscriber approaches a cell boundary but no channels in the new cell are available, the new cell must reassign one of its frequency channels within a reasonably short period. The hand-off control algorithms tend to decentralise the decision making process, which help to reduce the hand-off delays. There are four basic types of hand-off algorithms:

- Network-controlled hand-off (NCHO)
- Mobile-assisted hand-off (MAHO)
- Soft hand-off (SHO)
- Mobile-controlled hand-off (MCHO).

et t e is a centralised hand-off protocol, in which it is the cellular network that makes hand-off decision based on measurements of the signal quality of a mobile subscriber unit at a number of adjacent cell-sites. Specifically, if the received signal level from a serving mobile subscriber is measured to have a weaker signal in its present serving cell, while a stronger signal in a neighbouring cell then a hand-off decision could be made by the cellular network to switch a cell-site from the present serving cell to the new target cell. Such a type of hand-off usually takes 100–200 milliseconds and produces a noticeable interruption in the conversation. The overall delay of this type of hand-off is generally in the range of 5–10 seconds. Thus, this type of hand-off is not suitable in a rapidly changing environment and a high density of subscribers due to the associated delay. In NCHO, the load of the network is high since the network handles the complete hand-off process itself. NCHO is used in the first-generation analog cellular systems such as Advanced Mobile Phone System (AMPS) where the mobile telephone switching office (MTSO) is responsible for overall hand-off decision. In NCHO, the network evaluates the necessary RSSI measurements and hand-off decision.

i e i te is a decentralised hand-off protocol and distributes the hand-off decision process. In order to reduce the load of the network, the mobile subscriber is responsible for making RSSI measurements and send them periodically to a cell-site in MAHO. Based on the received RSSI information data, the cell-site or the mobile switching centre (MSC) decides when to hand-off. MAHO is used in the second generation digital cellular system such as Global System for Mobile Communications (GSM). In a digital cellular system, the mobile receiver is capable of monitoring the signal strength of the control channels of the neighbouring cells during a call. When the received signal strength of its voice channel is weak, the mobile subscriber unit can request a hand-off with a candidate hand-off site. MSC then chooses the proper neighbouring cell for hand-off to take place based on the signal strengths of both forward and reverse control channels. Compared to NCHO, this mechanism has more distributed control, thereby helping to improve the overall hand-off delay, typically one second.

S t S is often used in conjunction with MAHO. Rather than immediately terminating the connection between a mobile subscriber and a serving cell-site in the process of hand-off, a new connection is established first between the mobile subscriber and a new target cell-site, while maintaining the old communication link. Only after the new communication link can stably transmit data, the old link is released. Thus, SHO algorithm is a make-before-break type mechanism. This mechanism helps in ensuring the service continuity which is, however, at the expense of using more resource during the hand-off process as two connections are established simultaneously. The soft hand-off algorithm is generally applied to a CDMA digital cellular system which uses the same radio frequency in all the cells. There is no need to change from one frequency to another frequency but change from one code to another code during hand-off.

In the *i e t e* algorithm, the mobile subscriber unit controls and makes complete decisions on hand-off. A mobile subscriber keeps on measuring signal strength from all the neighbouring cell sites. If the mobile subscriber finds that there is a new cell-site with a stronger signal than that of an old serving cell-site, it may consider to hand-off from the old cell to the new cell, after reaching a defined signal threshold level. MCHO is the highest degree of hand-off decentralisation, thereby enabling it to have a very fast hand-off speed, typically of the order of 0.1 second only. MCHO extends the role of the mobile subscriber by giving overall control to it. The mobile subscriber and cell-site both make the necessary measurements and the cell-site sends the measurement data to the mobile subscriber. Then, the mobile subscriber decides when to hand-off based on the information gained from the cell-site and itself. MCHO algorithm is used in Digital European Cordless Telephone (DECT) system.

7.12.5 Problems while Handing Off

Hand-off is a critical feature in cellular systems, and therefore, hand-off algorithms are under extensive research. Issues in hand-off algorithm or its parameters may lead into call drops with a direct effect on subscriber satisfaction. This is particularly critical for 3G systems, where high data rate subscribers will be prime candidates for being dropped.

Unnecessary hand-offs lead to degraded call quality and waste of signaling capacity. The number of times of cell boundary crossings increases because smaller cells are deployed in order to meet the demands for increased capacity. Since the cell-size is constantly decreasing, it is important for the hand-off algorithm to identify subscribers with different mobility and data-rate characteristics.

Each hand-off requires network resources to reroute the call to the new cell-site. The measurement reports are transmitted periodically from the mobile subscriber to cell-site on the assigned logical contol channel. If the expected number of occurrence of hand-offs are minimised, the switching load minimises as well.

7.12.6 Prioritising Hand-off

Hand-off calls can be allowed at a higher priority than new calls. In order to manage the hand-off requests based on priority, it is necessary to reserve some channels exclusively for this purpose. A popular technique to reserve some channels for handling hand-off requests is the guard channel method. Let N_g be the number of channels reserved for allowing hand-off requests, and N_u be the number of unallocated channels. The guard channel method specifies the following criteria for hand-off requests on priority:

– If $N_u > N_g$, allow either a new call request or hand-off request
– If $N_u \leq N_g$, allow a hand-off request only and deny a new call request

The above criteria specified by the guard channel method is a fixed reservation strategy. This, however, reduces the total carried traffic as fewer channels are available for originating calls. If the system knows exactly the number or rate of hand-off requests during any time period, then it can estimate the value of N_g exactly. This will enable to introduce dynamic reservation methods to reserve the desired number of channels for hand-off requests.

Another hand-off algorithm is based on the relative power difference (δ) measurement of a mobile signal received by two cell sites—serving hand-off cell-site and target hand-off cell-site. This algorithm is not based on the absolute received signal strength level, but on a relative (power difference) measurement. The hand-off occurs depending on a preset value of δ which could be +ve or –ve. For example, the following situation can occur:

$\delta > 3$ dB	request for a hand-off
1 dB $< \delta <$ 3 dB	prepare for a hand-off
-3 dB $< \delta <$ 0 dB	monitor the received signal strength
$\delta < -3$ dB	no hand-off request

In many cases, a two-level hand-off-algorithm is used which provides more opportunity for a successful hand-off.

Key Terms

- Amplitude Modulation (AM)
- Amplitude Shift Keying (ASK)
- Analog modulation
- Analog signal
- Baseband
- Baud rate
- Block error correction code
- Block interleaving
- Bps
- Burst error
- Carrier frequency
- Carrier signal
- Cell coverage
- Cell splitting
- Codeword
- Convolutional code
- Convolutional interleaving
- Cyclic code
- Cyclic Redundancy Check (CRC)
- Data rate
- Dibit
- Digital modulation
- Diversity
- Equalisation
- Error control
- Error correction
- Error-correction code
- Error detection
- Error-detection code
- FEC
- Frequency Modulation (FM)
- Frequency Shift Keying (FSK)
- Full-wave antenna
- Gaussian Frequency Shift Keying (GFSK)
- Gaussian Minimum Shift Keying (GMSK)
- Hand-off
- Interleaving
- Minimum Shift Keying (MSK)
- Modulation
- Modulation Index
- Offset Quadrature Phase Shift Keying (OQPSK)
- Phase Modulation (PM)
- Phase Shift Keying (PSK)
- Quad bit
- Quadrature Amplitude Modulation (QAM)
- Quadrature Phase Shift Keying (QPSK)
- Quantisation
- Repeater
- Sidebands
- Traffic capacity
- Turbo code

Summary

This chapter provides an insight into various system design parameters in order to optimise the radio coverage, received signal quality, and enhanced traffic capacity. An overview of cell-splitting techniques is presented here which enables to meet the varying traffic requirements in specific areas. One of the goals of a cellular system is for the users to remain in touch even as they move through the system. When a user moves from the coverage area defining one cell into that of another, the system must provide the capability for that user to remain in touch with the cellular network. Generally, a large number of small cells, with each cell effectively covering mobile users located in that area, are deployed. Smaller cells mean more frequent hand-offs, which requires greater system resources to support and coordinate. Hand-off is really a localised form of mobility. Different types of viable hand-off mechanisms for maintaining the call continuity across the radio coverage boundaries of the cells for highly mobile users are presented here. In the next chapter, the techniques to access the control channel by multiple mobile users avoiding any collision are discussed.

Important Equations

- $r_2 = (P_{t2} / P_{t1})^{1/4} r_1$ (7.4)
- $P_{tn}(\text{dBm}) = P_{t0}(\text{dBm}) - 12\,(n)$ (7.14)
- $S_{AM}(t) = A_c \; 1 + m_a \cos(2\pi f_m t) \; \cos(2\pi f_c t)$ (7.19)
- $P_{AM} = P_c \; 1 + (m_a^2 / 2)$ (7.20)
- $S_{\;M}(t) = A_c \cos \; (2\pi f_c t) + + m_f \sin(2\pi f_m t)$ (7.21)
- $m_f = (k_f A_m) / f_m = \Delta f / f_m$ (7.22)
- $S_{QPSK}(\text{t}) = 1/\sqrt{2} \;\; I(\text{t}) \cos(2\pi f_c t) - Q(t) \sin(2\pi f_c t)$ (7.32)
- Code rate, $r = k / n$ (7.40)

Short-Answer Type Questions with Answers

A7.1 What is meant by coverage hole

Coverage hole is an area within the radio coverage footprint of a wireless communication system in which the received RF signal level is below the specified threshold value. Coverage holes, also called weak received signal spots, are usually caused by physical obstructions such as buildings, hills, dense foliage as well as hard-to-reach areas such as within buildings (indoor), or in valleys or in tunnels.

A7.2 List various techniques available for filling the coverage holes.

The various techniques available for filling the coverage holes are wideband wireless enhancers or repeaters, channelised wireless enhancers or repeaters, passive reflectors, diversity receivers, feedforward cophase combiners, and feedback cophase combiners.

A7.3 What is the importance of an intelligent frequency planning

Intelligent frequency planning means the dynamic assignment of channels to new call requests in such a way so as not to cause any interference to on-going communications. It is needed in order to avoid the interference problems from cochannel and adjacent channel interference. Depending on the active channels at any time, some free channels may be quiet, some may be noisy, and some may be vulnerable to channel interference. These factors should be considered in assigment of voice channels to a particular mobile subscriber among a set of available channels. This is an essence of intelligent frequency planning.

A7.4 How does the type and installation of cell-site antenna system play a critical role in determining the impact of system interference
The design of an antenna radiation pattern—omnidirectional or directional—including downward tilting, enables to confine the radiated signal energy within a small area, thereby reducing interference. Depending on the traffic demand, a strong signal may be needed in some directions and no signal may be needed in some other directions. Proper selection of cell-site location as well as optimum antenna height, depending upon the terrain conditions along with reducing transmitted power, can be more effective in reducing interference while trading off with the radio coverage area.

A7.5 Define cell splitting.
Cell splitting is the process of dividing a larger congested cell into smaller cells, each with its own cell-site with a corresponding reduction in transmitter power and antenna height. With cell splitting, frequency can be reused more frequently in a given service area, thereby increasing the overall system capacity.

A7.6 Why is transmitter power in the uplink direction normally limited
Transmitter power is particularly limited in the reverse direction from mobile subscriber to base station because the mobile subscriber phone equipments have low transmitter power, battery powered, and are small in size.

A7.7 Why do all cells not have uniform size in a practical cellular network
A typical cell distribution in a practical cellular network comprises of large cells in the rural service area of low-traffic density, medium-sized cells in the suburban service area of medium-traffic density, small cells in the town or highways areas of high-traffic density.

A7.8 Contrast the features of linear and continuous-phase digital modulation techniques.
Linear digital modulation is more spectral efficient, but requires a linear power amplifier so as to avoid the signal amplitude variations which may result in intermodulation products. Continuous-phase digital modulation techniques avoid the linearity requirements of RF power amplification at the transmitter. They have quite narrow power spectra but the spectral efficiency is somewhat lower.

A7.9 What is the significance of using a pre-modulation low-pass filter with Gaussian characteristics in GMSK digital-modulation technique
The use of a pre-modulation low-pass filter with Gaussian characteristics with the MSK approach achieves the requirement of spectral containment as well as uniform envelope. In GMSK, the Gaussian filter is used to suppress out-of-band noise and adjacent channel interference. It provides high spectrum efficiency and a constant amplitude envelope that allows class C power amplifiers to be used, minimising power consumption.

A7.10 What are the two most important metrics for evaluating the performance of a hand-off procedure
Call-blocking probability and forced-call termination probability are the two most important metrics for evaluating the performance of a hand-off procedure. The call blocking probability is the probability of blocking a new call request. The forced call termination probability is the probability of dropping an active call due to hand-off failure. The prime objective of a hand-off procedure is not to increase call-blocking probability significantly while decreasing forced call termination probability.

A7.11 Differentiate between inter-cell hand-off and intra-cell hand-off.
In inter-cell hand-off, the present serving and the new target cells are different cells (even if they are available on the same cell site). In intra-cell hand-off, the present serving and the new target cells are one and the same cell and only the occupied channel is changed during the hand-off. The need to change an occupied channel may arise when it may be interfered or affected by fading, with a new clearer or least fading channel.

A7.12 Why is hand-off always implemented on the voice channel not control channel
In cellular systems, once a call is established, the set-up or control channel is not used during the call

duration. Therefore, hand-off is always implemented on the voice channel.

A7.13 What are the parameters of the signal in the voice channel monitored and assessed in order to decide when a hand-off may be necessary

During a call, the system parameters such as the received signal level, the received signal-to-noise ratio, bit error rate, frame error rate, received speech quality level, estimated distance between the mobile subscriber and the cell-site in the present serving cell are monitored and assessed in order to decide when a hand-off may be necessary.

A7.14 How is queuing for hand-off more important

Queuing for the hand-off is more important because call drops upset mobile subscribers more than call blockings of initiated calls. Queuing hand-off calls prioritisation scheme queues the hand-off calls when all the available channels are occupied in a cell-site. When a channel is released, it is assigned to one of the hand-off calls in the queue.

A7.15 Define two peculiar situations where hand-offs are necessary but cannot be made.

Firstly, when the mobile subscriber is located at a signal-strength hole within a cell but not at the boundary, the call must be kept in the old frequency channel until it is dropped as the result of an unacceptable received signal level. Secondly, when the mobile subscriber approaches a cell boundary but no channels in the new cell are available, the new cell must reassign one of its frequency channels within a reasonably short period.

Self-Test Quiz

S7.1 The cell-site transmitter power increases by 3 dB. It means it is increased by
(a) two times (c) four times
(b) three times (d) ten times

S7.2 In a flat operating terrain, doubling the cell-site antenna height results into an increase in gain by
(a) 3 dB (c) 6 dB
(b) 4 dB (d) 8 dB

S7.3 Interference on voice channels usually causes
(a) missed calls (c) dropped calls
(b) blocked calls (d) cross talk

S7.4 Cell splitting allows a system to grow by replacing large cells with smaller cells. The minimum cochannel reuse ratio between co-channel cells
(a) remains same (c) decreases
(b) increases (d) becomes insignificant

S7.5 Cell splitting involves the changes in cellular architecture with respect to
(a) frequency reuse plan
(b) channel assignment
(c) coverage area of a split cell
(d) all of the above

S7.6 The radius of split cells is one half of the radius of the original cell. The coverage area of a split cell is ______ the coverage area of the original cell.
(a) equal to (c) one-fourth of
(b) one half of (d) one-tenth of

S7.7 The cell-site transmit power of a split cell with one half of the radius of the original cell must be reduced by ______ than that of original cell.
(a) 3 dB (c) 9 dB
(b) 6 dB (d) 12 dB

S7.8 If the cell splitting is done twice with each split cell having a radius one-half of its previous cell, then the traffic load would increase by
(a) 16 times (c) 4 times
(b) 8 times (d) 2 times

S7.9 ______ type of digital modulation is used more in fixed than in mobile radio communication because it requires high SNR value.
(a) QPSK (c) GMSK
(b) QAM (d) BFSK

S7.10 A modem uses 4 different amplitudes and 16 different phase angles. To transmit each symbol, ________ of bits are transmitted.

(a) 4 (c) 8
(b) 6 (d) 16

S7.11 The North American TDMA digital cellular standard transmits at 24.3 ksps using DQPSK. The channel data rate is ________.

(a) 12.15 kbps (c) 48.6 kbps
(b) 24.3 kbps (d) 97.2 kbps

S7.12 The GSM cellular standard uses GMSK digital-modulation scheme in a 200-kHz channel, with a channel data rate of 270.833 kbps. The bandwidth efficiency is

(a) 1.35 kbps/Hz (c) 2.70 kbps/Hz
(b) 1.35 bps/Hz (d) 2.70 bps/Hz

S7.13 ________ is considered one of the main probable reasons of call drops.

(a) Cochannel interference
(b) External noise
(c) Hand-off
(d) Channel assignment

S7.14 When lower voice quality is acceptable, the radio capacity

(a) remains unaffected (c) increases
(b) decreases (d) approaches to zero

S7.15 The spectral efficiency of GMSK ($B \times T_b =$ 0.3) digital-modulation technique used in GSM standard is ________ bps/Hz.

(a) 1.62 (c) 0.72
(b) 1.35 (d) 0.67

S7.16 An advantage of the ________ is that a call communication link uses only one channel at any moment.

(a) hard hand-off (c) softer hand-off
(b) soft hand-off (d) Intra-cell hand-off

S7.17 ________ is a decentralised hand-off protocol and distributes the hand-off decision process.

(a) Network-controlled hand-off
(b) Mobile-assisted hand-off
(c) Soft hand-off
(d) Mobile-controlled hand-off

S7.18 NCHO protocol is used in the ________ cellular networks.

(a) AMPS (c) USDC
(b) ETACS (d) GSM

S7.19 The overall hand-off delay in mobile-assisted hand-off algorithm is typically

(a) 5 -10 seconds (c) 1 second
(b) 2-3 seconds (d) less than 1 second

Answers to Self-Test Quiz

S7.1 (a); S7.2 (c); S7.3 (d); S7.4 (a); S7.5 (d); S7.6 (c); S7.7 (d); S7.8 (a); S7.9 (b); S7.10 (b); S7.11 (c); S7.12 (a); S7.13 (c); S7.14 (c); S7.15 (b); S7.16 (a); S7.17 (b); S7.18 (a); S7.19 (c)

Review Questions

Q7.1 Suggest the ways to increase the radio coverage of a cell.

Q7.2 List the techniques of coverage hole fillers. What are the typical applications of leaky feeders?

Q7.3 What measures can be taken to reduce interference in a cellular system?

Q7.4 How can the traffic capacity be increased?

Q7.5 What is meant by cell splitting? How does cell splitting affect the system design? Does the value of the frequency-reuse ratio ($q = D/R$) change due to cell splitting?

Q7.6 Differentiate between permanent and dynamic cell-splitting techniques.

Q7.7 What should be the general criteria to split the cells in order to prevent dropped calls?

Q7.8 How is leaky waveguide different from a slot antenna?

Q7.9 Compare the spectral efficiencies of IS-136, GSM, and IS-95 cellular standards.

Q7.10 List four significant factors which influence the choice of speech coders in mobile communication systems.

Q7.11 What is a trellis in the context of a convolutional code?

Q7.12 State the conditions when hand-off is needed. Define the two-level hand-off algorithm.

Q7.13 Describe briefly the various types of hand-offs. Explain the procedure of intersystem hand-off with a suitable illustration.

Q7.14 Why is hand-off always implemented on the voice channel, and not set-up channel?

Q7.15 What are the disadvantages associated with hand-offs taking place at a very high rate?

Q7.16 Differentiate between a blocked call and a dropped call.

Analytical Problems

P7.1 In a cellular system, the minimum acceptable received signal threshold level is lowered by 12 dB. Prove that the coverage area increases by four times. Assume that the transmitted power and other factors remain unchanged in a mobile radio environment.

P7.2 In a cellular system, the minimum acceptable received signal level is required to be as strong as 3 dB more than the initial value. Cell-site transmitted power and other factors governing cell range remains the same. Show that the cell radius reduces to 0.84 times and the corresponding coverage area reduces to 0.7 times than the old values.

P7.3 The transmitted power of a cell-site is increased by 6 dB. For the same minimum acceptable received signal power and all other factors remaining unchanged, compute the percentage increase in the coverage area. Assume path-loss exponent value as 4.

P7.4 The transmitted power of an original cell before splitting is 200 watts. It is split into smaller cells in such a way that the radius of the split cells is exactly one-half of the radius of the original cell. What should be the transmitter power of each split cell so as to maintain the same reception quality? Assume the path-loss exponent in a mobile environment as 4.

P7.5 For an identical received power at the boundaries of original larger cell with a 3-km radius and the new split cell with a 1.5-km, radius, compute the required transmitter power at the cell-site of the split cell if the transmitter power at the cell-site of the original larger cell is 50 W. Assume the path-loss exponent as 4.

P7.6 An original large regular hexagonal cell is split into four identical small regular hexagonal cells.

(a) What would be the approximate radius of the split cell if the radius of the original cell is 5 km?
(b) Find the per cent reduction in coverage area of each split cell in comparison to coverage area of the original cell.
(c) How much capacity is increased if 120 voice channels are assigned in the original cell as well as each split cell?

P7.7 In a US AMPS cellular system, a service provider is allocated a total spectrum of 10 MHz. Out of this, 1 MHz spectrum is reserved for signaling and control channels. If 7-cell reuse pattern is used, calculate

(a) the radio capacity of a cluster
(b) the radio capacity of a cell

P7.8 An example system of cell splitting is shown in Fig. 7.31, in which the cell-sites are located at the corners of the cells. The cell-site marked as c1 is split into six new smaller split cell-sites marked as 'a', 'b', 'c', 'd', 'e', 'f' in such a way so as to preserve the frequency reuse plan of the system.

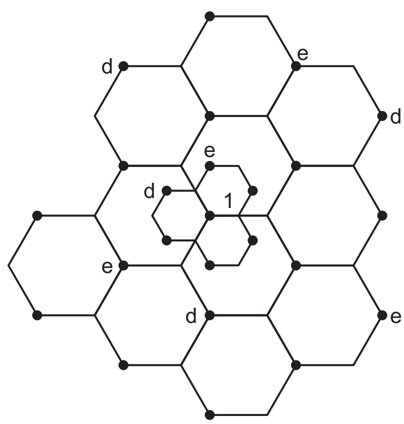

Fig. 7.31 | *For Problem P7.8*

(a) What is the radius of the split cell as compared to the original cell c1?
(b) If the transmitter power of the cell c1 is 40 dBm, what should be the transmitter power of the split cells in order to maintain an identical received power levels at the boundaries of two different-sized cells.
(c) How should the regrouping of channels be carried out so as to maximise the overall system capacity?
(d) Which of the two—larger or smaller—cells should be dedicated for high-speed traffic and why?
(e) What additional measures can be taken to limit the radio coverage of newly formed split cells?

P7.9 Consider a cellular system that employs omni-directional antennas at cell-sites. In order to increase system capacity, each cell is split into 4 smaller cells having a radius that is one-half of the radius of the original cell.
(a) How should the transmitter power of a split cell be changed? Assume the path-loss exponent in a mobile operating environment as 3.
(b) How is it comparable with that of the system if the path-loss exponent is 4?
(c) If the transmitter power of the original cell is 40 dBm, compute the transmitter power of split cells in part (a) and (b) respectively.

P7.10 Consider the cellular system as shown in Fig. 7.32. Assume each cell-site uses 60 channels, regardless of cell size.

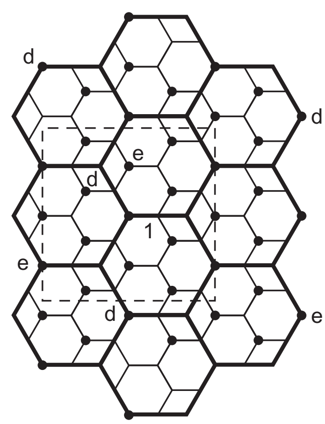

Fig. 7.32 | *For Problem P7.10*

If each original cell has a radius of 4 km and each microcell has a radius of 2 km, find the number of channels contained in a 12 km by 12 km square centered around the cell 'c1' under the following conditions:
(a) without the use of split cells;
(b) when the marked split cells as shown in Fig. 7.32 are used; and
(c) if all the original cell sites are replaced by split cells. Assume cells on the edge of the square to be contained within the square.

P7.11 A cell-site transmits 10 W of power. The maximum range obtained is 5 km. If the power is increased to 20 W, calculate the new coverage area.

P7.12 In a typical cellular system, the radio coverage area is 10 km^2 corresponding to acceptable received signal level of –94 dBm. If the threshold received signal level is lowered to –100 dBm, show that the radio coverage area is doubled, other parameters remaining same.

P7.13 A US digital cellular system sends symbols at 24 ksps at transmission frequency of 900 MHz. If the speed of the mobile is 80 kmph, show that 152 symbols are sent after the equaliser is trained.

P7.14 The required number of taps in the equaliser are determined by the ratio of path delay to bit duration. Compute the required number of taps in the equaliser for a GSM system working at 270 kbps channel data rate which is capable of correcting 16 μs delay to combat intersymbol interference.

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8

Multiple Access Techniques

The fundamental physical resource in wireless communications is the available radio spectrum. The advancements in wireless communications means finding more efficient ways of utilising the radio spectrum. The multiple access techniques describe the general approach to sharing the physical resources of a wireless medium by a large number of mobile subscribers at the same time in different locations. This chapter describes various multiple access techniques such as FDMA, TDMA, SSMA, SDMA, and hybrid multiple access that are used in analog and digital wireless communications systems. Their relative advantages and disadvantages have been outlined here. An overview of packet radio multiple access technique including CSMA is presented in the end.

8.1 INTRODUCTION

For any wireless service, only a fixed limited finite amount of radio spectrum (or number of channels) is available to provide simultaneous communication links to many subscribers in a given service area. Multiple access techniques are used to achieve high subscriber capacity by sharing the available limited spectrum among many subscribers simultaneously, while maintaining the desired quality of communications. There are four basic forms of multiple access techniques applied to wireless communications, depending on which particular resource is exploited: Frequency-Division Multiple Access (FDMA), Time-Division Multiple Access (TDMA), Spread-Spectrum Multiple Access (SSMA), and Space-Division Multiple Access (SDMA). The objective of all these multiple access strategies is to maximise the spectrum utilisation.

The choice of an access method will have a great impact on the capacity and quality of service provided by a wireless network. In practice, most wireless communication systems are a combination of one or more of these multiple access strategies. There are many instances in multiple access communications, in which a mobile subscriber is required to send a packet of information to the cell-site at a random instant in time, leading to contention-based packet radio protocols such as ALOHA and CSMA. The multiple-access packet radio protocols, also known as the medium access control sublayer protocols, are primarily a set of rules that communicating mobile subscribers need to follow.

8.2 FREQUENCY DIVISION MULTIPLE ACCESS

Frequency Division Multiple Access (FDMA) refers to sharing the available radio spectrum by assigning specific frequency channels to subscribers either on a permanent basis or on a temporary basis. The differentiation between the carrier frequencies of the forward channels (also called downlink-communication between the cell-site and mobile subscribers) and reverse channels (also called uplink-communication between the mobile subscribers and the cell-site) is an important design parameter related to FDMA technique.

If the forward channels and reverse channels use different carrier frequencies that are sufficiently spaced, the duplexing scheme is referred to as FDD. The FDD technique is mostly used in macrocellular communication systems designed for radio coverage of several kilometres. The base station dynamically assigns a different carrier frequency to each active mobile subscriber. In order to adjust and maintain the transmission and reception frequencies, a frequency synthesiser is used at the base station and the mobile station. The concept of FDMA is shown in Fig. 8.1.

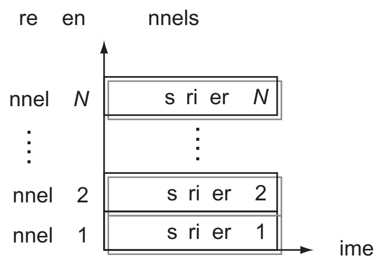

Fig. 8.1 *The concept of FDMA*

In FDMA, the available radio spectrum is divided into a set of continuous frequency channels labeled 1 through N, and the frequency channels are assigned to individual mobile subscribers on a continuous-time basis for the duration of a call. FDMA bandwidth structure is illustrated in Fig. 8.2.

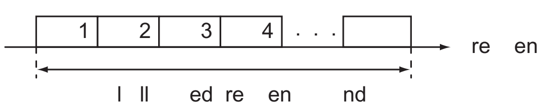

Fig. 8.2 *FDMA bandwidth structure*

Fig. 8.3 shows the basic structure of a FDMA system, consisting of a cell-site (CS) and many mobile subscribers. There is a pair of simplex channels for the communication wireless link between the CS and the mobile subscribers. The paired channel is called *forward channel* (downlink) and *reverse channel* (uplink). A forward channel is used to transfer data from the cell-site to the mobile subscriber and a reverse channel is used to transfer data from the mobile subscriber to the cell-site. Different frequency channels are assigned to different mobile subscribers. Each pair of communicating mobile subscribers is assigned different frequency channels to enable full duplex communication.

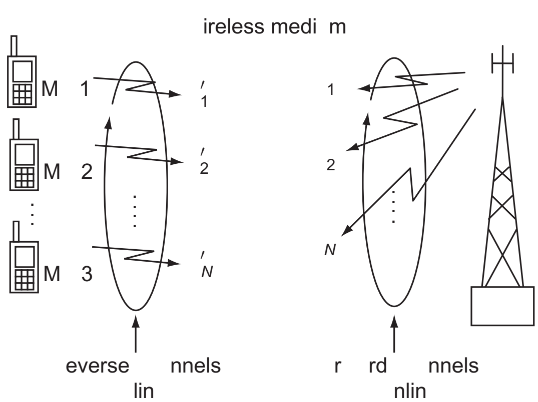

Fig. 8.3 *The basic structure of an FDMA system*

FDMA has been widely adopted in all first-generation analog cellular systems for handheld and vehicle-installed mobile subscribers. A duplex spacing is used between the forward and reverse channels. The structure of forward and reverse channels in FDMA is shown in Fig. 8.4.

The frequency bandwidth allocated to each mobile subscriber is called the subband B_c. If there are N channels in a FDMA system, the total bandwidth B_t is equal to $N \times B_c$. A guard band W_g is used to minimise adjacent channel interference between two adjacent channels, as shown in Fig. 8.5.

To ensure acceptable signal quality performance, it is important that each frequency channel signal be kept confined to the assigned channel bandwidth. Otherwise, there may be adjacent channel interference which can degrade signal quality. In both forward and reverse channels, the signal transmitted must be kept confined within its assigned channel bandwidth, and the out-of-band signal energy causes negligible interference to the

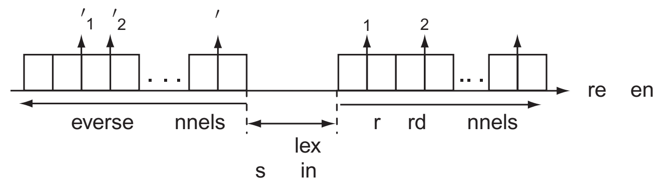

Fig. 8.4 *The structure of forward and reverse channels in FDMA*

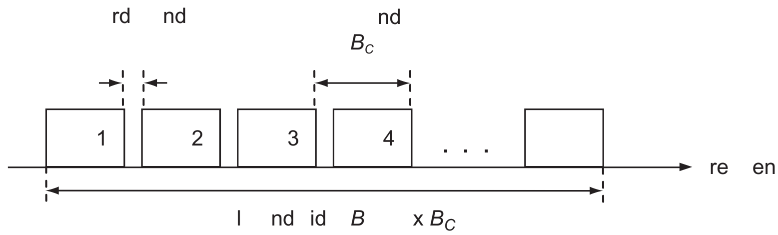

Fig. 8.5 *Guard band in FDMA channels*

subscribers using adjacent channels. In order to minimise adjacent channel interference, two design measures are usually considered:

– The power spectral density of the modulated signal is controlled so that the power radiated into the adjacent band is at least 60 to 80 dB below that in the desired band. This requirement can be achieved with the use of highly selective filters in the system design.
– Usually, it is extremely difficult to achieve the desired filter characteristic so as not to cause adjacent channel interference. Guard bands are inserted as buffer frequency zones in adjacent channels.

If a large number of mobile subscribers can operate satisfactorily within the allocated radio spectrum then the multiple-access system is said to be highly spectrally efficient. In general, the spectral efficiency in FDMA systems depends on how closely the individual channels (frequency subbands) can be assigned. There are several factors that limit the adjacent channel spacing, the most important of which is adjacent channel interference (ACI). The impact of ACI is illustrated in Example 8.1.

EXAMPLE 8.1 **Impact of ACI in FDMA system**

(a) What is the difference between the received signal strength of two mobile subscribers located at 10 m and 1 km away from a cell-site in an open area?
(b) Explain the effects of shadow fading on the difference in the received signal strength of two mobile subscribers obtained in part a).
(c) What would be the impact if the two mobile subscribers were operating in two adjacent channels? Assume out-of-band radiation that is 40 dB below the main lobe.
(d) Suggest the measure taken to overcome this problem in FDMA cellular systems.

Solution

(a) *To determine difference in received signal strengths of mobile subscribers*

Distance of MS1 from the cell-site, $r_1 = 10$ m (given)

Distance of MS2 from the cell-site, $r_2 = 1$ km or 1000 m (given)

Operating environment = Open area

Step 1. *In an open area environment, free-space propagation conditions exist*
The value of propagation path-loss exponent, $\gamma = 2$

Step 2. *In free-space propagation, the received signal strength decays at the rate of 20 dB per decade of distance.*

Step 3. *The difference between the received signal strength of two mobile subscribers located at r_1 and r_2, $\Delta P_r = 20 \log (r_2/r_1)^2$*
Therefore, $\Delta P_r = 20 \log (1000/10)^2$
Hence, $\Delta P_r = 80$ dB

(b) *The effects of shadow fading*
In addition to the decrease in the received signal strength value with distance, the multi-path and shadow fading due to the near–far problem because of large difference in the distances of two mobile subscribers, in radio channels cause received signal fluctuations of the order of typically 20 dB.
Therefore, the difference in the received signal levels from these two mobile subscribers may exceed even 100 dB.

(c) *Impact of operation in adjacent channels*
Out-of-band radiation below the main lobe= 40 dB (given)
It implies that out-of-band radiations may exceed the signal strength of the desired signal by almost (100 dB – 40 dB =) 60 dB

(d) *The measure taken to overcome the problem in FDMA cellular systems*
To handle the near–far problem in FDMA cellular systems, the following different measures may be adopted in the system.

– Channel assignment should be done in such a way so that the frequencies in each cell are grouped as far apart as possible from each other.
– Guard bands should be included in the frequency channel to further reduce adjacent channel interference. This, however, has the effect of reducing the overall spectrum efficiency.
– The transmitter power of the mobile subscribers should be controlled so as not to cause interference to other transmissions in the cell.

In an FDMA system, many channels share the same transmitting antenna at the base station. The transmitter RF power amplifiers or the transmitter multichannel power combiners are nonlinear devices when operated at or near saturation signal levels for maximum power efficiency. The nonlinearities cause spreading of the signal in the frequency domain and generate intermodulation frequencies which are undesirable harmonics. Harmonic frequencies generated within the operating frequency band cause interference to other subscribers active in the same wireless system at that time. Harmonic frequencies generated outside the operational frequency band cause interference to other wireless services operating in those adjacent bands.

The first-generation analog cellular communication systems use FDMA/FDD technique, with speech signals being transmitted over the forward or reverse channels using frequency modulation scheme. The data control functions are performed digitally by means of frequency-shift keying modulation scheme for data transmission. A useful feature of FDMA systems is that the radio transmission takes place over a narrow channel of bandwidth (B/N) Hz for each mobile subscriber. Due to narrowband transmissions, flat fading may be experienced by the signal.

EXAMPLE 8.2 FDMA/FDD in AMPS

Illustrate the concept of FDMA/FDD system commonly used in First Generation (1G) analog cellular communication systems such as AMPS.

Solution In FDMA/FDD systems, forward and reverse channels use different carrier frequencies, and a fixed subchannel pair is assigned to a subscriber during the communication session.

Fig. 8.6 shows the FDMA/FDD system commonly used in first generation analog cellular systems. At the receiving end, the mobile unit filters the designated channel out of the composite signal received.

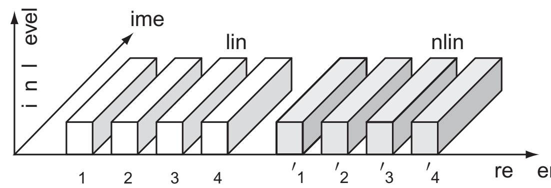

Fig. 8.6 | *FDMA/FDD concept*

The Advanced Mobile Phone System (AMPS) is based on FDMA/FDD. As shown in Fig. 8.7, the AMPS system allocates 30 kHz of channel bandwidth for each uplink (824 MHz–849 MHz) and downlink (869 MHz–894 MHz) frequency band.

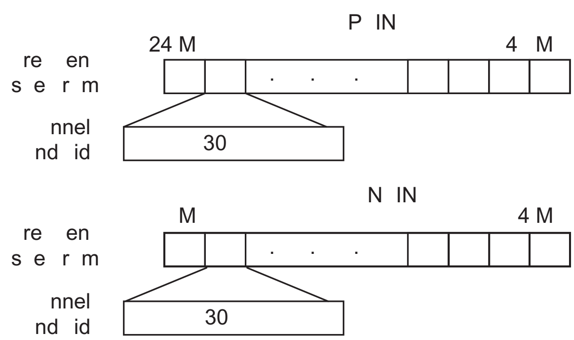

Fig. 8.7 | *FDMA/FDD in AMPS*

Some of the salient features of the FDMA/ FDD system concept are given here.

- During the call, a mobile subscriber occupies two simplex channels, one each on the uplink and downlink, for full-duplex communication.
- The two simplex channels are spaced by fixed duplex spacing. For example, duplex spacing in AMPS is (869 MHz–824 MHz = 45 MHz).
- When a call is terminated, or when hand-off occurs, the occupied channels are released which can be used by other mobile subscribers in the system.
- Multiple or simultaneous mobile subscribers are accommodated in AMPS by allocating each calling or called mobile subscriber a dedicated channel.
- Voice signals are sent on the forward channel from the base station to the mobile user, and on the reverse channel from the mobile user to the base station.
- In AMPS, analog narrowband frequency modulation technique is used to modulate the carrier.

The number of channels, N that can be simultaneously supported in an FDMA system is given by

$$N = (B_t \quad 2 B_g) / B_c \tag{8.1}$$

where B_t is the total spectrum allocation, B_g is the guard band allocated at the edge of the allocated spectrum band, and B_c is the channel bandwidth.

B_t and B_c may be specified in terms of simplex bandwidths where it is understood that there are symmetric frequency allocations for the forward band and reverse band.

EXAMPLE 8.3 | Number of channels in AMPS

A US AMPS analog cellular system is allocated 12.5 MHz for each simplex band. If the guard band at either end of the allocated spectrum is 10 kHz, and the channel bandwidth is 30 kHz, find the number of channels available in an FDMA system.

Solution

Allocated spectrum, $B_t = 12.5$ MHz (given)

Allocated guard band, $B_g = 10$ kHz (given)

Channel bandwidth, $B_c = 30$ kHz (given)

The number of channels available in the FDMA system is given as

$$N = (B_t - 2\,B_g)\,/\,B_c$$

Or, $N = (12.5 \times 10^6 - 2\,(10 \times 10^3)\;\;/\,(30 \times 10^3)$

Or, $N = 416$

Hence the number of channels available in an FDMA system is 416 channels

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EXAMPLE 8.4 | Number of simultaneous links in an FDMA system

A cellular system operator is allocated a total spectrum of 5 MHz for deployment of an analog cellular system based on the FDMA technique, with each simplex channel occupying 25 kHz bandwidth. Compute the number of simultaneous calls possible in the system.

Solution

Total spectrum allocated = 5 MHz (given)

Channel bandwidth = 25 kHz (given)

Step 1. *To determine number of simplex channels*

Number of simplex channels = Total spectrum allocated / Channel bandwidth

Number of simplex channels = 5 MHz / 25 kHz = 200

Step 2. *To determine number of duplex channels*

Number of simplex channels in a duplex channel = 2

Therefore, number of duplex channels = 200 / 2 = 100

Step 3. *To compute the number of simultaneous calls*

Hence, in a given analog cellular FDMA system, 100 full-duplex communication links can be established simultaneously as each link requires two simplex channels (one for uplink and another for downlink) or one duplex channel.

Therefore, the number of simultaneous calls = 100 calls

The FDMA channel carries only one dedicated communication link at a time. After the assignment of a voice channel, the base station and the mobile subscriber transmit simultaneously and continuously. If the assigned channel is not in use then it remains idle and cannot be used by other mobile subscribers. This is clearly wastage of spectrum resource. The utilisation of a channel during free time is essential to increase system capacity.

FDMA is usually implemented in narrowband systems. The bandwidths of FDMA channels are relatively narrow (for example, 30 kHz in AMPS) as each channel supports only one communication link per carrier. The symbol time of a narrowband signal is large as compared to the average delay spread. This implies that the amount of intersymbol interference is also low. So there may not be any requirement to implement equalisation in FDMA narrowband systems which is certainly an advantage.

Facts to Know!

Cable television is transmitted using FDMA over coaxial cable. Each analog television signal utilises 6 MHz of the 500 MHz bandwidth of the cable.

The complexity of FDMA wireless communication systems is lower as compared to that of TDMA systems. Due to continuous transmission in FDMA systems, fewer bits for synchronisation and framing are needed for overhead purposes as compared to TDMA. FDMA requires tight RF filtering to minimise adjacent channel interference. Therefore, there is a need to use costly bandpass filters to eliminate spurious radiations at the base stations. The FDMA mobile subscriber equipment uses RF duplexers since both the transmitter and receiver operate at the same time using a common antenna. This results in an increase in the cost of FDMA subscriber units and base stations. Because of the single channel per carrier design, FDMA systems have higher cell site system costs.

8.3 TIME-DIVISION MULTIPLE ACCESS

Time-division multiple access (TDMA) technique refers to allowing a number of subscribers to access a specified channel bandwidth on a time-shared basis. TDMA systems divide the carrier channel bandwidth into time slots, and in each time slot only one subscriber is allowed to either transmit or receive. TDMA utilises the digital technology with more efficient and complex strategies of sharing the available spectrum among a number of subscribers simultaneously. In TDMA systems, number of subscribers share the same frequency band by taking their assigned turns in time for transmission or reception.

Facts to Know!

If a user has no data to transmit during the assigned time slot, the frequency channel remains idle in TDMA.

The major advantage of the TDMA is the flexibility of its digital format which can be buffered and multiplexed efficiently, and assignments of time-slots among multiple subscribers which are readily adaptable to provide different access rates. With TDMA, a base-station controller assigns time slots to subscribers for the requested service, and an assigned time slot is held by a subscriber until it releases it. The receiver synchronises to the incoming TDMA signal frame, and extracts the time slot designated for that subscriber. Therefore, the most critical feature of TDMA operation is time synchronisation.

In TDMA, one carrier channel is used by several subscribers, and each subscriber is served in a round-robin method. The cell-site assigns different time slots to different subscribers. Let there be N number of time slots in a TDMA frame. Each subscriber occupies a cyclically repeating time slot which reoccurs in every frame periodically. The transmission in a TDMA system for any subscriber is noncontinuous and data is transmitted in a buffer-and-burst method. The splitting of a single carrier channel into several time slots and distribution of time slots among multiple subscribers is shown in Fig. 8.8.

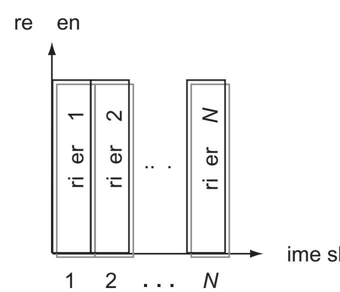

Fig. 8.8 *The concept of TDMA*

A TDMA system may operate in either of two modes:

- *TDMA/ DD mode* The forward and reverse channel frequencies differ.
- *TDMA/TDD mode* The forward and reverse channel frequencies are same.

In TDMA/FDD systems, the carrier frequencies are different but frame structures are same for the forward and reverse channels. In general, TDMA/FDD systems intentionally induce delay of several time slots between the forward and reverse time slots for a particular subscriber. This avoids the need of duplexers in the subscriber unit. The illustration of forward and reverse channels in a TDMA/FDD system employing the similar frame and time slot structure is given in Fig. 8.9.

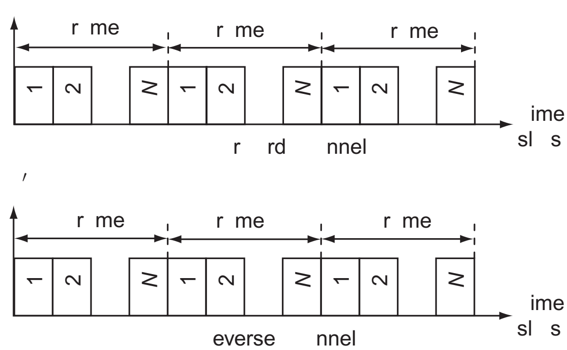

Fig. 8.9 *Structure of forward and reverse channels in a TDMA/FDD system*

In TDMA, a carrier channel is divided into N number of time slots. These time slots are allocated for each subscriber to transmit and receive information. The number of distinct consecutive time slots is called a frame before these time slots are repeated. Each frame of the TDMA structure contains N number of time slots of equal duration. Information data is transferred and received in the form of TDMA frames. The transmission rate for a digital TDMA channel is typically N times higher than that required for a single channel. The bit-wise structure of each time slot is different in different types of TDMA systems. Typically, the bits contained in each time slot of a TDMA frame are divided into two major functional groups:

Signalling and Control Data Bits These bits perform the functions which assist the receiver in performing some auxiliary functions such as synchronisation and frame error rate. Specifically, the synchronisation bits in a time slot enable the receiver to recover sinusoidal carrier essential for coherent detection. The frame error bits are used to estimate the unknown impulse response of the wireless channel, which is needed for decoding the received signal.

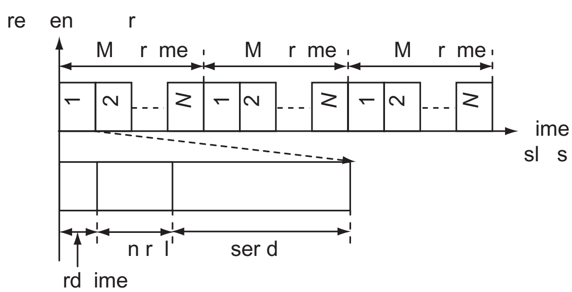

Fig. 8.10 *Typical frame structure of TDMA*

Traffic Data Bits These bits represent digitised speech bits or any other forms of information-bearing data bits.

Fig. 8.10 shows a typical frame structure of TDMA. The guard time between the time slots helps in minimising the interference due to propagation delays along different radio paths in the wireless channel.

Digital data encoding and digital-modulation schemes are used with TDMA. The transmission from various subscribers is interlaced into a uniformly repeating TDMA frame structure. As shown in Fig. 8.11, a TDMA frame consists of a preamble, an information data field, and tail bits. The information data field of a frame consists of a number of time slots.

In a TDMA frame, the preamble contains the address and synchronisation data that is used by both the base station and the subscribers to identify each other. Tail bits and guard bits allow synchronisation of the receivers between different time slots and frames. Various TDMA-based cellular standards such as USDC, GSM have different TDMA frame structures.

In a TDMA system, the communication channels essentially consist of many time slots, which makes it possible for one frequency carrier channel to be efficiently utilised by many mobile subscribers. Each mobile subscriber utilises a different time slot. The basic structure of a TDMA system is shown in Fig. 8.12. The number of mobile subscribers can communicate with the base station simultaneously on designated time slots

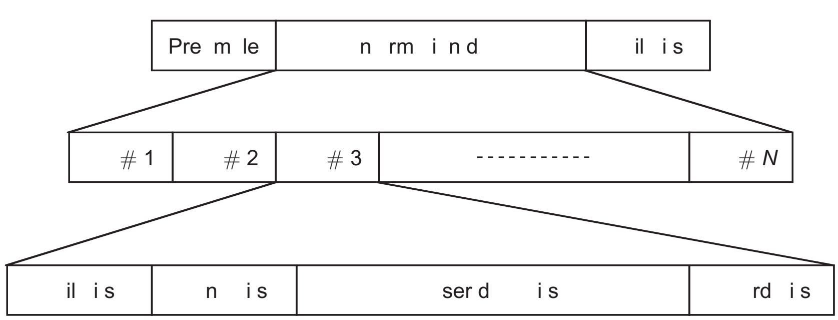

Fig. 8.11 *A TDMA frame and time slot (TS) structure*

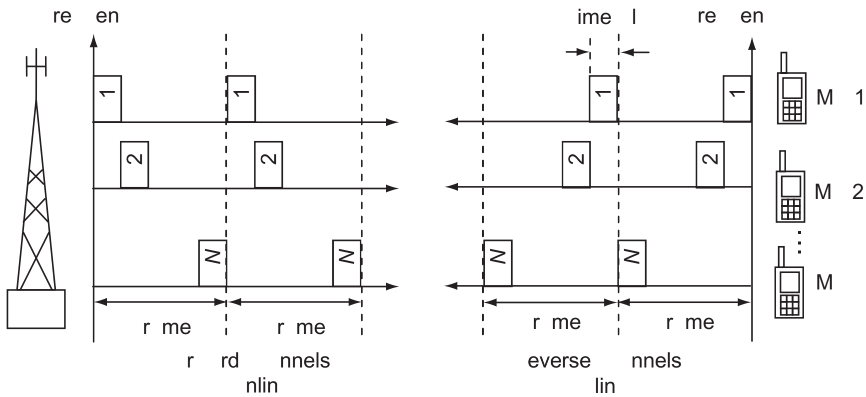

Fig. 8.12 *The basic structure of a TDMA system*

of TDMA frame on the forward and reverse channels. However, the system capacity is limited by the number of time slots per carrier channel and the number of carrier channels allocated to the system.

The features that distinguish TDMA systems from FDMA systems can be broadly classified in two categories:

– In TDMA each subscriber has access to the total bandwidth B_t of the carrier channel, whereas in FDMA each subscriber is assigned only a fraction of the channel bandwidth, that is, $B_c = B_t/N$.
– In TDMA, each subscriber accesses the channel for only a fraction of the time that it is in use and on a periodic regular and orderly basis, with the overall channel transmission data rate being N times the subscriber's required data rate. Whereas in FDMA, each subscriber accesses the channel on a continuous-time basis.

The total number of TDMA time slots that can be provided in a TDMA system is determined by multiplying the number of time slots per carrier channel by the number of channels available and is given by

$$= \quad \times \quad t - 2 \tag{8.2}$$

where N is the total number of TDMA time slots in a TDMA system

m is the number of time slots per carrier channel or the maximum number of TDMA subscribers supported on each carrier channel

B_t is the total allocated spectrum bandwidth in Hz

B_c is the carrier channel bandwidth in Hz

B_g is the guard bandwidth in Hz

Two guard bands, one at the lower end and another at the higher end of the allocated frequency band, are required to ensure that subscribers operating at the edges of the allocated frequency band do not interfere with other wireless communication service operating in an adjacent frequency band.

EXAMPLE 8.5 | Number of simultaneous subscribers in GSM

Consider Global System for Mobile, which is a TDMA/FDD system that uses 25 MHz band for the forward link, which is divided into radio channels of 200 kHz each. If 8 speech channels (time slots) are supported on a single radio channel, find the number of simultaneous subscribers that can be accommodated in GSM, assuming no guard band.

Solution

The allocated spectrum, $B_t = 25$ MHz $= 25 \times 10^6$ Hz (given)

The channel bandwidth, $B_c = 200$ kHz $= 200 \times 10^3$ Hz (given)

Number of speech channels, $m = 8$ per radio channel

The guard bandwidth, $B_g = 0$

The number of simultaneous subscribers that can be accommodated in the GSM system is given as

$N = m \times (B_t - 2B_g) / B_c$

Or, $N = 8 \times (25 \times 10^6 - 2 \times 0) / (200 \times 10^3)$

Or, $N = 8 \times (25 \times 10^6) / (200 \times 10^3) = 1000$ subscribers

Hence the GSM system can accommodate 1000 simultaneous subscribers.

If both forward and reverse channels use the same frequency band but they use alternating time slots in the same frame for full duplex communication, the system is referred to as TDMA/TDD system. In this system, 50% of the time slots in the frame are used for the forward channels and the other 50% of the time slots in the frame are used for reverse channels. Most of the RF components can be shared between the forward and the reverse channels because only one frequency carrier is needed for full duplex operation. The reciprocity of the forward and reverse channels also allows for simultaneous synchronisation as well as exact open-loop power control. TDD techniques are used in systems where minimum interference, low system complexity and low-power consumption are of utmost importance. Thus TDD based systems are quite often used in local area micro- or pico- cellular systems. The structure of forward and reverse channels in a TDMA/TDD system is shown in Fig. 8.13.

In TDMA/TDD based communication system, a simple RF switch is used in the subscriber equipment for use of a single antenna for transmitting and receiving. The common antenna can be connected to the transmitter when a data burst is required to be transmitted (thus disconnecting the receiver from the antenna) and to the receiver for the received signal at another time. An RF switch is different from duplexer which is used in TDMA/FDD based communication system. An RF duplexer is a device with the same functionality as that of an RF switch but is based on RF filter technique.

8.3.1 Salient Features of TDMA Technique

(a) Several subscribers share a single carrier frequency by using non-overlapping time slots. The number of time slots per frame depends upon several factors such as available bandwidth and digital-modulation scheme used. The transmission data rate is quite high as compared to that of in FDMA.

(b) The available bandwidth can be utilised on demand by different subscribers as more than one time slot per frame can be allocated to them. Thus, bandwidth can be supplied to different subscribers on demand by concatenating or reassigning time slots as per assigned priority.

(c) Data transmission is bursty and hence not continuous in time domain. This implies that a subscriber transmitter can be turned off when not in use, thereby saving battery power.

(d) A significant part of the voice call consists of quiet time, when neither the calling nor the called subscriber is speaking. Special signal-processing techniques can be employed to fill the quiet times with data or other voice calls. This leads to considerable improvement in the channel efficiency.

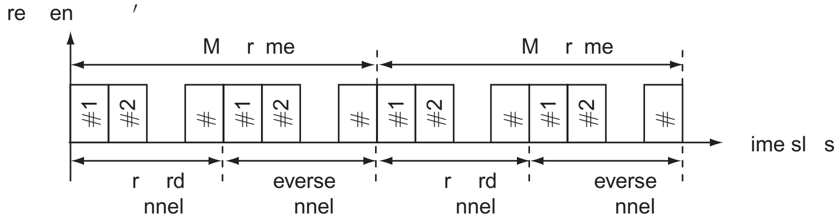

Fig. 8.13 *Structure of forward and reverse channels in a TDMA/TDD system*

(e) The hand-off process is much simpler for a mobile subscriber in a TDMA system due to discontinuous transmissions. During idle time slots, the mobile subscriber can monitor the signal levels from neighbouring base stations and inform the serving base station to assist in hand-off decisions.
(f) Duplexers are not required in the subscriber equipment since the system uses different time slots for transmission and reception. A fast RF switch is sufficient to switch between transmitter and receiver to use the common antenna.
(g) Synchronisation is essential and the guard time or time for synchronisation should be minimum. However, if the transmitted signal at the edges of a time slot is suppressed sharply in order to shorten the guard time, the resulting expanded spectrum will cause interference to adjacent channels.
(h) Large overheads (framing bits) are required because of discontinuous or bursty transmission. A substantial amount of signal processing is needed for matched filtering and correlation detection for synchronising with a time slot.
(i) The effects of the nonlinearity are much reduced since only one RF carrier is present at any time in the channel.
(j) The cell-site hardware can be significantly simplified because the same transmitter/receiver pair is shared between multiple sessions.
(k) The TDMA system can accommodate the transmission of source-channel encoded digital data alongside digitised speech.
(l) TDMA systems use power control to handle the near–far interference problem. Due to the near–far interference problem, the received signal on the reverse channel from a subscriber occupying a time slot can be much larger than the received power from the subscriber using the adjacent time slot.
(m) Adaptive equalisation is usually necessary because the transmission data rates are usually very high.
(n) High synchronisation overhead is required because the receivers need to be synchronised for each data burst. In addition, guard time slots are necessary to separate subscribers, and this result in larger overheads.

Facts to Know!

In TDMA, each user has access to the entire allocated RF bandwidth for a short duration of time (time slot) to transmit a preamble and traffic data burst. During the allocated burst time slot, the system transmits the data at much faster rate than the user information data rate. All users share the allocated frequency spectrum with all other users who have time-slot-burst allocations at other pre-assigned time slots.

EXAMPLE 8.6 Advantages of TDMA cellular over FDMA cellular systems

List the advantages of digital TDMA cellular systems over analog FDMA cellular systems.

Solution

The various advantages are listed below:

(a) TDMA systems transmit each signal with sufficient guard time between time slots. This enables to accommodate the transmission time delay because of propagation distance, predetermined delay spread, source time inaccuracies due to clock instability, and the tails of signal pulses due to transient responses.
(b) There is a threefold to sixfold increase in the number of mobile subscribers using a single carrier channel because of interleaving transmissions in the time domain. Digital compression techniques are used to realise timesharing. It produces bit rates which may be approximately one-tenth of the initial raw sample rate and about one-fifth of the initial sample rate after inclusion of error detection/correction bits.
(c) With the use of more advanced digital-modulation schemes and signal-processing techniques, digital signals are much easier to process than analog signals.
(d) Digital signals can be easily encrypted at the transmitting end and decrypted at the receiver end, leading to safeguarding against eavesdropping.

(e) It is possible to monitor the signal strength and bit error rates frame-by-frame which enable either subscribers or base stations to initiate and implement hand-offs.

(f) A flexible bit rate, not only for multiples of basic single channel data rate but also submultiples is allowed for low-bit-rate broadcast-type traffic application.

(g) The TDMA based cellular communication systems are interoperable and compatible with other digital formats such as those used in computer networks.

(h) Digital systems inherently provide a quieter environment and offer better signal quality in a mobile radio environment.

Facts to Know!

If a part of the available frequency spectrum is allocated to a particular group of users, this access method is referred to as narrowband TDMA, e.g, GSM and IS-136. If the complete available spectrum is allocated to each user during the user time slot or duration of the data burst, the system is referred to as wideband TDMA. Each user has to transmit data at a very high data rate, as the time slots have very short durations since many users access the same RF spectrum.

8.3.2 A Basic TDMA Communication Link

It is required to use complicated signal-processing techniques to implement various functional needs of TDMA systems efficiently. Some of these functions include source-coding and channel-coding techniques, sophisticated timing and fast acquisition operations for synchronising, and for the efficient and reliable transmission of data over the wireless channel. The fading is frequency selective which introduces intersymbol interference (ISI) because of wider channel bandwidths along with an increased data transmission rate. To mitigate the ISI problem, channel equalisation has to be provided. Passband modulation techniques are required to be used for the transmission of digitised speech and data over a wireless channel. This necessitates the use of synchronisation for the locally generated carrier frequency, carrier phase, and symbol timing at the receiver. Fig. 8.14 shows the block diagram of a basic TDMA link.

The speech signal input is first sampled to convert analog signal into equivalent digitised speech signal. In order to remove redundant information, the digitised speech signal is encoded without compromising the ability of the receiver to provide a high-quality reproduction of the original signal. The channel encoder introduces controlled redundancy bits into the speech-encoded signal to provide protection against channel noise. A wireless channel produces errors in the form of data bursts, mainly due to deep signal fades. To mitigate this particular channel impairment, an interleaver is used for the purpose of pseudo-randomising the order of the binary symbols in the channel-encoded signal in a deterministic manner.

The function of a packetiser is to convert the encoded and interleaved sequence of digitised speech data into successive packets. Each packet occupies a significant part of a basic TDMA frame. Each frame also includes synchronisation bits in order to synchronise the timing operations in the receiver with the corresponding ones in the transmitter. Knowing the estimate of the channel impulse response, channel equalisation at the receiving end of the TDMA communication link is made possible. The packetised speech data is then modulated onto a sinusoidal carrier for transmission over the channel.

The receiver side consists of a cascade of several functional blocks in order to reverse the corresponding operations performed by the transmitter and the wireless channel. The digital demodulator converts the modulated received RF signal into its baseband form without any loss of information. The baseband processor operates on the resulting complex baseband signal to estimate the unknown channel impulse response, and channel equalisation. The resulting output is then deinterleaved, channel decoded, source decoded, and, low-pass filtered for final delivery of an estimate of the original speech signal to the receiver output.

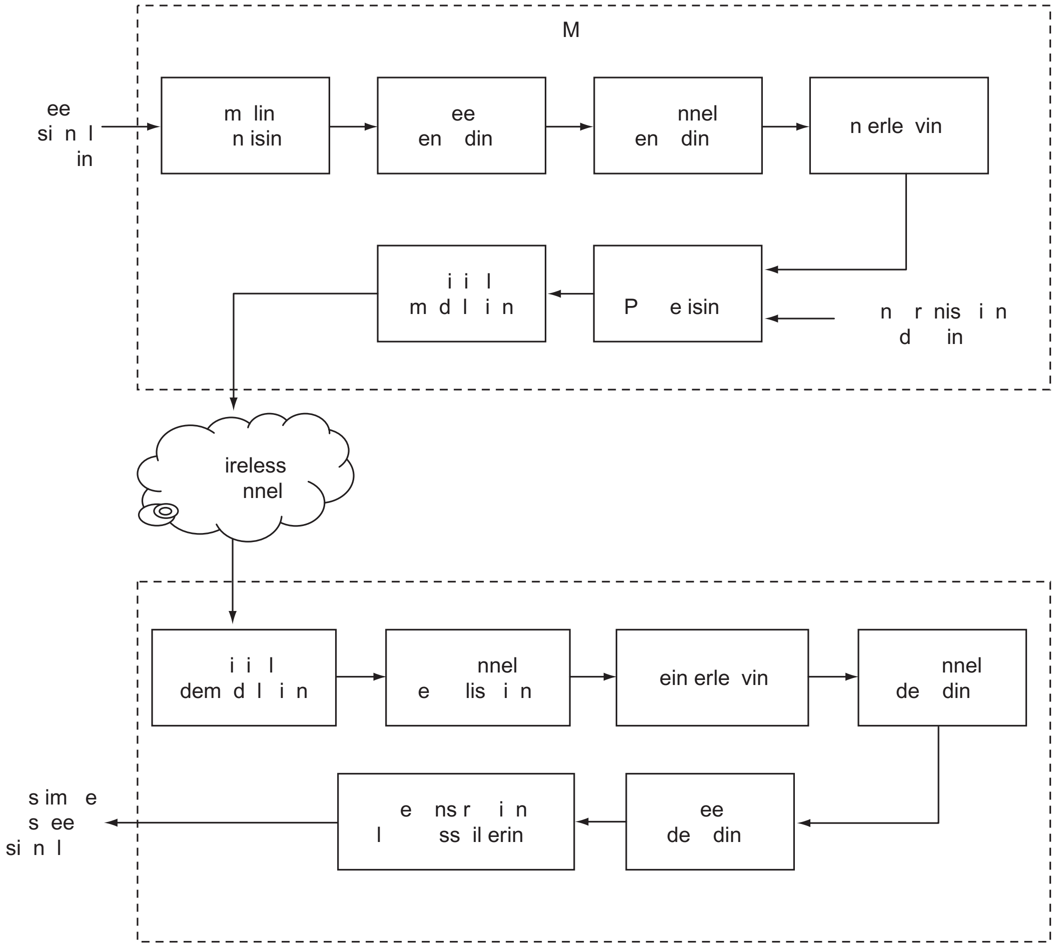

Fig. 8.14 *Block diagram of a basic TDMA link*

8.3.3 Frame Efficiency in a TDMA System

The frame efficiency of a TDMA system is defined as the number of bits representing digitised speech, expressed as a percentage of the total number of bits including the control overhead bits that are transmitted in a frame. For example, in a TDMA cellular system based on IS-136 standards, the forward channel contains 260 traffic data bits out of a total of 322 bits in a TDMA frame (ignoring 2 bits used as reserved bits). The frame efficiency in this case is ($260 / 322 \times 100 =$) 80.7%.

EXAMPLE 8.7 Frame efficiency of TDMA-based GSM system

The basic TDMA frame structure of a GSM cellular system comprises of 156.25 bits in a time slot, of which 40.25 bits are overhead (ignoring the 2 flag bits). Compute the frame efficiency.

Solution

Total bits in a TDMA frame = 156.25 bits (given)

Number of overhead bits = 40.25 bits (given)

The frame efficiency = 1 − (overhead bits / total bits) × 100

The frame efficiency = 1 − (40.25 / 156.25) × 100

Hence, the frame efficiency = 74.2 %

8.4 SPREAD-SPECTRUM MULTIPLE ACCESS

The objective of a cellular system is to reuse the allocated radio spectrum over a large area as many times as possible. Within a cell, either FDMA or TDMA technique can be used for sharing the wireless medium. Distinct sets of channel frequencies are assigned to each cell, with the channel frequencies being reused in sufficiently separated cells. A drawback of using FDMA and TDMA in cellular systems is that the reuse distance is typically limited by worst-case cochannel interference. Spread spectrum multiple access (SSMA) uses signals which have a transmission bandwidth that is several times greater than the minimum required RF bandwidth.

The spread-spectrum technique spreads the information-bearing data signal over a large bandwidth. As a result, it allows the same spectrum to be used simultaneously by many subscribers in adjacent cells. SSMA also provides immunity to multipath interference and robust capability for multiple access. SSMA is not very bandwidth efficient when used by a single subscriber. However, spread spectrum systems become bandwidth efficient in a multiple subscriber environment since many subscribers can share the same spread spectrum bandwidth without interfering with one another.

8.4.1 How Spread Spectrum Technique Works

Spread spectrum is a transmission technique wherein transmitted data occupy a larger bandwidth than required. Spreading of bandwidth is accomplished through the use of a code that is independent of the subscriber data. The same code is used to demodulate the received data at the receiving end. Spread-spectrum systems employ modulation techniques in which the information signal with bandwidth R_b, is spread to occupy a much larger transmission bandwidth R_c.

Facts to Know!

A pseudo-noise (PN) sequence converts a narrowband signal to a wideband noise-like signal before transmission.

To compare a spread-spectrum system with an FDMA system, consider a communication service for which the available bandwidth is $B_t = R_c$. An FDMA would divide this bandwidth into N channels of bandwidth $B_c = B_t / N$, and each subscriber would be allotted a channel of bandwidth B_c. Ordinarily, B_c would closely match the minimum bandwidth required by the subscriber. With spread-spectrum techniques, the spectrum is not divided. Rather, more than one subscriber is permitted to occupy all or any part of the spectrum when transmitting simultaneously.

Spread-spectrum techniques have greater tolerance for noise and interference because of use of wideband, noise-like signals. The spread signals are difficult to demodulate, detect, intercept or jam than narrowband signals. Spread signals are intentionally made to occupy much wider bandwidth than the bandwidth of the information signals they carry, to behave as more noise-like. Spread-spectrum signals use fast code signals having a data rate many times the data rate of the information signal. These spreading codes are referred to as *pseudorandom* or *pseudonoise* (not real Gaussian noise) codes.

Spread-spectrum transmitters use similar transmit power levels as that of any narrowband transmitters. Because of wide spread-spectrum signals, they transmit at a much lower spectral power density than narrowband transmitters. This enables spread signals as well as narrowband signals to occupy the same bandwidth, with little or no interference. In fact, spread spectrum can be considered as a class of digital modulation techniques, characterised by its wide frequency spectra. The modulated output spreaded signals occupy a much larger bandwidth than the signal's baseband information bandwidth. To qualify the modulated signal as a spread spectrum signal, there are two main criteria:

– The bandwidth of the transmitted spread signal is much greater than the bandwidth of the information data signal.
– Orthogonal PN code sequence other than the information data being transmitted, determines the actual transmitted bandwidth on-the-air.

Usually, spread spectrum systems transmit an RF signal having bandwidth as wide as 10 to 100 times the bandwidth of the information data being sent. Some spread-spectrum systems have even employed RF bandwidths 1000 to 1 million times the information bandwidth.

8.4.2 Direct Sequence Multiple Access

The direct sequence multiple access technique is based on the direct sequence spread spectrum (DSSS) modulation scheme. In a DSSS method, the information digital signal is multiplied by a pseudorandom sequence whose bandwidth is much greater than that of the information signal itself. In fact, DSSS is a modulation technique wherein a pseudorandom sequence directly phase modulates a data-modulated carrier. It results into considerable increase in the bandwidth of the transmission and lowering the spectral power density. The resulting spreading signal has a noiselike spectrum to all except the intended SS receiver. The received signal is despread by correlating it with a local pseudorandom signal identical to and in synchronisation with the code signal used to spread the data at the transmitting end.

Fig. 8.15 illustrates the basic concept of DSSS technique for spreading the information data signal and despreading the DSSS signal.

Direct sequence spread spectrum systems are so called because a high-speed PN code sequence is employed along with the slow-speed information data being sent, to modulate their RF carrier. The high-speed PN code sequence is used directly to modulate the carrier, thereby directly determining the transmitted bandwidth. Binary code sequences as short as 11 bits or as long as 2^{89} 1 bits at code rates ranging from less than a bps to several hundred Mbps are employed for various applications.

The result of modulating an RF carrier with such a spread code sequence is to produce a signal centered at the carrier frequency, direct sequence modulated spread spectrum with a $(\sin x) / x^{2}$ frequency spectrum. The main lobe of this spectrum has a bandwidth twice the clock rate of the modulating code, from null to null. The sidelobes have a null-to-null bandwidth equal to the code's clock rate. Direct sequence spectra vary somewhat in spectral shape depending upon the actual carrier and digital-modulation scheme used. The most practical system applications employing direct-sequence spread-spectrum techniques use digital modulation formats such as BPSK and QPSK.

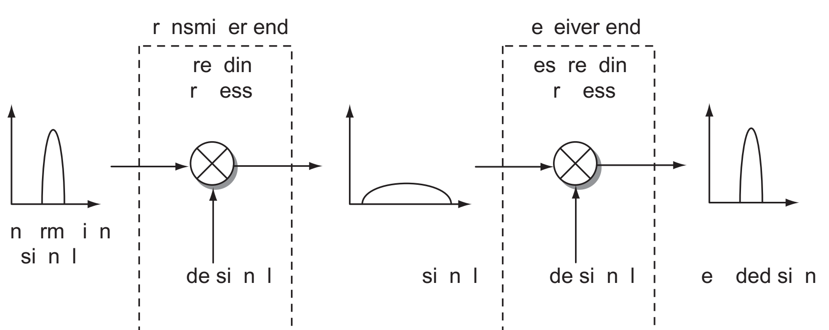

Fig. 8.15 | *Basic Concept of DSSS technique*

Fig. 8.16(a) shows the received spectra of the desired spread spectrum signal and the interference at the output of the receiver wideband filter. Multiplication by the spreading waveform produces the spectra of Fig. 8.16(b) at the demodulator input.

The signal bandwidth is reduced to B_s, while the interference energy is spread over an RF bandwidth exceeding B_{ss}, as depicted in the Figure. The filtering action of the demodulator removes most of the interference spectrum that does not overlap with the signal spectrum. Thus, most of the original interference energy is eliminated by spreading and minimally affects the desired receiver signal. An approximate measure of the interference rejection capability is given by the ratio B_{ss}/B_s, which is equal to the processing gain, termed G_p. The greater the processing gain of the system, the greater will be its ability to suppress in-band interference.

8.4.3 Features of DSSS Technique

Some of the distinct features of direct sequence spread-spectrum are described below.

(a) Increased Tolerance to Interference Direct-sequence spread-spectrum modulators process a narrowband information signal to spread it over a much wider bandwidth. With this approach, each subscriber is assigned a unique spreading code that makes each subscriber's transmissions approximately orthogonal to those of other subscribers. Spreading the information signal de-sensitises the original narrowband signal to some potential interference to the channel.

(b) Low Probability of Interception The transmitted energy remains the same, but the signal spectrum is often below the noise floor of receivers due to the much larger transmitted bandwidth. The signal looks like noise to any receiver that does not know the signal's code sequence.

(c) Increased Tolerance to Multipath Since multipath is viewed as a form of interference, increased tolerance to interference also means increased tolerance to multipath. In fact, multipath energy may be used to the advantage to improve the system performance.

(d) Increased Ranging Capability The increased ranging capability is due to the fact that timing error, say Δt is inversely proportional to the signal bandwidth, R_c and Δt is directly proportional to the range error, Δd. This property permits some spread-spectrum techniques to measure distance or equipment location, through a method known as triangulation.

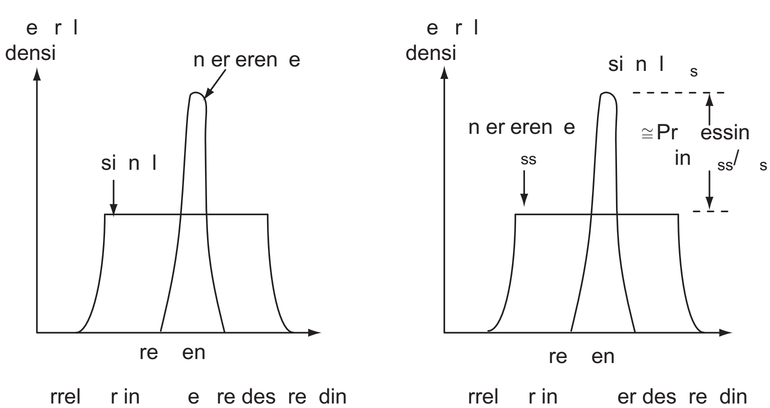

Fig. 8.16 *Spectra of desired received signal with interference*

8.4.4 Frequency Hopped Multiple Access

The frequency hopped multiple access technique is based on the frequency hopping spread spectrum (FHSS) modulation scheme. The wideband frequency spectrum is generated in a different manner in a frequency hopping technique. Frequency hopping involves a periodic change of transmission frequency over a wide band. The rate of hopping from one frequency to another is a function of the information rate, and the specific order in which frequencies are occupied is a function of a code sequence. The transmitted spectrum of a frequency-hopping signal is quite different from that of a direct sequence system. Instead of a $(\sin x)/x^{2}$ -shaped envelope, the frequency hopper's output is flat over the band of frequencies used.

A frequency-hopping signal may be regarded as a sequence of modulated data bursts with time-varying, pseudorandom carrier frequencies. The set of possible carrier frequencies is called the *hopset*. Hopping occurs over a frequency band that includes a number of channels. Each channel is defined as a spectral region with a central frequency in the hopset and a bandwidth large enough to include most of the power in a narrowband modulation burst (usually FSK) having the corresponding carrier frequency.

The bandwidth of a channel used in the hopset is called the instantaneous bandwidth. The bandwidth of the spectrum over which the hopping occurs is called the *total hopping bandwidth*. Data is sent by hopping the transmitter carrier frequencies to seemingly random channels, which are known only to the desired receiver. On each channel, small bursts of data are sent using conventional narrowband modulation before the transmitter hops again. The bandwidth of a frequency-hopping signal is simply '*w*' times the number of frequency slots available, where '*w*' is the bandwidth of each hop channel.

If only a single carrier frequency (single channel) is used on each hop, digital data modulation is called *single-channel modulation*. The time duration between hops is called the *hop duration* or the *hopping period* and is denoted by T_h. The total hopping bandwidth and the instantaneous bandwidth are denoted by B_{ss} and B_s, respectively. The processing gain, $G_p = B_{ss}/B_s$ for FH systems, is same as in case of DS systems.

In an FH method, a pseudorandom frequency hopping sequence is used to change the radio signal frequency across a broad frequency band in a random manner, as shown in Fig. 8.17.

A spread-spectrum modulation technique implies that the radio transmitter frequency hops from channel to channel in a predetermined but pseudorandom sequence. The RF signal is despread at the receiver end using a frequency synthesiser controlled by a pseudorandom sequence generator synchronised to the transmitter's pseudorandom sequence generator. Frequency hopping may be classified as fast frequency hopping or slow frequency hopping. *Fast frequency hopping* occurs if there is more than one frequency hop during each transmitted symbol. It implies that the hopping rate equals or exceeds the information symbol rate. *Slow frequency hopping* occurs if one or more symbols are transmitted in the time interval between frequency hops. A frequency hopper may be fast hopped where there are multiple hops per data bit. It may be slow hopped where there are multiple data bits per hop.

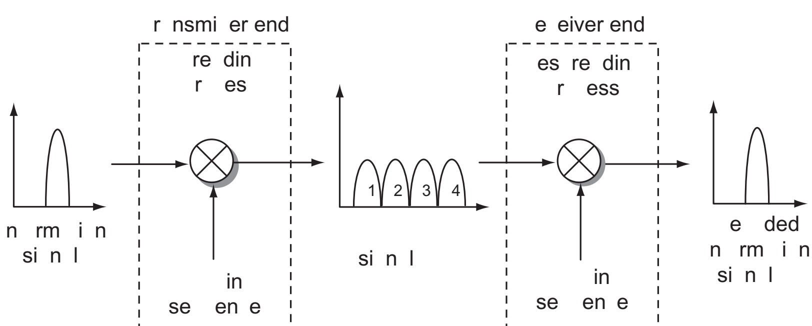

Fig. 8.17 *Basic concept of FHSS technique*

Facts to Know!

In slow-frequency hopping spread spectrum system, the hop rate is less than the baseband information bit rate. Thus two or more (more than 1000 in several implementations) baseband information bits are transmitted at the same frequency before hopping to the next RF frequency.

Multiple simultaneous transmissions from several subscribers is possible using frequency-hopping, as long as each subscriber uses different frequency hopping sequences with the condition that no more than one subscriber unit uses the same frequency subband at any given time. A pseudorandom hopping sequence that is known by the receiver is used by the transmitter. Frequency-hopped modulators at the transmitting end process the narrowband signal and change the carrier frequency every few symbols. To an undesired receiver, the signal appears to be transmitting on randomly selected frequencies, although the hop time on each frequency is usually constant and known.

Multiple frequency-hopped transmitters share the same frequency band by using different frequency hopping sequences. If the transmitters are synchronised, then the hopping sequences can be selected so that there is no possibility that both transmitters may communicate on the same frequency and at the same time. If the transmitters are not synchronised then Forward-Error Correction (FEC) techniques are used to minimise errors.

Facts to Know!

In a fast-frequency-hopping spread spectrum system, the hopping rate is greater than the baseband data rate. Thus, one information bit is transmitted by two or more frequency-hopped RF signals.

The frequency hop rate of a frequency hopped system is determined by the type of information being transmitted, the amount of redundancy used in FEC code, frequency agility of receiver synthesisers, and the distance to the nearest potential interferer. Frequency-hopped systems that transmit multiple symbols during each hop period are referred to as *slow frequency-hopped systems*. The same symbol is generally transmitted on multiple hops in fast frequency-hopping systems. Due to phase continuity limitations in the switching of the frequency synthesiser, frequency-hopped systems typically use a noncoherent form of modulation, such as FSK, rather than BPSK or QPSK. When BPSK is used, the pair of possible instantaneous frequencies changes with each hop. The frequency channel occupied by a transmitted symbol is called the *transmission channel*, and that occupied by the alternative transmitted symbol is called the *complementary channel*.

Most frequency-hopping systems use uniform frequency hopping over a band of frequencies. A digitally controlled frequency synthesiser radio can be converted to a frequency hopping radio with the addition of a pseudonoise (PN) code generator to select the frequencies for transmission or reception. A frequency-hopped system can use an analog or digital carrier modulation scheme and can be designed using conventional narrowband techniques. Frequency de-hopping in the receiver is done by a synchronised PN code generator that drives the local oscillator frequency synthesiser of the receiver.

The frequency-hopped multiple access (FHMA) is a digital multiple access system in which the carrier frequencies of the individual subscribers are varied in a pseudorandom sequence within a wideband channel. FHMA allows multiple subscribers to simultaneously occupy the same spectrum at the same time. Based on the particular PN code of the subscriber, each subscriber dwells at a specific narrowband channel at a particular instance of time. The digital information data of each subscriber is divided into uniform sized data bursts, which are transmitted on different channels within the allocated spectrum band. The instantaneous bandwidth of any one transmission burst is much smaller than the total spreaded bandwidth.

In the frequency-hopped receiver, a locally generated PN code is used to synchronise the receiver's instantaneous frequency with that of the transmitter. The pseudorandom change of the channel frequencies of the subscriber randomises the occupancy of a specific channel at any given time. A frequency-hopped signal only occupies a single, relatively narrow channel at any given point of time since a narrowband

FM or FSK modulation scheme is used. The difference between an FHMA and an FDMA system is that the frequency-hopped signal changes channels at relatively rapid intervals. FHMA systems often employ an energy-efficient constant envelope modulation scheme. This implies that linearity is not a problem, and the power of multiple subscribers at the receiver does not degrade the performance. A fast frequency-hopping system may be thought of as an FDMA system, which employs frequency diversity.

When a large number of channels are used, a frequency-hopped system provides a level of security, since an intercepting receiver that does not know the pseudorandom sequence of frequency hops must retune rapidly to search for the signal it wishes to intercept. Error control coding and interleaving techniques can be used to protect the frequency-hopped signal against deep fades, which may occasionally occur during the frequency hopping sequence.

8.4.5 Spread Spectrum and CDMA

Spread spectrum is a modulation technique that is quite tolerant of interference, and it forms the basis for the access technique known as spread-spectrum multiple access or code-division multiple access (CDMA). CDMA refers to a multiple access technique in which the individual mobile subscribers occupy the complete spectrum whenever they transmit. Many mobile subscribers can occupy the same spectrum at the same time. The integration of different types of traffic such as voice, data, and video can be readily accomplished in a CDMA environment, as subscribers do not require any specific coordination.

In principle, CDMA can accommodate various subscribers with different bandwidth requirements, switching methods and technical characteristics. However, implementations of precise power control techniques are essential in the efficient operation of a CDMA system because each subscriber signal contributes to the interference received by other subscribers.

CDMA is a form of spread spectrum modulation in which subscribers are allowed to use the available spectrum, but their signal must be spread with a specific PN code to distinguish it from other signals. In CDMA, all subscribers transmit information simultaneously by using the same carrier frequency. Each subscriber has its own code word, which is orthogonal to code words of other subscribers. To detect the information, the receiver should know the exact code word used by the transmitter and perform a time correlation operation. All other code words appears as noise due to de-correlation and power should be high to minimise this noise power at the receiver end.

In CDMA technique, one unique code is assigned to each subscriber and distinct codes are used for different subscribers. This code is employed by a subscriber to mix with each information bit before it is transmitted. The

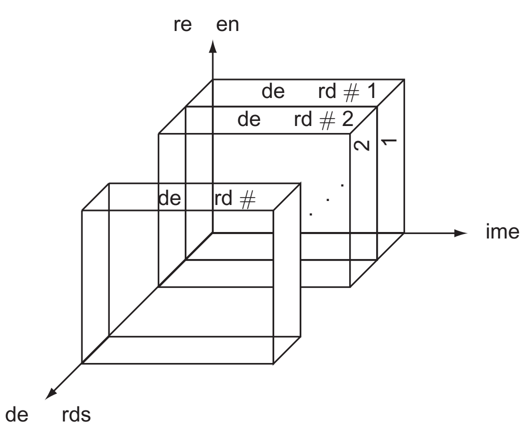

Fig. 8.18 | *The concept of CDMA*

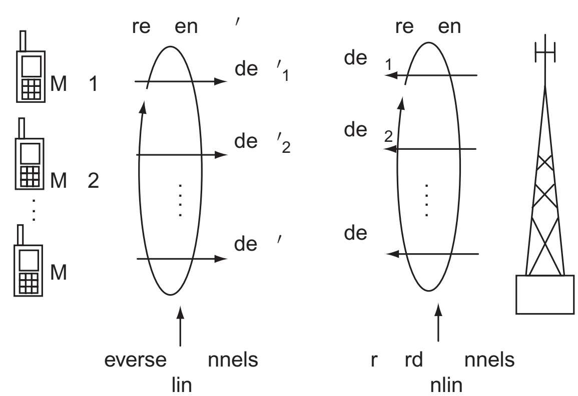

Fig. 8.19 | *Structure of a CDMA system*

same code is used to decode these encoded bits, and any mismatch in code interprets the received information as noise.

The CDMA technique utilises a wider frequency band for each subscriber. In a CDMA system, different spread-spectrum codes are generated by the PN code generator and assigned to each subscriber, and multiple subscribers share the same frequency, as shown in Fig. 8.18.

A basic structure of a CDMA system is shown in Fig. 8.19.

EXAMPLE 8.8 | Illustration of CDMA/FDD and CDMA/TDD concept

Illustrate the concept of CDMA, CDMA/FDD and CDMA/TDD techniques.

Solution

Consider that the available bandwidth and time as resources needed to be shared among multiple mobile subscribers. In a CDMA environment, multiple subscribers use the same frequency band at the same time, and the subscriber is distinguished by a unique code that acts as the key to identify that subscriber. Figure 8.20 depicts a simple CDMA concept.

These unique codes are selected so that when they are used at the same time in the same frequency band, a receiver can detect that subscriber among all the received signals with the help of the known code of that subscriber.

Figure. 8.21 illustrates the basic concept of CDMA/FDD that is used in second-generation IS-95 and third-generation IMT-2000 digital cellular systems in which the forward and reverse channels use different carrier frequencies.

The concept of CDMA/TDD system in its simplest form is shown in Fig. 8.22.

In a CDMA/TDD system, the same carrier frequency is used for uplink and downlink transmissions.

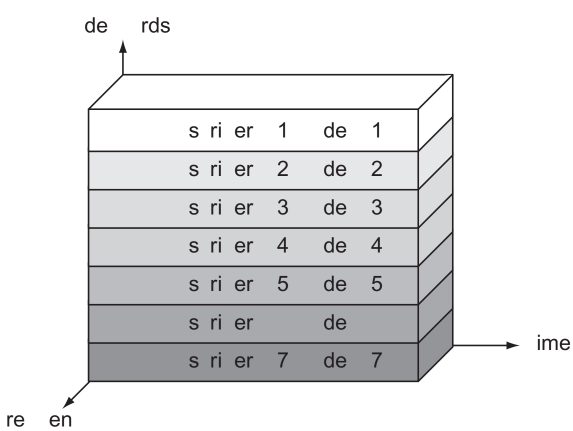

Fig. 8.20 | *Simple illustration of CDMA concept*

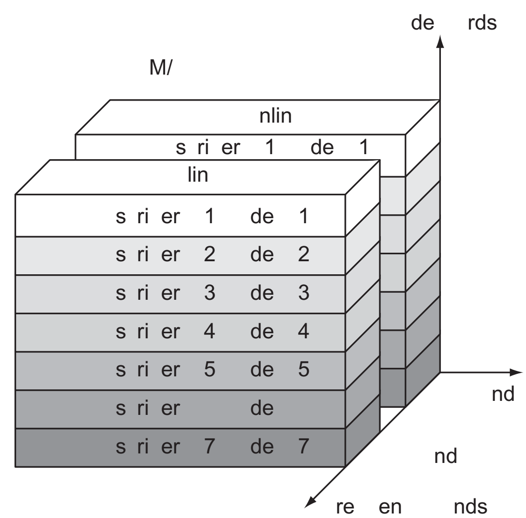

Fig. 8.21 | *CDMA/FDD concept*

In CDMA, each active mobile subscriber is a source of noise to the receiver of other active mobile subscribers. If the number of active mobile subscribers is increased beyond a certain number in the system, the whole CDMA system collapses because the signal received in each specific mobile receiver will be buried

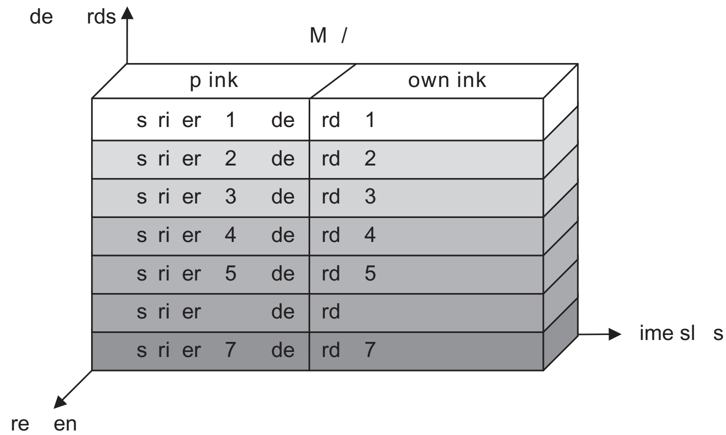

Fig. 8.22 *CDMA/TDD concept*

under the noise caused by many other mobile subscribers. The main concern in a CDMA system is how many active mobile subscribers can simultaneously use it before the system collapses!

A CDMA system is based on spectrum-spread technology by spreading the bandwidth of modulated signal substantially, which makes it less susceptible to the noise and interference. Resistance to fading can be achieved by the use of RAKE receiver concept because of its broadband characteristics. It is quite apparent that using a wider bandwidth for a single communication channel may be regarded as disadvantageous in terms of effective utilisation of available spectrum. The received signals at the cell-site from a faraway mobile subscriber could be masked by signals from a close-by mobile subscriber in the reverse channel due to the near–far problem. However, by using automatic power control that enables to adjust the mobile transmitting power enables the system to overcome the near–far problem, and achieve high efficiency of frequency utilisation in a CDMA system.

A CDMA system is usually quantified by the chip rate of the orthogonal PN codes, which is defined as the number of bits changed per second. The orthogonality of the codes enables simultaneous data transmission from many mobile subscribers using the complete frequency band assigned for a cell-site. Each mobile receiver is provided the corresponding PN code so that it can decode the data it is expected to receive. The encoding in the transmitter and the corresponding decoding at the receiver make the system design robust but quite complex.

Facts to Know!

Theoretically, the number of mobile subscribers being serviced simultaneously is determined by the number of possible orthogonal codes that could be generated.

8.4.6 Salient Features of CDMA Systems

Some second-generation digital cellular systems such as IS-95 and most of the third-generation cellular systems use CDMA technique. Many subscribers share the same frequency in conjunction with FDD or TDD. The number of active subscribers is not limited. This means that a CDMA system has a soft capacity. Increasing the number of active subscribers simply raises the noise floor in a linear manner. Thus, there is no absolute limit on the number of active subscribers. In fact, the system performance gradually degrades for all active subscribers as the number of active subscribers is increased, and improves as the number of active subscribers is decreased.

Multipath fading is substantially reduced because the signal is spread over a large spectrum. The spread spectrum bandwidth is greater than the coherence bandwidth of the channel, which implies that the inherent frequency diversity will mitigate the effects of small-scale fading.

The channel data rates are very high. Consequently, the duration of the symbol or chip is extremely short and usually much less than the channel delay spread. Since PN sequences have low autocorrelation, multipath delayed by more than a chip will appear as noise. A RAKE receiver concept at the receiver can be used to improve reception by collecting time-delayed versions of the required signals.

CDMA uses cochannel cells in adjacent cells, so it can use macroscopic spatial diversity scheme to provide soft hand-off. MSC can simultaneously monitor the signal strength of a particular subscriber from two or more base stations which enables it to perform soft hand-off if needed. The MSC may choose the best version of the received mobile signal at any time without the need of switching frequencies.

The self-jamming problem arises if spreading sequences of different subscribers are not exactly orthogonal. Also, the near–far problem occurs if the received signal power of a desired subscriber at the cell-site is less than that of the undesired subscribers. Each subscriber operates independently with no knowledge of other subscribers.

EXAMPLE 8.9 Multiple access techniques in cellular systems

Tabulate the various multiplexing as well as multiple access techniques used in different cellular communication systems. Comment on the significant of multiplexing techniques.

Solution

Table 8.1 shows the different multiple access techniques being used in various analog and digital cellular communications systems.

Table 8.1 *Multiple access techniques in cellular systems*

S. No.	*Type of cellular system*	*Standard*	*Multiplexing technique*	*Multiple access technique*
1.	1G Analog Cellular	AMPS	FDD	FDMA
2.	US Digital Cellular	USDC	FDD	TDMA
3.	2G Digital Cellular	GSM	FDD	TDMA
4.	Pacific Digital Cellular	PDC	FDD	TDMA
5.	US Narrowband Spread Spectrum Digital Cellular	IS-95	FDD	CDMA
6.	3G Digital Cellular	W-CDMA	FDD/TDD	CDMA
7.	3G Digital Cellular	Cdma2000	FDD/TDD	CDMA

The frequency division duplexing (FDD) technique utilises two distinct frequency bands for every communication link between the mobile subscriber and the cell-site — the forward channel provides traffic from the base station to the mobile subscriber, and the reverse channel provides traffic from the mobile subscriber to the base station. FDD allows simultaneous bidirectional full-duplex radio transmission and reception for both the mobile subscriber and the base station on the duplex channel pair. Regardless of the particular channel pair being used, the duplex spacing between each forward and reverse channel is constant throughout the system.

Time division duplexing (TDD) uses time instead of frequency to provide both a forward and reverse link simultaneously. Multiple mobile subscribers share a single radio channel by taking their respective turns for data transmission in the time domain. Individual mobile subscribers are allowed to access the channel in assigned time slots, and each duplex channel has both a forward time slot and a reverse time slot to facilitate bidirectional communication full duplex. The time separation between the forward and reverse time slot is usually very small, and the transmission and reception of voice/data appears continuous to the subscribers.

8.5 SPACE DIVISION MULTIPLE ACCESS

The three multiple access techniques, namely, FDMA, TDMA, and SSMA have increased spectral efficiency by increasing reuse in frequency, time, and codes. The cell-site antennas are assumed to be omnidirectional (or directional in sectorised cells). If the transmit and receive antenna could be focused directly at the other end of the link, then this would provide a number of improvements such as

- Reduction in the total transmitted power as all power would be transmitted in the desired direction only
- Reduction in the amount of interference generated by each transmitter because total transmit power is reduced and localised
- Receiving a stronger signal by the receiver due to directional antenna gain and less interference

All these features are part of the Space Division Multiple Access (SDMA) technique. Thus, SDMA techniques control the radiated energy for each subscriber in space by using directional or spot beam antennas at the cell-site. The wireless communication space is omni-directional by nature. It can be divided into spatially separable sectors. These different areas in space covered by the respective antenna beam at the cell-site may be served by the different frequencies in an FDMA system or same frequency in a TDMA and SSMA system. This is possible by having a base station to use smart antennas, allowing many subscribers to use the same frequency channel simultaneously. The communication characterised by either carrier frequency, time slot, or spreading codes can be used as shown in Fig. 8.23.

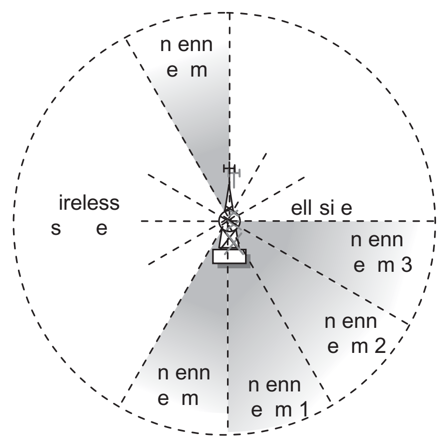

Fig. 8.23 *The concept of SDMA*

The deployment of high-gain directional antenna at the cell-site in a particular direction results in extension of communication range. The use of a smart antenna at the cell-site maximises the antenna gain in the desired direction. It reduces the number of cells required to cover a given geographical area. Moreover, such focused transmission reduces the interference from undesired directions.

A simplified version of transmission using SDMA is illustrated in Fig. 8.24. The cell-site (CS) forms different antenna beams for each spatially separable subscriber on the forward and reverse channels, The noise and interference for each subscriber and the cell-site is minimised. This not only enhances the quality of the communication link significantly but also increases the overall system capacity. Currently, SDMA technology is still being explored and its future looks quite promising.

In cellular systems, a few channels are broadcast by the cell-site on the downlink to transmit system information, and a few channels are shared by all mobile subscribers on the uplink. Almost all the traffic channels are point-to-point communication links between a cell-site and a specific mobile subscriber. This simply can be considered as the active communication link being highly directional in nature when in operation. As an example, a 7-cell frequency reuse pattern cellular system is presented in Fig. 8.25, which uses 3-sector directional antennas in each cell. This means that each directional antenna at the cell-site covers one sector which is 120 of the cell. In the illustration, each cell-site requires three non-overlapping directional antennas, each with a beamwidth of 120 .

Let there be N active mobile subscribers per cell at any time, P_t is the average power radiated per mobile subscriber by the cell-site, and G_t is the transmitting antenna gain. In the downlink,

The power radiated on one of the sector antennas $= (N/3)\ (P_t \times G_t)$ (8.3)

It is implied that $(P_t \times G_t)$ must be the same whether the antenna is omnidirectional or directional. Consequently, the total power radiated with a sector antenna is one-third of that radiated by an omnidirectional

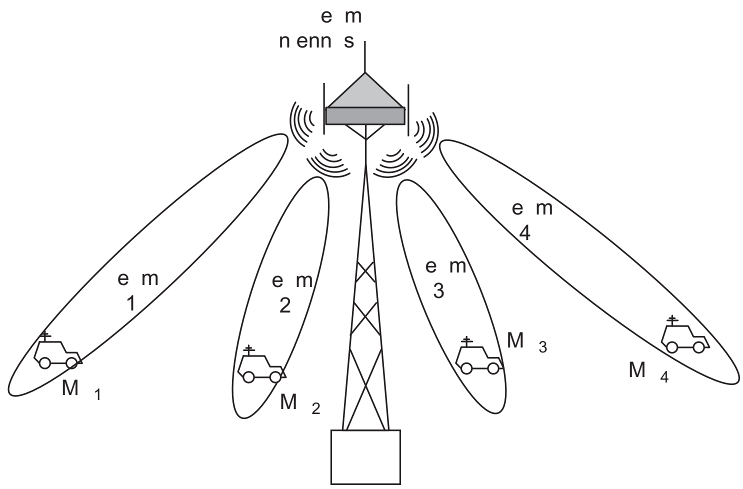

Fig. 8.24 *The basic structure of an SDMA system*

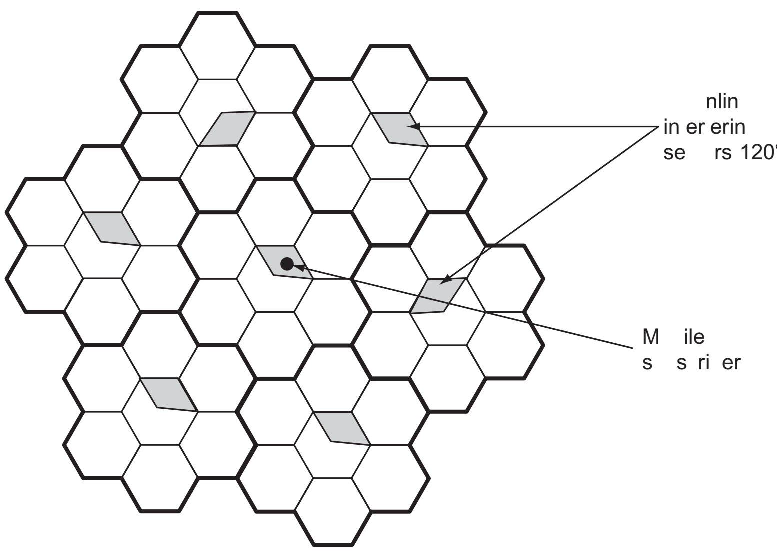

Fig. 8.25 *Cellular system with 120° sector antennas*

antenna. In other words, a mobile subscriber receives only one-third of the interference that would be produced by omnidirectional cell-site antennas with the same number of subscribers.

All mobile subscriber equipments use omnidirectional antennas in the uplink. Assuming uniform distribution of mobile subscribers in the cell at any time, only one-third of them are in any one sector. So the interference is reduced by two-thirds in the uplink direction as well. Hence it can be stated that with 120 sector antennas at the cell-site, the number of subscribers can be increased three times relative to the omnidirectional antenna case while maintaining the same interference levels.

8.5.1 Advantages of SDMA approach

(a) It can be applied with FDMA, TDMA, or CDMA.
(b) It allows many subscribers to operate on the same frequency and/or time slot in the same cell.
(c) It leads to more number of subscribers within the same allocated frequency spectrum with enhanced user capacity.
(d) This technology can be applied at the cell-site without affecting the mobile subscriber.

When subscriber density grows beyond the capacity of a single cell in conventional cell-sites using omnidirectional antennas, the growth is accommodated by dividing the initial larger cell into a number of smaller cells in a process known as cell splitting. Power control is used to reduce the interference among these smaller cells. Although sector antennas are more expensive than omnidirectional antennas, it is still more economical to add sector antennas than adding new cell-sites.

The reverse link (uplink) in a cellular communication system presents the various challenges for several reasons. First, the base station has complete control over the power of all the transmitted signals on the forward link. However, the transmitted power from each mobile unit must be dynamically controlled to prevent any single subscriber from increasing the interference level for all other subscribers because of different radio propagation paths between the base station and each mobile subscriber. Second, transmit power at the mobile unit is limited by battery consumption, which poses limits on the extent to which power may be controlled on the reverse link. The reverse link for each subscriber can be improved with lesser power requirement if the base station antenna is made to spatially filter each desired mobile subscriber so that more energy is detected from each mobile subscriber as in the SDMA system.

Some of the problems on the reverse link can be resolved by using adaptive antennas at the base station and eventually at the subscriber units. Adaptive antennas implement optimal SDMA in the limiting case of infinitesimal beamwidth and infinitely fast tracking ability, thereby providing a unique channel that is free from the interference of all other subscribers in the cell. This enables all the subscribers within the system to communicate using the same channel at the same time. In addition, a perfect adaptive antenna system would be able to track individual multipath components for each subscriber and combine them in an efficient manner to gather all the available signal energy from each subscriber.

Facts to Know!

Patch antennas emit an RF energy beam which is horizontally more wide but vertically taller than that of a directional yagi antenna. Adaptive or phased array antennas are similar to patch antennas but they are divided into a matrix radiating elements, instead of being just a single piece of metal.

8.5.2 SDMA and Smart Antennas

SDMA technique basically takes advantage of the directional nature of wireless communications and relies on the deployment of smart antennas at the cell-site. Some examples of smart antennas are given below:

– The simplest example of a smart antenna is the use of *sector antennas* at the cell-site. The sectored antenna arrangement provides significant capacity gains simply by dividing the service area of each cell-site into three (or more) angular sections.
– *Switched-beam antennas* are the next evolution of smart antennas. These antennas have a number of fixed beams that cover 360°. Switched-beam antennas are typically narrower than sector antennas. The mobile receiver selects the beam that provides the best signal level and interference reduction.
– *Adaptive antennas* are the most advanced example of smart antennas. Adaptive antennas provide a dedicated beam for each subscriber. These antenna dynamically adjusts its sequence to minimise the effects of noise, interference, and multipath.

Facts to Know!

Smart antennas are used mostly in cellular mobile applications and can track a mobile user by sending a narrower, more efficient beam of RF energy directed at the user, which also prevents interference with other transmitter antennas.

The development of such smart antennas could allow even greater reuse of the radio spectrum. The subscribers which are spatially separated by virtue of their angular directions in the cell forms the basis of SDMA. This results into significant improvements in spectral efficiency. Different subscribers are able to reuse the same radio spectrum as long as they are separated in angle. In particular, multibeam antennas are used to separate radio signals by pointing them along different directions of the subscribers.

There are many advantages to smart antennas for cellular mobile applications:

(a) Greater Range Since the antennas are directional, they have larger gains and can therefore provide stronger received signal strength for the same transmit power.

(b) Fewer Cell Sites Fewer cell-sites are required in those geographical areas with a low subscriber density because the existing cell-site has a greater range. In areas with a high subscriber density, there is less interference. Moreover smart antennas provide greater subscriber isolation. Hence, a single cell-site can serve more number of subscribers.

(c) Better Signal Penetration Due to the greater signal strength and increased transmitter gain, signal penetration through building structure is better.

(d) Less Sensitivity to Power Control Errors Due to better isolation among different subscriber signals, probability of power control errors reduces considerably.

Facts to Know!

SDMA improves system capacity by virtue of efficient spectrum reuse, minimisation of the effects of interference, and increasing signal strength for both the subscriber unit and the cell-site.

(e) Responsive to Hot Spots Traffic Conditions In strategic application areas such as airports, hotels and conference centres, subscriber densities can become quite high at times, and directional antennas allow one or a small number of cell-sites to service these areas effectively.

8.6 HYBRID MULTIPLE ACCESS TECHNIQUES

Practical cellular communication systems deploy usually a combination of two or more of the basic multiple-access techniques: FDMA, TDMA, CDMA, and SDMA. The main objective of hybrid multiple access approach is to provide a reasonable subscriber growth strategy, thereby reducing the network complexity as the hybrid technique remains backward compatible with the existing system. Although one approach may have a significant technical advantage over another, there may be other factors such as economic considerations that prevent the use of the basic multiple access technique in isolation.

Various hybrid multiple access techniques which are in use in different wireless systems are the following:

8.6.1 Hybrid TDMA/FDMA

In practical wireless communication systems, TDMA is generally implemented in an overlaid fashion on FDMA technique. Every wireless communication system has an FDMA technique as baseline, and multiple-access schemes such as TDMA are overlaid on this baseline. The North American IS-54 digital cellular system is an example of such a system, which is also called Narrowband TDMA system. The number of frequency channels is typically large but the number of subscribers time-sharing a single channel is relatively very small. The bandwidth of a channel in narrowband TDMA is relatively small, of the order of 30 kHz or less. GSM digital cellular system combines TDMA with FDMA as well as frequency hopping (optional). Accordingly, a physical channel is partitioned in both frequency and time. The carrier channel is divided in 8 time slots and each carrier channel supports eight simultaneous physical channels mapped onto the eight time slots. A time slot assigned to a particular physical channel is naturally used in every TDMA frame till the channel is engaged by a subscriber. Combined with a frequency-hopping algorithm, partitioning of the channel in frequency arises because the carrier assigned to such a time slot changes its frequency from one frame to the next.

In hybrid TDMA/FDMA technique, the allocated RF spectrum is divided into a number of frequency channels, each of defined channel bandwidth, thereby enabling FDMA operation, followed by dividing each carrier channel into a number of defined time slots in time domain, leading to TDMA/FDMA operation. Figure 8.26 shows a generalised view of FDMA/TDMA technique used in 2G digital cellular communication systems. Forward and reverse channels are separated in the frequency domain to enable FDD operation.

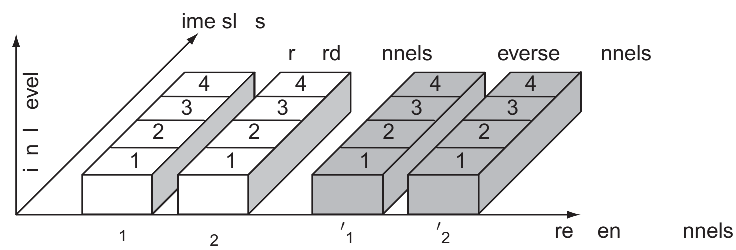

Fig. 8.26 | *Hybrid TDMA/FDMA*

EXAMPLE 8.10 | **Hybrid TDMA/FDMA in IS-136 and GSM**

Illustrate the concept of hybrid TDMA/FDMA technique commonly used in 2G digital cellular systems such as IS-136 and GSM cellular systems.

Solution

Hybrid TDMA/FDMA in IS-136 cellular system

Figure 8.27 shows the format and time-slot structure for the hybrid TDMA/FDMA technique with six time slots per carrier channel used in IS-136 US digital cellular system, both for the forward (base station to mobile subscriber) and reverse (mobile subscriber to base station) channels. Forward and reverse channels use separate carrier frequencies (FDD), the duplex separation being 45 MHz.

In the IS-136 cellular standard, each 30 kHz carrier channel has a gross transmission data rate of 48.6 kbps. The 48.6 kbps data stream is divided into six TDMA channels (time slots) of 8.1 kbps each. Each time slot

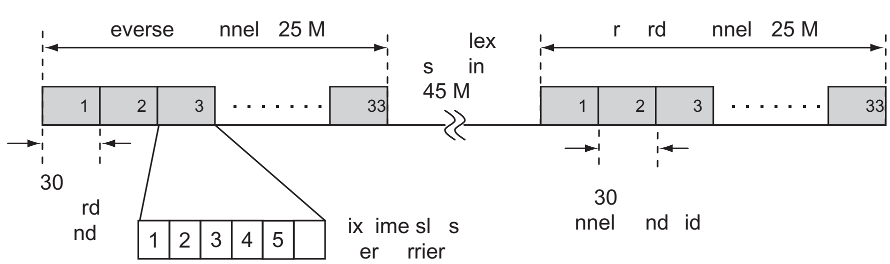

Fig. 8.27 | *Hybrid TDMA/FDMA in IS-136 standard*

contains 324 bits, comprising of 260 bits of subscriber data, and other bits of system control information. The duration of a TDMA frame is 40 ms, and that of a time slot is 6.67 ms time slots.

Hybrid TDMA/FDMA in GSM cellular system

Figure 8.28 shows a particular example of the 8-time-slots TDMA scheme used in the GSM digital cellular standard. Forward and reverse channels use separate carrier frequencies (FDD).

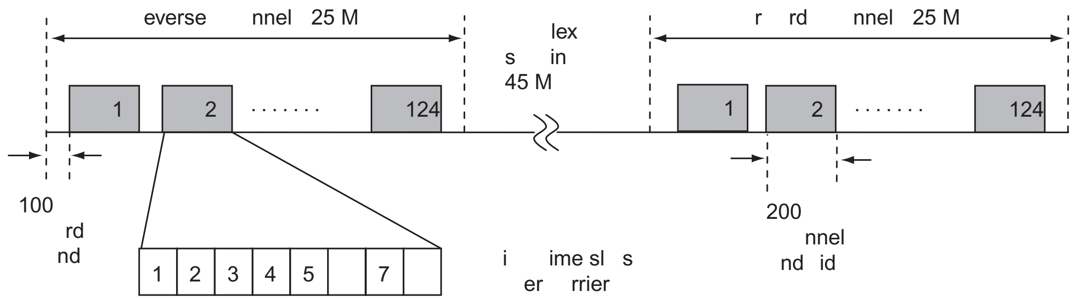

Fig. 8.28 | *FDMA/TDMA/FDD in GSM*

A total of 124 frequency carriers (FDMA) are available in the 25 MHz allocated band in forward and reverse channels (FDD). A guard band of 100 kHz is allocated at each edge of the overall allocated band. Each carrier channel of 200-kHz bandwidth can support up to eight simultaneous transmissions (TDMA), each using a 13-kbps encoded digital speech.

8.6.2 Hybrid TDMA/DSMA

In a hybrid time division direct sequence multiple access (TDMA/DSMA) technique, each cell is using a different spreading code (DSMA employed between cells) that is conveyed to the mobile subscribers operating in its coverage area. Inside each cell (inside a DSMA channel), TDMA is employed to multiplex multiple mobile subscribers. A particular time slot in a TDMA frame is allocated to one mobile subscriber per cell. This implies that only one mobile subscriber transmits in each cell at any time. This results in significant reduction of near–far effect. During the hand-off process from one cell to another cell or from one sector to another sector of the same cell, it is the spreading code of the mobile subscriber which changes while retaining the same allocated time slot of TDMA frame for maintaining the communication link.

> **Facts to Know!**
>
> TDMA techniques combined with slow FH-SS and DS-SS based multiple access techniques are proving to be more promising for future generations of mobile communications systems.

8.6.3 Hybrid TDMA/FHMA

In TDMA-based wireless communication systems, if the cochannel interference is excessive or the occupied channel coincides with a deep frequency selective fading, the received voice signal is distorted. One of the practical methods to minimise the excessive cochannel interference or reduce the duration of the frequency selective fades is to provide for a slow frequency-hopping sequence that imposes a restriction on the cochannel interference effects or duration of the frequency selective fading. This is termed the TDMA/FHMA technique. This is widely employed in severe cochannel interference and multi-path environments.

In the hybrid TDMA/FHMA technique, the mobile subscriber can hop to a new frequency at the beginning of every TDMA frame. At each time slot, the mobile subscriber is hopped to a new frequency according to a pseudorandom hopping sequence. Each successive TDMA frame in a given channel is carried on a different carrier frequency. Usually, the hopping sequence is predefined and the mobile subscriber is allowed to hop only on certain assigned frequencies to a cell. The hybrid TDMA/FHMA technique is employed optionally in the GSM cellular system that supports a frequency-hopping pattern of 217.6 hops per second. This results into an increase in the system capacity by several times in addition to improvement in the signal quality performance. In the case of slowly moving mobile subscribers such as pedestrians, the frequency-hopping algorithm built into the design of TDMA-based GSM system produces substantial gains against fades. The hybrid TDMA/FHMA technique is also used in piconets over a 79 MHz wideband radio channel at a hop rate of 1600 hops per second in Bluetooth technology.

8.6.4 Hybrid DSMA/FHMA

There are two basic types of spread-spectrum implementation methodologies: direct sequence (DS) and frequency hopping (FH). A hybrid direct sequence/frequency hopped multiple access (DS/FHMA) technique combines the advantages of both techniques. With direct-sequence spreading, the original signal is multiplied by a known code signal sequence of much larger bandwidth. The direct sequence technique is considered the most feasible generic method in wireless communication systems when the code is selected and assigned dynamically to each mobile subscriber. With frequency-hopped spreading, the centre frequency of the transmitted signal is varied in a pseudorandom sequence. Practically, it is difficult to use the frequency hopping unless a super-fast frequency synthesiser is used.

In the hybrid DSMA/FHMA technique, the signals are spread using spreading codes (direct sequence signals are obtained), but these signals are not transmitted over a constant carrier frequency; they are transmitted over a frequency-hopping carrier frequency. The centre frequency of a direct sequence modulated signal is made to hop periodically in a pseudorandom manner. In this technique, there is always possibility of transmitting the same data burst in more than one frequency-hopped channels, thereby improving the signal quality performance in a hostile mobile environment. The near–far effect can be avoided but it is difficult to achieve soft hand-off because the FH base station receivers are required to be synchronised to the multiple hopped signals.

Facts to Know!

The performance of a hybrid DSMA/FHMA system is usually better than that can be obtained with an individual spread-spectrum technique.

8.6.5 Hybrid FDMA/DSMA

In the hybrid FDMA/DSMA technique, the available wideband frequency spectrum is divided into a number of narrowband radio channels. Each one of these narrowband channel DSMA system has processing gain which is much lower than the original wideband DSMA system. Depending on the requirements of various mobile subscribers, different narrowband channels can be assigned to each one of these. The overall system capacity of the hybrid FDMA/DSMA technique remains the same as that of the original DSMA system.

8.6.6 Hybrid SDMA with FDMA/TDMA/CDMA

SDMA is generally used in conjunction with other multiple access schemes as there can be more than one subscriber in one antenna beam in any one direction. When SDMA is used with FDMA as well as TDMA (SDMA/FDMA/TDMA), the higher carrier-to-interference value can be exploited for better frequency channel reuse. When SDMA is used with TDMA as well as DSMA (SDMA/TDMA/DSMA), different service areas can be covered by the individual antenna beam, thereby providing a similar effect as obtained by frequency reuse. However, this requires more network resources for proper management because there will be more intra-cell hand-offs needed in SDMA approach as compared to TDMA or DSMA systems alone.

8.7 COMPARISON OF MULTIPLE-ACCESS TECHNIQUES

To summarise, the four multiple-access techniques discussed above justify a common objective of achieving increased efficiency in sharing the radio spectrum: FDMA, in which subscribers share the spectrum by dividing it into different frequency channels; TDMA, in which subscribers time-share the spectrum; CDMA, in which all subscribers use the same spectrum simultaneously, but limiting the number of subscribers due to multiple access interference; SDMA, in which subscribers share the spectrum in angular direction with the use of smart antennas.

There are some parameters of these four multiple access techniques which can be used for comparison purpose. Some of the most important aspects of comparison are given below:

(a) Concept FDMA divides the allocated frequency band into disjoint subbands. TDMA divides the time into non-overlapping time slots. CDMA-spreads the signal with orthogonal codes. SDMA divides the wireless space into angular sectors.

(b) Modulation FDMA and TDMA rely heavily on the choice of a modulation scheme to maximise the spectral efficiency. To achieve a higher throughput in the same bandwidth, higher order modulation schemes must be used. With CDMA, the BPSK modulation is usually required, although QPSK is often used for practical symmetry considerations. The choice of modulation scheme and the use of SDMA are not related.

(c) Source Coding The use of source coding improves the bandwidth efficiency of all multiple access techniques. However, CDMA takes better advantage of voice activation than other multiple access techniques, since its bandwidth efficiency is determined by average interference.

(d) Forward Error-Correction (FEC) Coding All multiple access techniques are affected by the vagaries of the wireless channel. With FDMA and TDMA, the redundancy introduced by FEC coding requires a higher transmission rate, and thus a greater bandwidth to maintain the same basic throughput. There is a tradeoff between bandwidth and power efficiency. With CDMA, FEC coding is used without increasing the bandwidth or affecting the processing gain. The inclusion of FEC is transparent to SDMA. If transmit diversity is implemented then there can be increased bandwidth requirement with SDMA.

(e) Active Mobile Subscribers In FDMA, all mobile subscribers remain active on their assigned frequency channels. In TDMA, various mobile subscribers are active in their specified time slot on the same frequency. In CDMA, all mobile subscribers are active on the same frequency. In SDMA, the number of mobile subscribers per antenna beam depends on FDMA/TDMA/CDMA technique used in conjunction with it.

(f) Signal Separation For signal separation among various mobile subscribers, frequency filtering in FDMA, time synchronisation in TDMA, code separation in CDMA, and spatial angular separation using smart antennas in SDMA is needed.

(g) Diversity To obtain diversity with FDMA, multiple receivers are required. The same is applicable with TDMA except when it is used as part of a TDMA/FDMA hybrid technique. In that case, frequency-hopped TDMA can provide some diversity advantage. The large bandwidth of CDMA naturally provides frequency diversity with the use of a RAKE receiver. With the deployment of smart antennas with SDMA, there will be a reduction in space diversity due to antenna directivity, but there will also be a corresponding reduction in fading effects.

(h) Hand-off or Handover The mobile subscribers in both FDMA and TDMA systems have single-receivers, the hand-off algorithms are limited. The 1G FDMA cellular systems often used the hard-decision hand-off in which the cell-site controller monitors the received signal from the cell-site and at the appropriate time switches the connection from one cell-site to another. TDMA systems use the mobile-assisted hand-off in which the mobile subscriber monitors the received signal from available cell sites and reports it to the cell-site controller which then makes a decision on the hand-off. Because adjacent cells in both FDMA and TDMA use different frequencies, the mobile subscriber has to disconnect from and reconnect to the network that will appear as a click to the mobile subscriber. Hand-offs occur at the edge of the cells when the received signals from both neighbouring cell-sites are weak as well as fluctuating. As a result, decision making for the hand-off time is often complex, and the subscriber experiences a period of poor signal quality and possibly several clicks during the completion of the hand-off process.

Since the same frequencies are used in each cell with CDMA, there is no need to change channels in order to change cell-sites. That is, multiple cell-sites receive the same signal in a form of space diversity. It is easier to implement a dual receiver and provide a soft handover capability. Because adjacent cells in a CDMA network use the same frequency, a mobile moving from one cell to another can make seamless handover by using signal combining technique. When the mobile subscriber approaches the boundary between cells, it communicates with both cells. The controller combines both the signals to form a better communication link. When a reliable link has been established with the new cell-site, the mobile switches over to a new cell-site smoothly. Soft handover provides a dual diversity for the received signal which improves the receive quality and eliminates clicking as well as the ping-pong problem.

In SDMA, there are additional handover requirements associated with subscribers moving between switched-antenna beams. If the antenna beams are fixed then a handover will be necessary whenever a mobile subscriber crosses a beam boundary. If the antenna beams can track mobile subscribers then handover may become necessary when two subscribers occupying the same channel move near to one another.

(i) Power Control Power control is necessary for FDMA and TDMA systems to control adjacent channel interference and mitigate the undesired interference caused by the near–far problem. In CDMA, the system capacity depends directly on the power control, and an accurate power control mechanism is needed for proper operation of the network. Better power control also saves on the transmission power of the mobile subscriber, which increases the life of the battery.

(j) Multiple-Access Interference Since FDMA and TDMA tend to be limited by worst-case cochannel interference, interference is often minimised in the system planning stage by the fixed assignment of frequency groups to specific cells. With CDMA, the same bandwidth is used everywhere, and performance is limited by average interference levels. However, CDMA relies heavily on accurate power control to eliminate the near–far problem. With SDMA, multiple-access interference depends upon the beamwidth and antenna sidelobes.

(k) Fading-Channel Sensitivity FDMA systems are typically narrowband and therefore suffer from flat fading. Multipath in wireless channels causes frequency selective fading. As the transmission bandwidth is increased, fading will occupy only a portion of the transmission band, providing an opportunity for a wide-band receiver to take advantage of the portion of the transmission band not under fade and a more reliable communication link. If the fading is not severe then simple channel estimation and forward error-correction techniques can often compensate for its effects. TDMA systems have typically medium channel bandwidth (200 kHz in GSM); consequently they observe some frequency selectivity. This requires the implementation of a robust tracking equalizer in wireless channels. Due to their large bandwidth (1.25 MHz in IS-95; 5 MHZ or 10 MHz in W-CDMA), CDMA systems face frequency-selective channels, but take advantage of in-band or time diversity of the wideband signal with a RAKE receiver, the robustness of the speech-coding algorithm, methods to tackle interference, handovers, and power control as well.

(l) Bandwidth Efficiency For single-cell systems, FDMA and TDMA systems are generally more bandwidth efficient than CDMA systems, because they do not have to cope with multiple-access interference. FDMA and TDMA have much lower frequency reuse rates because they are limited by peak interference levels. However, the maximum throughput is fixed. For multicell systems, CDMA can often add a subscriber at the expense of a small degradation in the signal quality of existing subscribers.

(m) Synchronisation Cellular systems using FDMA, TDMA, and CDMA show a progress in synchronisation resolution and complexity. The main concern of FDMA is symbol timing; TDMA subscribers must contend with both symbol timing and slot timing, and CDMA subscribers must contend with chip timing. With the increase in bandwidth, subscribers become less sensitive to frequency errors. SDMA has the additional synchronisation-like requirement related to determining the location of the mobile subscriber.

(n) Voice and Data Integration With TDMA, it is possible to make use of periods of voice inactivity to transmit data as well, thus making the system more efficient. CDMA can easily integrate voice and data, but usually it leads to multicode transmissions, which may reduce the efficiency of the subscriber's power amplifier.

(o) Format Flexibility One of the main reasons for migrating from analog FDMA to digital TDMA is that TDMA provides a more flexible environment for integration of voice and data. The time slots of a TDMA network designed for voice transmission can be used individually or in a group format to transmit data from subscribers and to support different data rates. The main advantage of CDMA is its flexibility in timing and the quality of transmission. CDMA subscribers are separated by their codes, unaffected by the transmission time relative to other subscribers. The transmit power of the subscriber equipments can also be adjusted with respect to others to support certain transmission quality. Each CDMA subscriber is independent from other subscribers, accommodating different service requirements to support a variety of transmission data rates with different transmission qualities to support multimedia or any other emerging application.

(p) System Flexibility Once the service is designed, FDMA is the least flexible of the multiple-access techniques; any change requires a redesign. With TDMA, higher data rates can be provided by assigning more slots per subscriber, usually with very little change to the hardware. With CDMA, different data rates can be provided by trading off the processing gain, making it very flexible. All of these techniques are transparent to SDMA.

(q) System Expandability Evolving from a small system to a large system is easiest with the FDMA approach. With TDMA, start-up efficiency is related to the transmission data rate; the system can evolve easily through the addition of more TDMA channels using an FDMA overlay. With CDMA, there is a large initial cost, because a large bandwidth to serve perhaps only a few initial subscriber equipments is needed. With SDMA, the evolution can be quite smooth, since that approach is primarily a cell-site technology. If smart antennas are to be deployed on mobile terminals, they may have to be retrofitted as the antenna beams become denser.

(r) User Equipment Complexity With the progression from FDMA through TDMA to CDMA comes an evolution of mobile subscriber equipment complexity. SDMA systems introduce a different and additional form of complexity, that is, one related to the antenna present in any of the other systems.

(s) System Complexity With an FDMA system, mobile subscribers can operate somewhat independently. With TDMA, the level of cooperation among mobile subscribers must increase to share time slots. With CDMA, the system must delegate spreading codes, power control information and synchronisation information. These complexity aspects are transparent to SDMA for which antenna direction and location of the mobile subscriber equipment are the main concerns.

(t) Implementation Complexity Spread spectrum is a two-layer modulation technique requiring greater circuit complexity than conventional analog or digital modulation schemes. Therefore, CDMA mobile equipments are more complex than FDMA or TDMA mobile equipments.

(u) Advantages FDMA is simple and robust, and has been widely used in analog cellular systems. TDMA and CDMA are flexible, and used in 2G/2.5G/3G digital cellular systems. SDMA is very simple and increases system capacity significantly.

(v) Disadvantages FDMA is inflexible because available frequencies are fixed, and requires guard bands. TDMA requires guard time, and may have a synchronisation problem. CDMA has complex receivers, and requires accurate power control to avoid near–far problem. SDMA is inflexible, and requires network monitoring to avoid intracell hand-offs.

8.8 PACKET RADIO MULTIPLE ACCESS TECHNIQUES

There are many instances in multiple access mobile radio communication systems in which a mobile subscriber is required to send a packet of information to the base station at a random instant of time, for example, when the mobile subscriber has to get registered with the system or wishes to make a phone call. These type of random requests by mobile subscribers can be serviced in a number of ways such as

– By permanently assigning a dedicated channel to each subscriber; but this approach results into wasteful of spectrum
– By polling each subscriber at regular interval of time to check if anyone has any data to transmit; in case of mobile subscribers, the polling process becomes quite complicated; moreover, this approach might result in long delays if there is a large number of subscribers
– By providing access to the common channel randomly at any time; this is basically a packet radio multiple access technique to assign a random-access channel

Packet Radio Multiple Access (PRMA) techniques are basically contention-based protocols in which many subscribers make an attempt to access the common channel either in a random access (without any coordination among them) or collision resolution (with limited coordination) manner. The contention-based protocols used in mobile data networks can be classified into two groups according to the ways collisions are resolved: random access protocols and collision resolution protocols. The first group based on *random access protocols* consists of ALOHA-based access methods for which the subscribers transmit their data packet without any coordination between them. The second group based on *collision resolution protocols* is the carrier-sense based random access techniques for which the subscriber senses the availability of the common channel before it transmits its data packets.

A random access packet radio protocol does not require any call set-up procedure. It is very easy to implement and provides a flexible way to access and manage the channel for bursts of data packets at any time. In this technique, the subscriber transmits its data packets at any time it has information ready to be sent on the common channel. The sender senses the acknowledgement of data packets sent to ensure successful transmission. Since other subscribers are also free to transmit their data packets on the same channel as and when they are ready to send them, there is every possibility for collisions of data packets sent by different subscribers at the same time on the same channel. In case of collision, the sender subscriber retransmits the collided data packet after a random interval of time which is at least greater than the one round-trip transmission delay time of the network.

> **Facts to Know!**
>
> The packet radio multiple access (PRMA) techniques are suitable for bursty type traffic in the form of packets.

The PRMA contention-based techniques can serve a large number of subscribers with least overhead or control mechanism. The performance of PRMA techniques is generally evaluated on the basis of the average delay (D) experienced by a typical data packet and the throughput (T). Throughput is defined as the average number of messages successfully transmitted per unit time.

8.8.1 Pure ALOHA Packet Radio Protocol

The original ALOHA protocol is also called pure ALOHA in order to distinguish it from subsequent versions of the original protocol. The pure ALOHA protocol is a random access packet radio protocol used for packet data transmission. In this technique, it is assumed that there are a large number of subscribers that operate independently of each other and that each subscriber has no knowledge of when the other subscribers will transmit their data packets.

The basic concept of pure ALOHA protocol is very simple. A subscriber transmits an information packet on the physical layer when the packet arrives from the upper layers of the protocol stack. Each packet is encoded with an error-detection code. The base station checks the parity of the received packet. If the parity checks properly, the base station sends a short acknowledgment packet to the sender. Since the data packets by subscribers are transmitted at random times, there will be collisions between packets sent by different subscribers whenever packet transmissions overlap by any amount of time, as indicated in Fig. 8.29. Thus, after transmitting a data packet, the subscriber waits for a time more than the round-trip delay for an acknowledgment from the receiver. If no acknowledgment is received, the data packet is assumed having lost in a collision, and it is transmitted again with a randomly selected delay to avoid repeated collisions. As the number of subscribers increase, a greater delay occurs because the probability of collision increases.

Throughput of Pure ALOHA To evaluate the performance of pure ALOHA protocol technique in terms of throughput, it is assumed that all data packets transmitted by all subscribers in the system have a constant and fixed packet length, L. Let the average transmission rate or mean arrival rate be λ (packets/s). This means that there are, on the average, λ data packets transmitted per second by all the subscribers. This type of situation is commonly modeled as a Poisson distribution process. The probability that there are k arrivals in the time period $0, t$ is given by

$$\text{Probability for } k \text{ arrivals in the time period } 0, t\ , P_s = (\lambda t)^k / k! \quad e^{-\lambda t} \tag{8.4}$$

The Poisson distribution process has a memoryless property. It implies that the probability that there are k arrivals in the period $0, t$ is the same as the probability that there are k arrivals in the period $s, t + s$ for some arbitrary value of s.

If two packets from two subscribers collide (overlap in time), it is assumed that the information in both the packets is lost. It is desired to determine the throughput S of this common channel (throughput is defined as the average number of packets that can be transmitted per time slot). If there is only one subscriber transmitting data at any time then the maximum throughput would be unity. However, in the case of a large number of subscribers, the probability that two or more packets will collide must be considered. It is obvious that

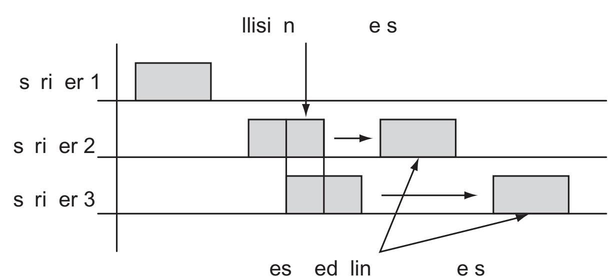

Fig. 8.29 Concept of pure ALOHA protocol

for a packet transmitted at time t_0, any other packet transmitted in the interval $t_0 - T, t_0 + T$), where T is the packet duration, will cause a collision.

With a Poisson distribution process for packet arrival times, if a packet is being transmitted, then the probability that no additional packets arrive during the period $t_0 - T, t_0 + T$, or $t = 2T$ is given by substituting $k = 0$ and $t = 2T$ in Eq. (8.4).

Probability that a packet is successfully received in time $2T$, $P_s = (\lambda t)^0 / 0!\ \ e^{-\lambda (2T}$

Or, Probability that a packet is successfully received in time $2T$, $P_s = e^{-2\lambda T}$ (8.5)

Consequently, the throughput S of an ALOHA system is given by the product of the packet arrival rate λ and the probability that a packet is successfully received in time $2T$, P_s; that is,

$$\text{Throughput of an AL HA system } S(\text{AL HA}) = \lambda\, e^{-2\lambda} \tag{8.6}$$

Normalising this equation to packets received per packet time T, the normalized throughput, S_0 (ALOHA) is given as

Normalized throughput of an ALOHA system, $S_0\ (\text{ALOHA}) = \lambda\, T e^{-2\lambda T}$ (8.7)

Or, $S_0\ (\text{ALOHA}) = G\, \mathrm{e}^{-2G}$ (8.8)

where $G = \lambda$ T is the traffic occupancy or the normalised channel traffic (measured in Erlangs). The parameter G is defined as the normalised loading per packet period, that is, the average number of packets per slot time T, due to arriving and buffered packets. It is a relative measure of the channel utilisation. If $G > 1$ then the packets generated by the subscribers exceed the maximum transmission rate of the channel. It implies that in order to obtain a reasonable acceptable throughput, the rate at which new packets are generated must lie within $0 < G < 1$.

Under normal loading conditions, the throughput T is the same as the total offered load, L. The total offered load L is the sum of the newly generated packets and the retransmitted packets that suffered collisions in previous transmissions. The normalised throughput is always less than or equal to unity and may be considered as the fraction of time (or fraction of an Erlang) a channel is utilised. The normalised throughput is given as the total offered load times the probability of successful transmission.

The peak normalised throughput S_{0peak} occurs at $G = 0.5$. Therefore,

$$S_{0peak}\ (\text{ALOHA}) = 0.5\ e^{-2(0.5)} = 0.184 \text{ packets per packet time}$$

This means that with an ALOHA random-access channel, the maximum throughput is approximately 18.4% or less than 19% of the full channel capacity. In practice, the throughput is maintained at a much smaller value so as to ensure stability of the random-access approach.

EXAMPLE 8.11 | Throughput of pure ALOHA

(a) Compute the maximum throughput of a pure ALOHA network with a large number of subscribers and a transmission rate of 1 Mbps.
(b) Determine the throughput of a TDMA network with the same transmission rate.
(c) What is the throughput of the ALOHA network if only one subscriber is active?

Solution

(a) *To compute the maximum throughput of a pure ALOHA network*

Transmission data rate = 1 Mbps (given)

For a large number of subscribers, each using a transmission rate of 1 Mbps to access a base station using ALOHA protocol, the maximum data rate that successfully passes through to the base station is given by

Maximum throughput of a pure ALOHA system, S_{max} (ALOHA),

$$S_{max}\ (\text{ALOHA}) = (G\, e^{-2G}) \times \text{transmission rate, for } G = 0.5$$

Therefore, S_{max} (ALOHA) $= (0.5\, e^{-2\times0.5}) \times 1$ Mbps
Or, $= 0.184 \times 1$ Mbps
Or, $= 184$ kbps

(b) *To determine the throughput of a TDMA network*

Transmission data rate = 1 Mbps (given)

In a TDMA system with negligible overhead (possible in long data packets), the throughput is nearly 100 per cent and the base station receives data at a maximum rate of 1 Mbps. Therefore,

Throughput of a TDMA network @ 1 Mbps rate = 1 Mbps

(c) *To determine the throughput of the ALOHA network for one active subscriber*

Transmission data rate = 1 Mbps (given)

Number of active subscribers = 1 (given)

If only one subscriber is active in the ALOHA network, there cannot be any collision and the only subscriber can transmit all the time. Therefore,

Throughput of the ALOHA network @ 1 Mbps rate = 1 Mbps

The advantage of pure ALOHA protocol is that it is very simple to implement. It does not require any synchronisation between subscribers. The subscribers transmit their data packets as and when they are ready for transmission. If there is a packet collision, they simply retransmit. The disadvantage of the pure ALOHA protocol is its low throughput under heavy traffic load conditions. Assuming that data packets having same length arrive randomly, and they are generated from a large number of subscribers, the maximum throughput of the pure ALOHA is about 18 per cent.

8.8.2 Slotted ALOHA Packet Radio Protocol

In wireless communication channels where bandwidth limitations often impose serious concerns for data communications applications, pure ALOHA technique is altered to its synchronised version referred to as slotted ALOHA. The performance of a pure ALOHA system can be improved by providing a framing structure. This framing structure includes fixed slot times, and subscribers are required to synchronise their transmissions with the slot times. Fig. 8.30 illustrates the basic concept of slotted ALOHA protocol.

In slotted ALOHA protocol, the transmission time is divided into equal time slots of length greater than the packet duration T as shown in the Figure. The base station transmits a pre-defined broadcast control signal, called beacon signal. All subscribers synchronise their time slots to this beacon signal. Each subscriber has a synchronised clock and transmits a data packet only at the beginning of a new time slot. This results in a discrete distribution of packets and prevents even partial collisions of data packets with one another. When a subscriber generates a packet of data, the packet is buffered and transmitted at the start of the next time slot. In this way, partial packet collision is eliminated.

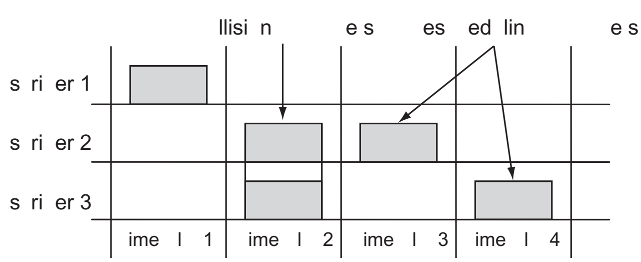

Fig. 8.30 *Basic concept of Slotted ALOHA protocol*

The delay characteristics of the traffic is determined primarily by the number of time slots which a subscriber waits prior to retransmitting the lost packet. The vulnerable period for slotted ALOHA is only one packet duration instead of two packet duration in pure ALOHA. The vulnerable period for slotted ALOHA is less due to prevention of partial collisions through synchronisation of packet transmissions. Assuming equal length of data packets, either there is a complete collision or no collisions. This doubles the throughput of the network. The retransmission mechanisms remain the same as in pure ALOHA. Because of its simplicity, the slotted ALOHA protocol is commonly used in the early stages of registration of a mobile subscriber to initiate a communication link with the base station in cellular mobile communication systems.

The timing of the slotted ALOHA frame is based on the timing of a forward-link broadcast channel. Slotted ALOHA is a modification of pure ALOHA having slotted time with the slot size equal to the duration of a packet T. If a subscriber has a packet to transmit, it waits before sending the packet until the beginning of the next time slot. Thus, the slotted ALOHA is an improvement over pure ALOHA by reducing the vulnerable period for packet collision to a single time slot. It clearly means that a packet transmission will be successful if and only if exactly one packet is scheduled for transmission for the current time slot.

Throughput of Slotted ALOHA With slotted ALOHA, a collision occurs only if the two subscribers transmit during the same time slot T. Since the sequence of the newly generated and retransmitted packets follows Poisson distribution process, the probability of successful transmission is given by

$$P_{sS} = e^{-\lambda T} \tag{8.9}$$

Consequently, the throughput S_S of a slotted ALOHA system is given by the product of the packet arrival rate and the probability that a packet is successfully received; that is,

$$\text{Throughput of a slotted AL\quad HA system } S_S = \lambda\, e^{-\lambda T} \tag{8.10}$$

Normalising this equation to packets received per packet time T, and then the normalised throughput is given by

$$\text{Normalised throughput of a slotted ALOHA system, } S_{0S} = \lambda\, T\, e^{-\lambda T} \tag{8.11}$$

Using the definition of the normalised offered load $G = \lambda\, T$,

$$S_{0S}\,(\text{Slotted ALOHA}) = G\, \mathrm{e}^{-G} \tag{8.12}$$

The probability that no other data packets will be generated during the vulnerable period is e^{-G}.
The peak throughput with slotted ALOHA occurs at $G = 1$, Therefore,

S_{0peak} (Slotted ALOHA) $= 1\, e^{-1} = 1 / e \approx 0.368$ packets per packet time.

That is, with a slotted ALOHA random-access protocol, the maximum throughput is 36.8% of the full channel capacity. This is double that of pure or unslotted ALOHA. In other words, the slotted ALOHA protocol provides a maximum channel utilisation of 0.368 Erlangs, double that of pure ALOHA protocol. Thus, the maximum throughputs of pure ALOHA and slotted ALOHA protocols are equal to 0.184 and 0.368, respectively.

The GSM cellular system uses slotted ALOHA protocol to establish a wireless communication link between the mobile subscriber and the base station. The initial contact between the mobile subscriber and the base station is performed through a random access channel using slotted ALOHA protocol to establish a traffic channel for TDMA voice communications. Other voice-oriented cellular communication systems adopt similar approaches as the first step in the registration process of a mobile subscriber.

8.8.3 Reservation ALOHA Protocol

Throughput of slotted ALOHA protocol is approximately 36 per cent which is very low for wireless data applications. This technique is sometimes combined with TDMA systems to form the so-called reservation-ALOHA (R-ALOHA) protocol, as shown in Fig. 8.31.

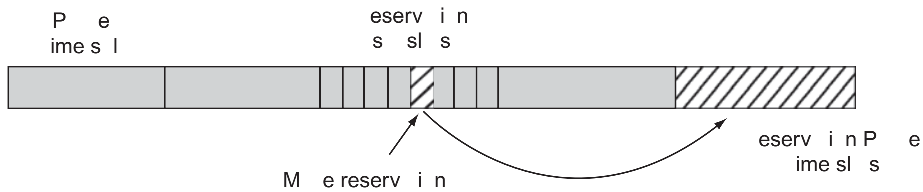

Fig. 8.31 Basic concept of reservation ALOHA protocol

In the R-ALOHA protocol, time slots are divided into contention periods as well as contention-free periods. During the *contention period*, a subscriber uses very short packets to contend for the upcoming contention-free intervals that will be used for transmission of longer data packets. In this protocol, certain packet slots are assigned with priority. It is possible for subscribers to reserve some packet slots for the transmission of data packets either permanently or on request basis. The reservation period is followed by a transmission period. Usually, a very small data packet is sent to reserve transmission time slot. Reservations of time slots on request basis offers better throughput for high traffic conditions although this scheme causes a larger delay even under a light load.

Based on the detailed implementation of R-ALOHA protocols, there are two distinct forms such as *Dynamic Slotted ALOHA protocol*, and *Packet Reservation Multiple Access (PRMA)*. In the dynamic slotted ALOHA protocol, the number of free slots of a certain length, each for contention, can change depending on the nature of traffic. The PRMA protocol is a method for transmitting a variable mixture of voice packets and data packets in a wireless environment. The PRMA system merges characteristics of slotted ALOHA and TDMA protocols and thus is closely related to R-ALOHA. PRMA uses a discrete packet time technique which is similar to reservation ALOHA and combines the cyclical frame structure of TDMA protocol in a way that allows each TDMA time slot to carry either voice or data. PRMA serves as a means of integrating speech packet and bursty data. Similar to TDMA systems, PRMA defines a frame structure. Within each frame, there are a fixed number of time slots which may be designated as either available or reserved, depending on the nature of traffic as determined by the controlling base station.

The key feature of packet reservation multiple access (PRMA) protocol is the utilisation of subscriber data transmission to gain access to the radio resources. Once the radio resource has been acquired, it is up to the base station to release the reservation. If a base station encounters traffic congestion from many mobile subscribers, data packets are dropped and speech packets are given priority since speech requires that the speech packets be delivered in order.

Facts to Know !

PRMA has an advantage to increase capacity of the radio channel by utilising the discontinuous nature of speech with the help of a voice activity detector (VAD). PRMA supports both analog speech data and digital packet data simultaneously.

In many communication systems, the packet is retransmitted if there is a collision. For small number of retransmissions, the previous results hold. But in case of high collision rate, large number of retransmissions can add appreciably to the total traffic load presented to the network at any particular time. With random-access protocols, collisions cause delays in delivering the data packets. The delay depends upon the propagation delay as well as the retransmission strategy. Consequently, it is difficult to compare directly the delays of different random-access techniques.

Retransmission of collided packets cannot occur as soon as a collision is detected because that will result in another collision with high probability. Instead, there has to be a random back-off wait period for retransmissions. However, it is assumed that there are no packet errors due to external noise and interference.

Hence, there is always a tradeoff between the two key parameters — throughput and delay. The throughput approaches zero, and the average delay per packet approaches infinity with high traffic load.

Table 8.2 shows the multiple access techniques which should be used for different types of traffic conditions.

Table 8.2 *Multiple access techniques for different types of traffic*

S. No.	*Type of traffic*	*Multiple access technique*
1.	Voice	Contention-free multiple access techniques such as FDMA, TDMA, CDMA
2.	Bursty, short messages	Contention-based protocols such as Pure ALOHA, Slotted ALOHA
3.	Bursty, long messages, large number of subscribers	Reservation protocols – Reservation ALOHA
4.	Bursty, long messages, small number of subscribers	Reservation protocols with fixed TDMA reservation channel – PRMA

There is a need to find another way of improving throughputs further to support high-speed wireless communication networks. It is possible to achieve better throughput if possible collision can be prevented by simply listening to the common channel before transmitting a packet. In this way, collisions of data packets could be avoided. This is known as a Carrier Sense Multiple Access (CSMA) protocol. The IEEE 802.11 standard for WLANs employs a version of the CSMA protocol.

8.9 CARRIER SENSE MULTIPLE ACCESS PROTOCOLS

The Carrier Sense Multiple Access (CSMA) protocol involves sensing the presence of a carrier and accessing the medium only if the carrier is idle. This mechanism enables to prevent potential collision of data packets before transmitting them. In this way, collisions could be avoided; and overall throughput can be improved as compared to the maximum throughputs of 0.184 and 0.368 achieved in pure and slotted ALOHA protocols, respectively.

In the CSMA protocol, each subscriber can sense the transmission of all other subscribers, and the propagation delay is generally kept small as compared with the transmission time of a data packet. There are several variants of the basic CSMA protocols, as described briefly here.

8.9.1 Nonpersistent CSMA Protocol

In the nonpersistent CSMA protocol, the subscriber senses the medium first whenever the subscriber has a packet to send. If the medium is busy, the subscriber waits for a random amount of time and senses the medium again. If the medium is idle, the subscriber transmits the packet immediately. If a collision occurs, the subscriber waits for a random amount of time and starts transmitting the packet all over again. When the packets are transmitted at any arbitrary time, it leads to *unslotted nonpersistent CSMA*. When the packets are sent during a slotted period, it leads to *slotted nonpersistent CSMA*.

8.9.2 Persistent CSMA Protocol

In the persistent CSMA protocol, whenever the subscriber has a packet ready to be transmitted, it senses the medium. If the medium is busy, the subscriber keeps listening to the medium and transmits the packet immediately after the medium becomes idle. Whenever the subscriber finds the medium to be idle, it transmits with a probability of 1 (100%), this protocol is called *1-persistent CSMA protocol*. However, in this protocol, there will always be a collision if two or more subscribers are waiting for the medium to become free, have packets ready to be transmitted, and start transmitting at the same time. This will cause many collisions if many subscribers wish to send packets and block each others' transmission.

This situation can be improved if the time can be slotted in such a way so as to lead toward *p-persistent CSMA protocol*. In this protocol, the subscriber senses the medium when it has a packet to send. If the medium is busy, the subscriber waits until the next slot and checks the medium again. If the medium is idle, the subscriber transmits with probability p (less than 1) or defers transmission with probability $(1 - p)$ until the next slot. If a collision occurs, the subscriber waits for a random amount of time and starts all over again. To create fairness for some subscribers waiting for a longer time and some subscribers waiting for a shorter period, well-defined back-off algorithms are implemented. Intuitively, this protocol is considered as an optimal access strategy.

8.9.3 CSMA with Collision Detection (CSMA/CD)

In a typical CSMA protocol, if two subscribers begin transmitting at the same time, each subscriber will transmit its complete packet, even though the packet may collide. This results into wastage of the medium for an entire packet time and can be addressed by a new protocol version called CSMA with collision detection (CSMA/CD) protocol. In this protocol, the transmission of packets is terminated immediately after detection of a collision. The subscriber senses the medium when it has a packet to be transmitted. If the medium is free, the subscriber transmits its packet immediately. If the medium is busy, the subscriber waits until the medium becomes idle. However, if a collision is detected during the transmission, the subscriber aborts its transmission immediately. It attempts to transmit again after waiting for a random amount of time. Although the original CSMA protocol minimises the number of collisions, the CSMA/CD protocol can further reduce the effect of a collision as it renders the medium ready to be used as soon as possible. The collision-detection time is at the most two times end-to-end propagation delay time. This results in the improvement in throughput of the slotted non-persistent CSMA/CD protocol.

Facts to Know!

An efficient use of resources is especially important in requesting access to the base station so that it can assign exclusive access to individual traffic channels to each requesting subscriber using one of the multiple access techniques in a wireless cellular system.

8.9.4 CSMA with Collision Avoidance (CSMA/CA)

A modified version of the CSMA/CD protocol is called CSMA with collision avoidance (CSMA/CA) protocol has been adopted by the IEEE 802.11 standard and is used in wireless LAN applications. The IEEE 802.11 WLAN standard supports operation in two distinct modes: a distributed coordination mode and a centralised point-coordination mode. In the basic CSMA/CA protocol technique, all subscribers sense the medium in the same way as in the CSMA/CD protocol. A subscriber that is ready to transmit a data packet senses the medium. The subscriber transmits its data packet if the medium is idle for a time interval that exceeds the pre-defined distributed interframe space. Otherwise, the subscriber waits for an additional predetermined time period, and then picks a random backoff period within its contention window to wait before transmitting its data packet.

The backoff period is used to initialise the backoff counter. When the medium is idle, the backoff counter can count down. Otherwise, it is frozen as soon as the medium becomes busy. After the busy period, the backoff counter resumes its counting down only after the medium has been free longer than the distributed interframe space. The subscriber can start transmitting its data packets when the backoff counter becomes zero. Collisions can still occur in case two or more subscribers select the same time slot to transmit their packets.

CSMA/CA with ACK In this protocol, an immediate positive acknowledgment (ACK) is included to indicate a successful reception of each data packet. It is obvious that explicit ACKs are required in wireless transmissions since a transmitter cannot listen to its own data transmissions while transmitting and hence cannot determine if the data packet is successfully received which is otherwise possible in the case of wired LANs. This is accomplished by making the receiving-end subscriber send an acknowledgment packet immediately after a time interval of another pre-defined short interframe space. If an ACK is not received, the data packet is presumed to be lost and a retransmission is automatically scheduled by the transmitting-end subscriber.

CSMA/CA with RTS and CTS This protocol involves an alternative way of transmitting data packets by using a special handshaking mechanism. It sends request to send (RTS) and clear to send (CTS) packets prior to the transmission of the actual data packet. A successful exchange of RTS and CTS packets attempts to reserve the medium for the entire time duration required to transfer the data packet under consideration within the transmission ranges of sender subscriber and receiver subscriber.

The rules for the transmission of an RTS packet are the same as those for a data packet under the basic CSMA/CA protocol. It means that the transmitting-end subscriber sends an RTS packet after the medium has been idle for a time interval exceeding the distributed interframe space. On receiving an RTS packet, the receiving-end subscriber responds with a CTS packet. The CTS packet acknowledges the successful reception of an RTS packet, which can be transmitted after the medium has been idle for a time interval exceeding short interframe space. After the successful exchange of RTS and CTS packets, the data packet can be sent by the transmitter after waiting for a time interval equal to short interframe space. RTS packet is retransmitted following the backoff rule as specified in the CSMA/CA with ACK procedures.

> **Facts to Know!**
>
> CSMA/CA with RTS and CTS type multiple access protocol is important to avoid the presence of a garbled packet. This also utilises the available bandwidth optimally. This aspect gains more significance when it becomes utmost essential to minimise collisions among more than one subscriber using the same channel in a wireless environment.

8.10 MULTICARRIER MULTIPLE ACCESS SCHEMES

Multicarrier multiple access schemes use multiple carrier signals at different frequencies, sending some of the bits on each channel. There are a number of such schemes which find application in advanced wireless communication systems.

8.10.1 OFDMA

Orthogonal Frequency-Division Multiple Access (OFDMA) is a multi-user version of the Orthogonal Frequency-Division Multiplexing (OFDM) digital-modulation scheme. This is similar to FDM in concept. However, in the case of OFDM, all of the sub-channels are dedicated to a single data source. The OFDM scheme uses advanced digital signal-processing techniques to distribute the data over multiple carriers at precise frequencies. The precise relationship among the subcarriers is referred to as *orthogonality*. This means that the peaks of the power spectral density of each subcarrier occur at a point at which the power of other subcarriers is zero. With OFDM, the subcarriers can be packed tightly together because there is minimal interference between adjacent subcarriers.

Fig. 8.32 illustrates the concept of OFDM.

Let there be a data stream operating at R bps and an available bandwidth of $(N \times \ {}_b)$, where N is an integer related to the number of subcarriers, and ${}_b$ is the base frequency, centered at ${}_0$. The entire bandwidth could be used to send the data stream, in which case each bit duration would be $1/R$. The alternative is to split the data stream into N substreams, using a serial-to-parallel converter. Each substream has a data rate of R/N bps

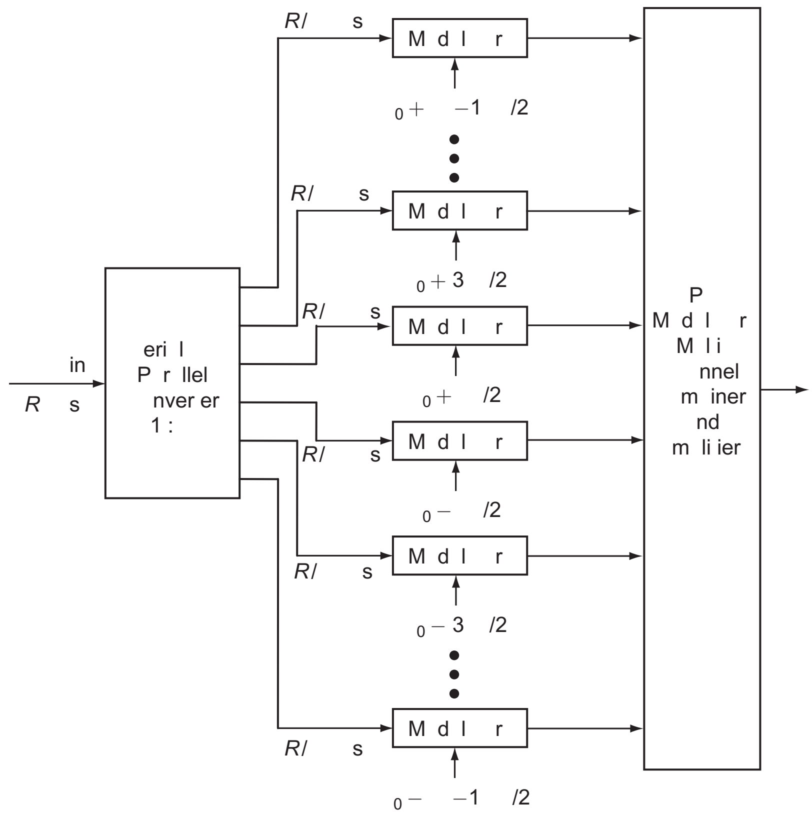

Fig. 8.32 Concept of Orthogonal Frequency Division Multiplexing

and is transmitted on a separate subcarrier, with a spacing between adjacent subcarriers of $_{b}$. Now the bit duration is N/R. The base frequency, $_{b}$ is the lowest-frequency subcarrier. All of the other subcarriers are integer multiples of the base frequency, namely 2 $_{b}$, 3 $_{b}$, and so on.

For transmission, the set of OFDM subcarriers is further modulated to a higher frequency band. A digital modulation scheme used with OFDM is Quadrature Phase Shift Keying (QPSK). In this case, each transmitted symbol represents two bits. To minimise ISI, data are transmitted in bursts, with each burst consisting of a cyclic prefix followed by data symbols. The cyclic prefix is used to absorb transients from previous bursts caused by multipath. The resulting waveform created by the combined multipath signals is not a function of any sample from the previous burst.

OFDM has several advantages. First, frequency-selective fading affects some subchannels only and not the complete signal. If the data stream is protected by a forward error-correcting code, the impact of frequency-selective fading can be minimised. Moreover, OFDM overcomes intersymbol interference in a multipath environment. ISI has a greater impact at higher data rates because the time duration between symbols is smaller. With OFDM, the data rate is reduced by a factor of N, which increases the symbol period by a factor of N. Thus, if the symbol period is T for the data source stream, the period for the OFDM signals is $N \times T$. This reduces the effect of ISI. Usually, N is chosen so that $N \times T$ is much greater than the root-mean-square delay spread of the wireless fading channel. As a result of these design considerations, it may not be necessary to include equalisers with the use of OFDM.

Multiple access in OFDMA is achieved by assigning subsets of subcarriers to individual users, thus allowing simultaneous low-data-rate transmission from several users as well as to support differentiated Quality of Service (QoS), that is, to control the data rate and error probability individually for each user. Based on feedback information about the channel conditions, adaptive user-to-subcarrier assignment can be achieved. By employing a sufficiently fast assignment, a significant improvement in robustness to fast fading and narrow-band cochannel interference can be obtained. This makes OFDMA scheme achieve even better system spectral efficiency.

Facts to Know!

OFDMA is considered as highly suitable for broadband wireless networks, due to advantages including scalability and MIMO-friendliness, and ability to take advantage of channel frequency selectivity. In spectrum sensing cognitive radio, OFDMA is a possible approach to filling free radio frequency bands adaptively. OFDMA is used in the mobility mode of the IEEE 802.16 Wireless MAN standard, commonly referred to as WiMAX.

8.10.2 SC-FDMA

Single-carrier FDMA (SC-FDMA) is a frequency-division multiple access scheme. SC-FDMA is a new multiple access technique, which utilises single-carrier modulation, DFT-spread orthogonal frequency multiplexing, and frequency domain equalisation. It has similar structure and performance to OFDMA. SC-FDMA is currently adopted as the uplink multiple access scheme in 3GPP, and a variant of SC-FDMA using code spreading is used in 3GPP2 uplink. IEEE 802.16 is also considering it for uplink.

SC-FDMA can be viewed as a linearly precoded OFDMA scheme (LP-OFDMA). It can also be viewed as a single-carrier multiple access scheme. In fact, it is a multi-user version of the Single-Carrier Frequency-Domain-Equalisation (SC-FDE) modulation scheme. Single Carrier FDMA (SC-FDMA) is an extension of SC-FDE to accommodate multiple-user access. SC-FDMA is also regarded as DFT-precoded or DFT-spread OFDMA. The main advantage of SC-FDE and SC-FDMA/LP-OFDMA signals over conventional OFDM and OFDMA signals is that they have lower peak-to-average power ratio (PAPR) because of its inherent single carrier structure.

In SC-FDMA, guard intervals with cyclic repetition are introduced between blocks of symbols in view to efficiently eliminate time spreading (caused by multi-path propagation) among the blocks, similar to that available in OFDM. In OFDM, inverse FFT (IFFT) on the transmitter side, and Fast Fourier Transform (FFT) is applied on the receiver side on each block of symbols. In SC-FDE, both FFT and IFFT are applied on the receiver side, but not on the transmitter side, whereas in SC-FDMA, both FFT and IFFT are applied on the transmitter side as well as on the receiver side.

In SC-FDMA, multiple access is made possible by inserting silent Fourier-coefficients on the transmitter side before the IFFT, and removing them on the receiver side before the IFFT. Different users are assigned to different Fourier-coefficients (sub-carriers).

In SC-FDMA, equalisation is achieved on the receiver side after the FFT calculation, by multiplying each Fourier coefficient by a complex number. This operation is identical to that of in OFDM as well as SC-FDE. Thus, it is more capable of combating frequency-selective fading and phase distortion. One of the main advantages is that frequency domain equalisation and FFT requires less computation power than conventional time-domain equalisation.

Fig. 8.33 depicts the pictorial representation of the relationship among SC-FDMA, OFDMA, and DS-CDMA/FDE.

In terms of bandwidth expansion, SC-FDMA is very similar to the DS-CDMA system using orthogonal spreading codes. Both spread narrowband data into broader band. Time symbols are compressed into chips after modulation, and spreading gain (processing gain) is achieved. SC-FDE or SC-FDMA delivers performance similar to OFDM with essentially the same overall complexity, even for long channel delay. It has advantage over OFDM in terms of low PAPR, robustness to spectral null, and less sensitivity to carrier frequency offset. Its disadvantage to OFDM is that channel-adaptive subcarrier bit and power loading is not possible.

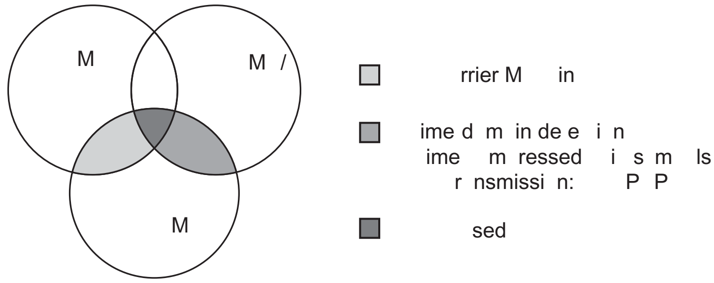

Fig. 8.33 *Relationship among SC-FDMA, OFDMA, and DS-CDMA/FDE*

8.10.3 MC-CDMA

The Multi-Carrier Code Division Multiple Access (MC-CDMA) scheme is a combination of OFDM and DS-CDMA. MC-CDMA shows high envelope power fluctuation as in OFDM. For an N-subcarrier system, the peak power becomes N times the average power in the worst case and the signal is distorted in the RF power amplifiers, yielding spurious power emission. To reduce the distortion, the operating point in the amplifiers can be backed off, but this may lead to inefficient power usage.

MC-CDMA maintains the original signaling interval while it spreads the signal over wide bandwidth like DS-CDMA. To transmit 1 Mbps data with the processing gain of 20 dB, the chip rate required in DS-CDMA is 100 Mcps, which necessitates four times faster internal digital front-end processor or at least a 100-MHz analogue matched filter. This requirement can be easily achieved by using multi-code assignment for high-speed data rate but at the cost of reduced user capacity.

> **Facts to Know!**
>
> MC-CDMA requires the conventional cell planning in cellular environment by using a PN code. Thus, it loses one of the greatest benefits of DS-CDMA, which is the universal frequency reuse.

Small delay spread and small Doppler spread enable the MC-CDMA scheme to work efficiently. Small delay spread reduces the guard interval in MC-CDMA and makes it power efficient. MC-CDMA is sensitive to frequency offset and small Doppler spread is preferred. The difference in the arrival times of multipath signals in indoor wireless environment is typically much less than 1 μs. The multipath resolvability is proportional to the user chip rate. To make the rake receivers work properly, the chip rate should be much faster than 1 Mcps even when there is no need for high data rate service. In such a situation, the MC-CDMA scheme is a viable alternative.

When there is a deep frequency-selective fading, OFDM loses the corresponding data on corrupted subcarriers. As MC-CDMA spreads an information bit over many subcarriers, it can make use of information contained in some subcarriers to recover the original symbol. MC-CDMA gathers nearly all the scattered powers effectively using the cyclic prefix insertion technique. As the received signals are sampled at the original symbol rate in MC-CDMA, the sampling points may not be optimum. In general, the performance of MC-CDMA is equivalent to the m-finger rake receiver in DS-CDMA, where m is the number of symbols in cyclic prefix of MC-CDMA. Various types of frequency domain equalisers are used for MC-CDMA which perform better than rake receivers used for DS-CDMA.

8.10.4 MC-DS-CDMA

The Multi-Carrier Direct Sequence Code Division Multiple Access (MC-DS-CDMA) scheme is a combination of time-domain spreading and Orthogonal Frequency Division Multiplexing (OFDM), while Multi-Carrier Code Division Multiple Access (MC-CDMA) is a combination of frequency-domain spreading and OFDM. In MC-CDMA, a good Bit Error Rate (BER) performance can be achieved by using Frequency-Domain Equalisation (FDE), since the frequency diversity gain is obtained. On the other hand, conventional MC-DS-CDMA cannot obtain the frequency diversity gain. However, MC-DS-CDMA can obtain the frequency diversity gain by applying a Frequency Domain Equaliser (FDE) to a block of a number of OFDM symbols.

For broadband multi-path channels, conventional time domain equalisers are impractical because of complexity, very long channel impulse response in the time domain, and prohibitively large tap size for the time-domain filter. On the other hand, using Discrete Fourier Transform (DFT), equalisation can be done in the frequency domain. Because the DFT size does not grow linearly with the length of the channel response, the complexity of FDE is lower than that of the equivalent time-domain equaliser for a broadband channel. Most of the time-domain equalisation techniques such as MMSE equaliser, DFE, and turbo equaliser can be implemented in the frequency domain.

Key Terms

- ALOHA
- Bands
- Bandwidth
- Binary exponential backoff
- Collision
- CSMA
- CSMA/CA
- CSMA/CD
- Direct Sequence Spread Spectrum (DSSS)
- Downlink
- Fast FHSS
- FHSS
- Frame efficiency
- Frequency Division Duplexing (FDD)
- Frequency Division Multiple Access (FDMA)
- Frequency hopping
- MACA
- Packet Radio access
- PRMA
- Random access
- RTS/CTS
- Signal-to-noise ratio
- Slow FHSS
- Smart antennas
- Spread spectrum
- Spreading code
- Time Division Duplexing (TDD)
- Time Division Multiple Access (TDMA)
- Time slots
- Uplink

Summary

In this chapter, various multiple access techniques that are used in analog and digital cellular mobile standards are discussed in detail. The efficient ways to access the limited radio spectrum by number of potential subscribers simultaneously in a wireless-environment span through division of space, frequency, time and code are discussed. For an efficient use of resources by multiple number of subscribers simultaneously, multiple access techniques such as FDMA, TDMA, or CDMA are used in a wireless cellular system. Thus communication channels are used by system subscribers and there are many multiple access techniques that can be used effectively. Their relative advantages and disadvantages have been outlined here and problems and limitations of using such resources have been widely discussed. A number of subscribers need to access the control channel on shared basis at random times and for random periods. Controlling access to a shared medium is important from the point of view that, at any given time, only one subscriber is allowed to talk while the rest of the subscribers listen. An overview of packet radio multiple access technique is also presented here. A functional schematic of basic cellular system along with its operation and performance criteria is the main attention in the next chapter.

Important Equations

- ❑ $N = (B_t - 2\,B_g) / B_c$ (8.1)
- ❑ $N = m \times (B_t - 2B_g) / B_c$ (8.2)
- ❑ Throughput of an ALOHA system, $S\,(\text{ALOHA}) = \lambda\, e^{-2\lambda T}$ (8.6)
- ❑ Throughput of a slotted ALOHA system, $S_S = \lambda\, \mathrm{e}^{-\lambda T}$ (8.10)

Short-Answer Type Questions with Answers

A8.1 How does time-division duplexing function
Time Division Duplexing (TDD) uses time instead of frequency to provide both a forward and reverse link simultaneously. Multiple mobile subscribers share a single radio channel by taking their respective turns for data transmission in the time domain. The time separation between the forward and reverse time slot is usually very small, and the transmission and reception of voice/data appears to be continuous to the subscribers.

A8.2 How can desired signal quality be maintained in FDMA systems
To ensure acceptable signal quality in FDMA systems, it is essential that each frequency channel signal be kept confined to the assigned channel bandwidth, and the out-of-band signal energy causes negligible interference to the subscribers using adjacent channels. In order to minimise adjacent channel interference, the power spectral density of the modulated signal is controlled so that the power radiated into the adjacent band is at least 60 to 80 dB below that in the desired band with the use of highly selective filters in the system design. In addition, guard bands are inserted as buffer frequency zones in adjacent channels.

A8.3 Suggest some measures which can be adopted in FDMA cellular systems to overcome the problem of near–far interference.
Channel assignment is done in such a way so that the frequencies in each cell are grouped as far apart as possible from each other. Guard bands are included in the frequency channel to further reduce adjacent channel interference. The transmitter power of the mobile subscribers is controlled so as not to cause interference to other transmissions in the cell.

A8.4 Why is FDMA usually implemented in narrowband communication systems
The bandwidths of FDMA channels are relatively narrow (for example, 30 kHz in AMPS) as each channel supports only one communication link per carrier. After the assignment of a voice channel, the base station and the mobile subscriber transmit simultaneously and continuously. If the assigned channel is not in use then it remains idle and cannot be used by other mobile subscribers. This is clearly a wastage of spectrum resource. The utilisation of the channel during free time is essential to increase system capacity. Therefore, FDMA is usually implemented in narrowband communication systems.

A8.5 What is not required to implement equalisation in FDMA narrowband systems
Due to continuous transmission in FDMA systems, fewer bits are needed for synchronisation and framing purpose. The symbol time of a narrowband signal is large as compared to the average delay spread. This implies that the amount of intersymbol interference is also low. So there may not be any requirement to implement equalisation in FDMA narrowband systems.

A8.6 Differentiate between TDMA/FDD and TDMA/TDD systems.
In TDMA/FDD systems, the carrier frequency bands are different but TDMA frame structures are same for both forward and reverse channels. In TDMA/TDD systems, both forward and reverse channels

use the same frequency band but they use alternating time slots in the same TDMA frame. Both systems provide full duplex communication.

A8.7 Why is it necessary to include signalling and control data bits in every TDMA frame
Signalling and control data bits perform the functions such as synchronisation and frame error rate which assist the receiver to recover sinusoidal carrier essential for coherent detection and to estimate the unknown impulse response of the wireless channel respectively. This is needed for reliable decoding of the received signal.

A8.8 Define frame efficiency of a TDMA system.
The frame efficiency of a TDMA system is defined as the number of bits representing digitised speech, expressed as a percentage of the total number of bits including the control overhead bits that are transmitted in a TDMA frame.

A8.9 What is meant by direct sequence spread spectrum modulation
Direct Sequence Spread Spectrum (DSSS) is a modulation technique wherein a pseudorandom sequence directly phase modulates a data-modulated carrier signal. Direct sequence spread spectrum systems are so called because a high-speed PN code sequence is employed along with the slow-speed information data being sent, to modulate their RF carrier. This results into considerable increase in the bandwidth of the transmission and reduction in the spectral power density.

A8.10 How is wideband frequency spectrum generated in a frequency-hopping technique
Frequency hopping involves a periodic change of transmission frequency over a wide band. The rate of hopping from one frequency to another is a function of the information rate, and the specific order in which frequencies are occupied is a function of a code sequence. A pseudorandom frequency hopping sequence is used to change the radio signal frequency across a broad frequency band in a random manner.

A8.11 Classify frequency-hopping technique.
Frequency hopping may be classified as fast frequency hopping or slow frequency hopping. *ast frequency hopping* occurs if there is more than one frequency hop during each transmitted symbol. It implies that the hopping rate equals or exceeds the information symbol rate. *Slow frequency hopping* occurs if one or more symbols are transmitted in the time interval between frequency hops. A frequency hopper may be fast hopped where there are multiple hops per data bit. It may be slow hopped where there are multiple data bits per hop.

A8.12 Distinguish between FDMA and FHMA systems.
The difference between FDMA and FHMA systems is that the frequency-hopped signal changes channels at relatively rapid intervals. FDMA systems employ analog frequency modulation scheme, whereas FHMA systems often employ energy-efficient constant envelope digital-modulation scheme. A fast frequency-hopping system may be thought of as an FDMA system, which employs frequency diversity.

A8.13 How is the near–far interference problem overcome in CDMA systems
The received signals at the cell-site from a faraway mobile subscriber could be masked by received signals from a close-by mobile subscriber in the reverse channel. This phenomenon is termed the near–far interference problem. This problem can be overcome by using automatic power control techniques that enable to adjust the mobile transmitting power, and achieve high efficiency of frequency utilisation in CDMA systems.

A8.14 What is the principle of operation of the SDMA technique
The SDMA technique is based on the principle of the directional transmission of wireless communications and relies on the deployment of smart antennas at the cell-site. The wireless communication space is omni-directional by nature. It can be divided into spatially separable sectors. These different areas in space covered by the respective antenna beam at the cell-site may be served by the different frequencies in an FDMA system or same frequency in a TDMA or SSMA system. SDMA techniques control the radiated energy for each subscriber in space by using directional or spot beam antennas at the cell-site.

A8.15 What is Packet Radio Multiple Access (PRMA) technique Classify them.

PRMA techniques are basically contention-based protocols in which many subscribers make an attempt to access the common channel either in a random access or collision resolution manner. The contention-based protocols used in mobile data networks can be classified into two groups according to the ways collisions are resolved: random access protocols (without any coordination among them) and collision resolution protocols (with limited coordination).

Self-Test Quiz

S8.1 The differentiation between the carrier frequencies of the forward channels and reverse channels is an important design parameter related to technique.

(a) FDMA (b) TDMA
(c) CDMA (d) SDMA

S8.2 The difference between the received signal levels in an open area at two mobile subscribers located at 100 metres and 1 km away from a cell-site respectively (other factors remaining constant) is

(a) 20 dB (b) 40 dB
(c) 80 dB (d) 100 dB

S8.3 The most critical feature of TDMA operation is

(a) dividing the carrier channel bandwidth into time slots
(b) assignment of time-slots among multiple subscribers
(c) time synchronisation to the incoming TDMA frame
(d) providing different access rates to subscribers

S8.4 The guard time between the time slots in a TDMA frame helps in minimising the interference due to along different radio paths in the wireless channel.

(a) propagation delays
(b) adjacent channel
(c) multipath fading
(d) timing inaccuracies

S8.5 To mitigate the inter-symbol interference problem in TDMA systems, technique has to be provided.

(a) source coding
(b) channel coding
(c) interleaving
(d) channel equalisation

S8.6 technique allows multiple subscribers to simultaneously occupy the same frequency spectrum at the same time.

(a) FDMA (b) SSMA
(c) FHMA (d) SDMA

S8.7 The use of a(n) at the cell-site maximises the antenna gain in the desired direction.

(a) omnidirectional antenna
(b) high-gain directional antenna
(c) switched-beam antenna
(d) dish antenna

S8.8 The techniques are suitable for bursty type traffic in the form of packets.

(a) TDMA (b) SSMA
(c) FHMA (d) PRMA

S8.9 Throughput of a pure ALOHA system is given by

(a) $\lambda\, e^{-2\lambda T}$ (b) $\lambda\, e^{-\lambda T}$
(c) $(1/\lambda)\, e^{-2\lambda T}$ (d) $(1/\lambda)\, e^{2\lambda T}$

where λ is the packet arrival rate, and T is the packet duration.

S8.10 With a slotted ALOHA random-access protocol, the maximum throughput is of the full channel capacity.

(a) 18.4% (b) 36.8%
(c) 50% (d) 100%

S8.11 Throughput of slotted ALOHA protocol is for wireless data applications.

(a) very high (b) high
(c) low (d) very low

S8.12 For bursty long messages and small number of subscribers, the multiple access technique used is
(a) Pure ALOHA
(b) Slotted ALOHA
(c) Reservation ALOHA
(d) PRMA

S8.13 The IEEE 802.11 standard for WLANs employs a version of ______ protocol.
(a) ALOHA (b) PRMA
(c) CSMA (d) TDMA

S8.14 As the throughput approaches zero, the average delay per packet approaches ______ with high traffic load.
(a) infinity (b) unity
(c) zero (d) 0.5

Answers to Self-Test Quiz

S8.1 (a); S8.2 (b); S8.3 (c); S8.4 (a); S8.5 (d); S8.6 (c); S8.7 (c); S8.8 (d); S8.9 (a); S8.10 (b); S8.11 (d); S8.12 (d); S8.13 (c); S8.14 (a)

Review Questions

Q8.1 Explain how the following multiple access techniques are used in wireless communications:
(a) FDMA
(b) TDMA
(c) SDMA

Q8.2 What is the difference between the guard band and the guard time? Why are they important in a cellular system?

Q8.3 List the advantages of CDMA over FDMA and TDMA.

Q8.4 Explain the type of interference in frequency, time, code, and space domain for multiple access techniques. What are the possible solutions to reduce interference in each one of them?

Q8.5 How does the near–far effect influence CDMA systems? What are the countermeasures available in CDMA systems?

Q8.6 How are multiple access radio protocols classified? Distinguish among various types of multiple access protocols.

Q8.7 What are the key issues for contention-based multiple access protocols? How is it different from contention-free multiple access techniques?

Q8.8 How does multiple radio access protocol slotted ALOHA improve the throughput as compared to pure ALOHA?

Q8.9 In a an ALOHA network, how does the subscriber come to know that its transmitted packet has collided?

Q8.10 Compare the features of pure ALOHA and slotted ALOHA protocols.

Q8.11 What are the merits and demerits of contention-based and non-contention-based protocols?

Q8.12 Explain the difference between carrier-sensing mechanisms between wired and wireless channels.

Q8.13 What are the relative advantages and disadvantages of persistent and non-persistent CSMA protocols?

Q8.14 Can CSMA/CD be used in cellular wireless networks? If not, why?

Q8.15 What is the difference between collision detection and collision avoidance? What are the major factors affecting the throughput of CSMA/CA?

Q8.16 What are the purposes of using RTS/CTS in CSMA/CA? List the relative advantages and disadvantages of basic CSMA/CA and CSMA/CA with RTS/CTS protocols.

Analytical Problems

P8.1 If 20 MHz of total spectrum is allocated for a duplex wireless cellular system and each duplex channel has a 25-kHz RF bandwidth, find the number of duplex channels. Assume the system uses FDMA technique.

P8.2 In a TDMA system operating at a signal transmission rate of 512 kbps, the worst-case difference in the path lengths traversed by a signal is 4 km from the BS to a subscriber along direct and reflected paths. Estimate the minimum value of guard time that must be used and the separation in bits.

P8.3 A TDMA-GSM cellular system uses a TDMA frame structure where each frame consists of eight time slots, and each time slot contains 156.25 bits, and data is transmitted at 270.833 kbps in the channel. Find

(a) the time duration of a time-slot
(b) the time duration of a TDMA frame

P8.4 A normal TDMA-GSM time slot consists of six trailing bits, 8.25 guard bits, 26 training bits, and two traffic bursts of 58 bits of data. Compute

(a) the number of bits in a time slot
(b) the number of bits in a frame
(c) the number of overhead bits in a frame
(d) frame efficiency

P8.5 In a pure ALOHA system the packet arrival sequence forms a Poisson process having a rate of 10^3 packets/sec. If the bit rate is 10 Mbps and there are 1000 bits/packet, find

(a) the normalised throughput of the system
(b) the number of bits per packet that will maximise the throughput

P8.6 In a slotted ALOHA system, the packet arrival rate is 10^3 packets/sec in a Poisson process. If the bit rate is 10 Mbps and there are 1000 bits/packet, find

(a) the normalised throughput of the system
(b) the number of bits per packet that will maximise the throughput

P8.7 Determine the propagation delay in packet transmission units if a 19.2 kbps channel data rate is used and each packet contains 256 bits. Assume a line-of-sight radio path exists for a mobile subscriber 10 km away from the base station. If slotted ALOHA is to be used for this system, what is the optimum choice for the number of bits/packet for this system (assuming that 10 km is the maximum distance between the transmitter and receiver)?

P8.8 In a CSMA/CA protocol, a random delay is allowed whenever a collision occurs. What is the guarantee that future collision between previously collided subscribers will not reoccur?

P8.9 In CSMA/CA with RTS/CTS protocol, let the propagation delay be 10 ms. If distributed interframe space is 30 ms, short interframe space is 10 ms, RTS and CTS are 50 ms then show that the earliest time for the receiver to send the CTS message is 100 ms.

P8.10 Suppose the propagation delay is 5 ms in CSMA/CA with RTS/CTS protocol. If distributed interframe space is 15 ms, short interframe space is 5 ms, RTS and CTS are 25 ms then show that the shortest time for the receiver to send the ACK signal is 1095 ms, if the data packet is 1000 ms long.

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9

A Basic Cellular System

A cellular system is designed to establish calls among mobile subscribers and connectivity with landline subscribers. This chapter begins with the limitation of conventional mobile telephone systems and how these limitations are overcome in a cellular system. The basic functional block diagram and operations are described here. Various performance metrics are discussed in detail, and finally an overview of the main aspects of cellular-system planning leading to optimum utilisation of network resources is given.

9.1 LIMITATIONS OF CONVENTIONAL MOBILE TELEPHONE SYSTEM

The primary motivation for development of cellular mobile telephone communication systems is to overcome some of the major operational limitations of conventional mobile telephone systems. The conventional mobile systems deploy a high-transmitter-power base station in a large autonomous geographical service area providing limited service capability. The inefficient frequency spectrum utilisation, low subscriber capacity, poor service performance, high blocking probability during busy hours, and no continuation of call between different service areas are the major concerns in conventional mobile systems. The cellular concept is a major breakthrough in resolving the issues of spectral congestion and subscriber capacity, besides offering many utility services to highly mobile subscribers. It can provide very high subscriber capacity in a limited spectrum allocation without any major technological changes. But cellular communication systems are much more complex and require sophisticated computing power in the mobile phone.

Facts to Know!

Using a cellular telephone today is as simple as using a traditional wired telephone. In fact, the cellular telephone is replacing the existing landline telephone lines in many cases. Moreover, in many parts of the world, especially where telephone cable infrastructure does not exist, cellular phone services are being made available to users.

9.2 CONSIDERATION OF COMPONENTS OF A CELLULAR SYSTEM

A basic cellular system comprises of many low-power transmitters, each specifically designed to serve only a small area called a cell. Cellular systems rely on the frequency reuse concept, which requires that the frequency in neighbouring cells should be different. The same frequency could be reused in different cells separated from each other so as to cause negligible interference effects among active subscribers using the same channel. Each geographic area or cell can generally service many different frequency channels simultaneously. The number of user channels (frequency, time slots or codes) in a cell depends on the corresponding multiple access technique used. For example, within a cell, each radio-frequency channel can support up to 20 mobile subscribers at any time.

The assignment of channels to mobile subscribers may be static or dynamic. Statically assigned channels are allocated to a mobile subscriber for the duration of a call. Dynamically assigned channels are allocated to a mobile subscriber only when it is being used. With both static and dynamic channel assignments, mobile subscribers can be allocated any available radio channel at that time.

By setting a relatively small number of forward control channels as part of the common air interface, cellular mobile subscribers can rapidly scan all the possible forward control channels to determine the strongest (maximum received signal strength) control channel at any time. After determining the strongest received signal, the cellular phone receiver remains tuned to the particular forward control channel. The cell-site broadcasts the same signaling and control data on all forward control channels at the same time. In this way, the cell-site is able to signal all mobile subscribers within the cellular system and can be sure that any incoming call from a landline network will be routed to the called mobile subscriber.

9.2.1 A Basic Cellular System Connected to PSTN

A basic cellular radio network covers a number of geographical areas (cells) connected with landline or wireless telephone communication network deploying Public Switched Telephone Network (PSTN). Within the cell, cellular mobile subscribers can communicate with one another using the cellular network. The cellular network is defined by a set of transceivers located at the centres of each of the cells, and the locations of these radio-frequency transceivers are called *base stations*. A base station serves as an air interface as well as local central control for all mobile subscribers within that cell. Mobile phone equipment, either installed in vehicles or portables carried by users or handheld by subscribers, communicate directly with the nearest base stations.

The base stations, in turn, communicate directly with a Mobile Telephone Switching Office (MTSO). The MTSO controls channel assignment, call set-up, call processing, and call termination. The complete process includes allocating radio-frequency channels, signaling, switching, and supervision. Basically, the MTSO provides a centralised administration and maintenance point for the entire cellular network including interfaces with the public telephone network over voice trunks and data links. Figure 9.1 shows a basic cellular system connected to a landline network PSTN.

A standard common air interface is defined to establish full duplex communication between the mobile subscribers and the base station in a cellular system. For this purpose, two distinct types of channels are specified in each direction (uplink and downlink). Control channels carry data messages for call initiation and service requests, and are continuously monitored by mobile subscribers when they are not engaged in voice call. The signaling and control information is transferred from the base station to the mobile subscribers on a forward control channel. Forward control channels are also used to continuously broadcast all of the traffic requests for all the mobile subscribers in the system. The signaling and control information from the mobile subscribers to the base station is transferred on a reverse control channel. The control channels are often involved in initial setting up of a call and then moving it to an available voice channel. The voice transmission from the base station to mobile subscribers take place on forward voice channels, and the voice transmission from mobile subscribers to the base station use reverse voice channels. Supervisory and monitoring data messages are sent

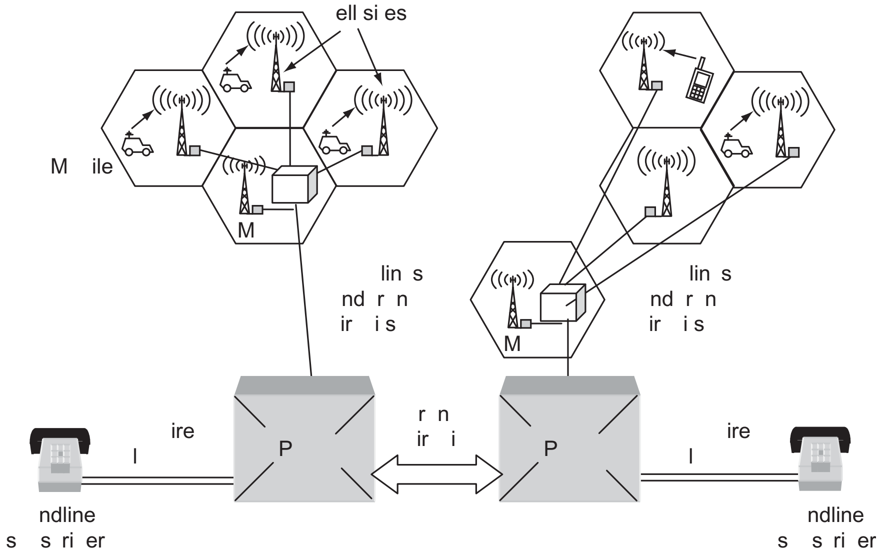

Fig. 9.1 *A basic cellular system connected to PSTN*

in a number of ways to facilitate handoff instructions and automatic channel changes for the mobile subscribers before and during a call.

Each cell-site has several radio transceivers (one per channel) called *Base Transceiver System* (BTS) or simple *base station*. The base stations are distributed over the geographical area of the desired system coverage. BTS includes a wideband RF power amplifier to provide the transmit power for all channels in a site or sector. Cell-site antennas installed on a high tower are part of the BTS. The cell-site's radio equipment is controlled by an on-site cell-site controller called Base Station Controller (BSC). It handles the air interface between cell-site radio equipment and mobile subscribers including allocation of traffic channels, monitoring and control of power levels, signaling tones/data, and so on.

Base stations also communicate directly with the MTSO over dedicated data control links. The MTSOs, also called Mobile Switching Centres (MSCs), route calls using coaxial cables, fiber optic links, or microwave links. Sometimes the BSC and MTSO are combined units. The MTSO contains various databases for storing the locations of local and roaming mobile subscribers, authorising calls, initiating hand-offs, and billing. Trunk circuits interconnect MTSOs with landline telephone exchange offices within the PSTN. This interconnection allows calls to be made between landline telephone subscribers and cellular mobile subscribers, as well as among mobile subscribers of different cellular service providers in the same operating area.

9.2.2 Main Parts of a Basic Cellular System

A basic cellular system consists of mainly three parts: Cell-Site Equipment (CSE), Mobile Telephone Switching Office (MTSO), and Mobile Subscriber Unit (MSU) as shown in Fig. 9.2.

There is an air interface between the MSU and CSE. The interconnectivity between the CSE and MTSO, MTSOs, and the MTSO and PSTN is through wirelines or dedicated point-to-point microwave links.

Cell-Site Equipment (CSE) A cell-site is a fixed base station used for wireless communication with a mobile subscriber on one side as well as signaling/data communication with the MTSO on the other side. It is usually located at the centre or the edge of the coverage region of a cell. A cell-site consists of a number of transreceivers, *Tx*/*Rx* antennas mounted on a tall tower, data links, and power plant. The radio transmitting

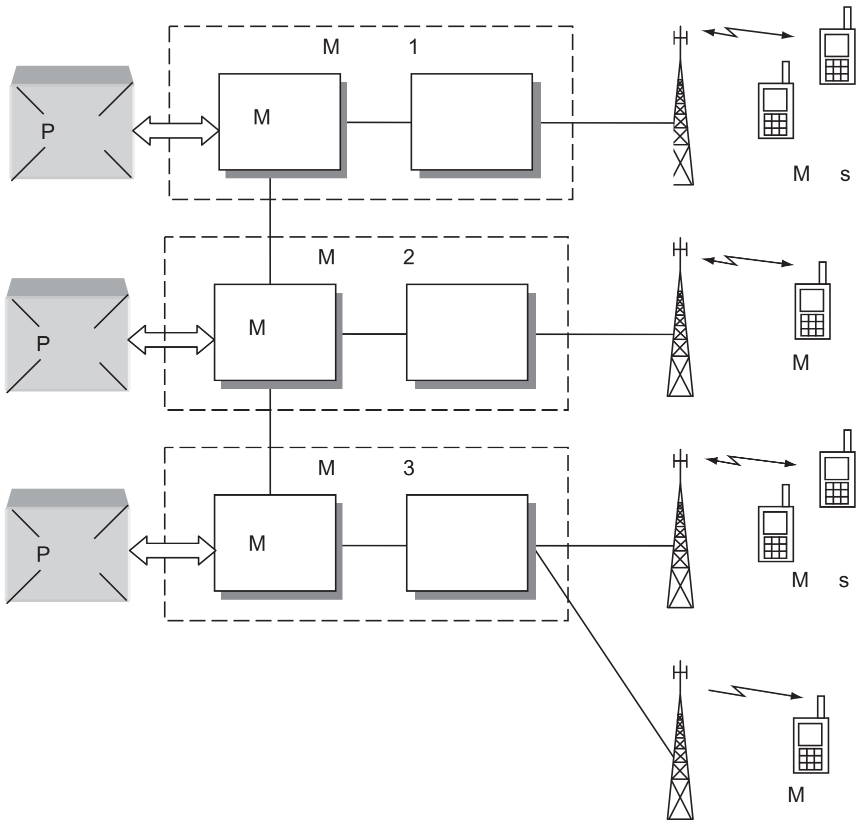

Fig. 9.2 *Parts of a basic cellular system*

equipment operates at considerably higher RF power than do the mobile equipments. Tx power is shared among all the channels that are used at the cell-site. Similarly, there are as many receivers for each control and voice channel in use at the cell-site, as well as additional receivers for monitoring the signal strength of mobile subscribers in adjacent cells.

Cell-site equipment basically comprises of two main parts—cell-site transceiver and cell-site controller. There may be adequate number of transceiver modules at the cell-site equipment in order to meet the subscriber capacity requirement within a cell. Data links are used to carry multiple-channel data from the cell-site to the MTSO. The transmission data rate on data links vary from 10 kbps to several Mbps. Many data-link channels can be multiplexed and passed through a wideband T-carrier (or E-carrier) wireline or a point-to-point microwave radio link operating at 850 MHz or higher frequency.

Radio transceivers are part of the cell-site equipment. The radio transceivers meant for voice channels can be either narrowband FM for analog systems or QAM/PSK modulation for digital systems with an effective audio-frequency band (approximately 300 Hz to 3000 Hz) comparable to a standard telephone circuit. The control channels use either FSK or PSK modulation scheme. The cell-site controller operates under the control of the central switching centre MSC or MTSO. The cell-site controller manages each of the radio channels at each cell-site, turns the radio transmitter and receiver on and off, transfers data onto the control and voice channels, monitors calls, and performs built-in diagnostic tests on the cell-site equipment.

The issues affecting the cellular system design in selection of cell-site antennas include antenna pattern, antenna gain, antenna tilting, and antenna height. The antenna radiation pattern can be omnidirectional, directional, or any other shape in both the horizontal and vertical planes. The antenna-radiation patterns are different as viewed in the cellular mobile operating environment from the antenna-radiation patterns as viewed in free space. Antenna gain compensates for the transmitted power. Antenna tilting can reduce the interference to the neighbouring cells and enhance the weak signal spots in the radio coverage of the cells. The height of the cell-site antenna can affect the area and shape of the coverage pattern in the cellular system.

Facts to Know!

The cell-site equipment is much more complex, bulky, and expensive than the individual mobile subscriber phones. Transmit and receive antennas may be separate antennas or combined ones at the cell-site. Generally, one Tx antenna and two Rx antennas are used at each cell-site or each sector. Two Rx antennas provide space diversity to counteract the effects of fading.

Mobile Telephone Switching Office (MTSO) It is the central coordinating element for all the cell-sites connected to it. It comprises of the switch and the processor. It also interfaces with the Public Switch Telephone Network (PSTN), controls call processing and handles billing activities. It uses voice trunks as well as data links between the cell-sites and the central processor. Microwave radio links or T-carriers (wirelines) carry both voice and data between the cell-site and the MTSO because the high-speed data link cannot be transmitted over the standard telephone trunks. The capacity of switching equipment in cellular systems is not based on the number of switch ports but on the capacity of the processor associated with the switches. The processor should be as large as possible. Also, it is important to consider when the switching equipment would reach the maximum capacity. It determines the service life of the switching equipment. More control modules can be added to increase the system capacity. Switching equipment can be linked to other switching equipments for better utilisation of hand-off.

The electronic switching centre located in the MTSO or MSC is a sort of digital telephone exchange that becomes the heart of a cellular telephone system. Electronic switches communicate with cell-site controllers using a data-link protocol, such as X.25, at a transmission rate of 9.6 kbps or higher. The electronic switching centre performs two essential functions:

- It controls switching between the public landline telephone network and the cell-site base stations for landline-to-mobile, mobile-to-landline, and mobile-to-mobile calls.
- It processes data received from the cell-site controllers concerning mobile subscriber status, diagnostic data, and bill-compiling information.

Mobile Subscriber Unit (MSU) Basically, a mobile subscriber unit comprises of a single antenna, transreceiver, and microprocessor-based control circuit. Because the cellular system is full duplex, the transmitter and receiver must operate simultaneously with a single antenna. A duplexer is used to separate the transmit and receive signals. The 45-MHz band separation between transmit and receive frequencies makes the operation relatively easy, and simplifies frequency synthesiser design.

For example, GSM mobile subscriber comprises of two parts––the mobile equipment (ME) and an electronic smart card called a subscriber identity module (SIM). The ME is the hardware used by the subscriber to access the cellular network. The SIM is a card, which plugs into the ME. This card identifies the MS subscriber and also provides other information regarding the service that the subscriber should receive.

Each mobile subscriber consists of a mobile antenna, a multiple-frequency radio transceiver, and a control/logic unit. The transceiver uses a frequency synthesiser to tune into any designated cellular system channel. The control unit houses all the user interfaces, including a built-in handset or earphone or external microphone/speaker arrangement. The logic unit interrupts subscriber actions and system commands while managing the operation of the transceiver including transmit power.

Facts to Know!

Vehicle-installed mobile and portable/handheld mobile phone equipments are essentially identical except that portable/handheld mobile subscribers have a lower output power, have a less efficient antenna, and operate exclusively on batteries. Vehicle-installed mobile phone equipment have relatively large output power, have omnidirectional gain antenna mounted on the rooftop of the vehicle, and can operate from vehicle batteries.

9.3 OPERATION OF A CELLULAR SYSTEM

Voice calls over cellular communication networks require two full-duplex radio-frequency channels simultaneously. Two types of channels are available between the mobile subscriber and the base station: control channels and traffic channels. *Control channels* are used to exchange information concerning initiating and maintaining calls and with establishing of a relationship between a mobile subscriber and the nearest base station. The control channel is also used for transferring control and diagnostic information between mobile subscribers and a central cellular switch through a cell-site. Traffic channels carry a voice or data connection between subscribers. The traffic channel is the actual voice channel where calling mobile subscribers communicate directly with other called mobile subscribers and landline telephone subscribers through the cell-site and MTSO. Base stations transmit on the forward control channel and forward voice channel and receive on the reverse control channel and reverse voice channel. Similarly, mobile subscribers transmit on the reverse control channel and reverse voice channel and receive on the forward control channel and forward voice channel.

Establishment of a voice call within a cellular communication system is similar to completing a telephone voice call using the landline PSTN. The use of a cellular system is fully automated and requires no action on the part of the mobile subscriber other than placing or answering a call. When a mobile subscriber is first turned on, it performs a series of start-up procedures and then samples the received signal strength on all control channels. The mobile subscriber automatically gets tuned to the control channel with the strongest received signal strength level and synchronises to the control data transmitted by the cell-site controller. The mobile subscriber interprets the data and continues monitoring the control channel(s). The mobile subscriber automatically rescans the control channels periodically to ensure that it is using the best control channel. From a subscriber's point of view, the operation can be divided into four parts and a hand-off procedure.

- Mobile-unit initialisation or registration
- Mobile-originated calls
- Network-originated calls
- Call termination
- Hand-off procedure

9.3.1 Mobile-Unit Initialisation or Registration

Immediately after the mobile subscriber is switched on, it first scans the group of forward control channels and selects the strongest one, which usually belongs to the nearest cell-site. It then continuously monitors that control channel until its received signal level drops below the pre-defined threshold received level. In case the signal strength of the control channel becomes weak, the mobile subscriber again begins scanning of the forward control channels in search of the strongest signal. This self-location scheme is subscriber-independent. After pre-determined time, this procedure is repeated to update the availability status of the forward control channel. Cells assigned with different frequency bands broadcast on different forward control channels repetitively. The mobile subscriber gets registered itself with the cell-site as being active and this process is repeated periodically. The MTSO can then track the location of the desired mobile subscriber by paging it on the forward control channel.

For any cellular system, the total number of allocated channels is divided into control channels and voice traffic channels. About 5% of the total number of channels available in the system are defined as control channels and standardised over the entire geographic area covered. The remaining 95% of the total number of channels are dedicated to voice and data traffic for the mobile subscribers. Since the control channels are standardised and are identical throughout different service areas within a large geographic service area for a particular cellular operator, every mobile subscriber phone scans the same set of control channels.

A very important aspect for successful operation of numerous system functions in the cellular system is that each and every mobile subscriber units must be registered at one of the MTSOs or MSCs. This is maintained for authentication and identity verification, access privileges, and also for billing purposes. Moreover, the cellular system needs to know whether the MSU is currently located in its own home service area or is visiting some other service area. This enables incoming calls meant for roaming mobile subscribers to be routed to an appropriate cell location and assures desirable support for outgoing calls.

Facts to Know!

When a cellphone user moves around within the same cell, the base station for that cell handles all the transmission.

CSEs periodically broadcast control signals to determine and test nearby MSUs. This is done by exchanging signals known as handshake signals between the CSE and the MSU. Each MSU listens for broadcast control signals transmitted by CSEs. Some of the information contained in the broadcast forward control signals includes cellular network identifier, timestamp, ID (identification) of the paging area, gateway MSC address, and other system parameters of the CSE. If the MSU listens to a broadcast forward control signal from the new CSE, it updates its information database. The MSU uses this information to locate the nearest CSE and establish an appropriate communication link with the cellular system through the nearest operational CSE as a gateway.

9.3.2 Mobile-Originated Calls

When a mobile subscriber originates a call, a call initiation request is sent on the reverse control channel. The mobile subscriber enters the called subscriber number on its mobile equipment and presses the send button. A request for service is sent on an available reverse control channel. With this request, the mobile subscriber transmits its own telephone number, electronic serial number of the mobile equipment, station class mark which indicates what the maximum transmitter power level is for the calling subscriber, and the called subscriber number (of another mobile subscriber or landline telephone subscriber). The nearest cell-site receives this complete data on the reverse control channel and sends a request to the MTSO for allocation of required resources to establish the voice communication link between the calling mobile subscriber and the desired called subscriber. The MTSO validates this call request. After authentication, the MTSO directs a cell-site to assign an available forward voice channel for the call. The MTSO also connects the called mobile subscriber or makes a connection to the called landline subscriber through the PSTN. The MTSO also instructs the cell-site and the calling mobile subscriber to shift to an unused forward and reverse voice channel pair to allow the conversation to begin. Figure 9.3 depicts the call processing using various parts of a cellular system.

Within a cellular communication system, the following types of mobile-originated calls can take place involving mobile cellular subscribers originating calls:

– Mobile (cellular)-to-landline (PSTN) call
– Mobile (cellular)-to-mobile (cellular) within the same cell
– Mobile (cellular)-to-mobile (cellular) operating in different cells

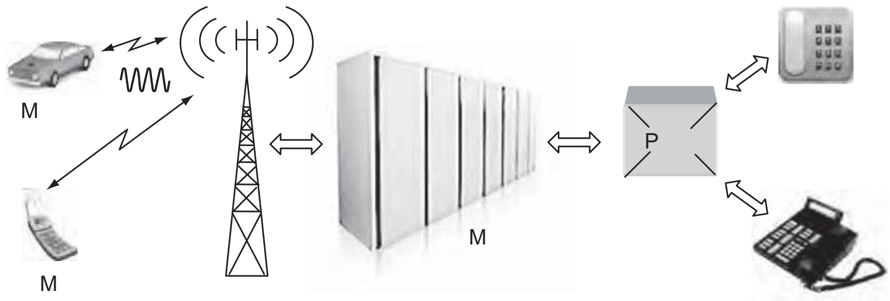

Fig. 9.3 *Call processing in a cellular system*

A general description for the sequence of events involved with connecting a call initiated by a mobile subscriber in a cellular system is briefly described here.

Mobile (Cellular)-to-Landline (PSTN) Call Procedures

Step 1. Calls from mobile subscribers to landline telephone subscribers can be initiated by entering the landline telephone number into the mobile unit's keypad. The mobile subscriber then presses a send key, which transmits the called landline telephone number as well as the mobile unit's identification number (MIN), ESN and Station Class Mark over a reverse control channel to the base station.

Step 2. The base station receives a call-initiation request along with the MIN, ESN, and Station Class Mark. If the calling mobile unit's ID number is valid, the cell-site controller routes the called landline telephone number over a wireline trunk circuit to the MTSO.

Step 3. The MTSO uses either standard call progress signals or the SS7 signaling protocol network to locate a switching path through the PSTN to the called landline telephone subscriber.

Step 4. Using the cell-site controller, the MTSO assigns the calling mobile subscriber an available traffic or voice channel and instructs the mobile subscriber to get tuned to that channel.

Step 5. After the cell-site controller receives verification that the mobile subscriber has tuned to the selected voice channel and it has been determined that the called landline telephone number is not busy, the mobile subscriber receives an audible call progress tone (ring-back) while the landline telephone caller receives a standard ringing tone.

Step 6. If a suitable switching path is available to the landline telephone number, the call is completed when the landline party answers the incoming call on its telephone.

Mobile (Cellular)-to-Mobile (Cellular) Call Procedures

Step 1. The originating mobile subscriber initiates the call in the same manner as it would do for a mobile-to-landline call.

Step 2. The cell-site controller receives the caller's identification number and the destination telephone number through a reverse control channel, which are then forwarded to the MTSO.

Step 3. The MTSO sends a page command to all cell-site controllers to locate the called mobile subscriber (which may be anywhere within or out of the service area).

Step 4. Once the called mobile subscriber is located, the destination cell-site controller sends a page request through a forward control channel to the called mobile subscriber to determine if it is on and not busy.

Step 5. After receiving a positive response to the page, the available free traffic channels are assigned to both the calling and called mobile subscribers.

Step 6. Call-progress tones are given to both the calling and called mobile subscribers (ring-back and ringtones respectively).

Step 7. When the MTSO receives a response that the called mobile subscriber has answered the incoming call, the call-progress tones are terminated, and the conversation begins.

Step 8. If a mobile subscriber wishes to initiate a call and all traffic channels are busy, the MTSO sends a directed retry command, instructing the calling mobile subscriber's unit to reattempt the call through a neighbouring cell.

Step 9. If the MTSO cannot allocate traffic channels through a neighbouring cell, it sends an intercept message to the calling mobile subscriber over the forward control channel. During the mobile-initiated call stage, if all the traffic channels assigned to the nearest cell-site are busy, then the mobile subscriber makes a preconfigured number of repeated attempts. After a certain number of failed attempts, a busy tone is returned to the calling mobile subscriber. This situation is termed as *call blocking*.

Step 10. If the called mobile subscriber is busy, the calling mobile subscriber receives a busy signal.

Step 11. If the called mobile number is invalid, the calling mobile subscriber receives a recorded message announcing that the call cannot be processed by the network.

9.3.3 Network-Originated Calls

When a telephone call is placed by a landline telephone subscriber to a mobile subscriber, the MTSO dispatches the request to all cell-sites in the cellular system, or it sends a paging message to certain cell-sites based on the called mobile subscriber number and search algorithm. Each cell-site transmits the page on its forward control channel. The called subscriber's mobile phone number is then broadcast as a paging message over all of the forward control channels throughout the cellular system. The mobile receives the paging message sent by the base station which it monitors, and responds by identifying itself over the reverse control channel. It also locks on to the assigned voice channel and initiates a subscriber alert tone.

The cell-site relays back the acknowledgment signal sent by the called mobile subscriber and informs the MTSO of the successful handshake. At this point, an alert message is transmitted to instruct the called mobile subscriber to ring, thereby instructing the mobile subscriber to answer the incoming call. Then, the MTSO instructs the cell-site to move the call to the available free forward and reverse voice channel pair. The step-by-step procedure given below shows the sequence of events involved for landline (PSTN)-to-mobile (cellular) call in a cellular telephone system. All of these events occur within a few seconds and are not noticeable by the subscriber.

Step 1. The landline telephone goes off hook to complete the wireline loop, receives a dial tone from PSTN, and then inputs the mobile subscriber's phone number.

Step 2. The mobile phone number is transferred from the PSTN switch to the cellular network switch (MTSO) that services the called mobile subscriber.

Step 3. The cellular network MTSO translates the received digits, and locates the cell-sites nearest the called mobile subscriber, which determines if the mobile subscriber is on and ready to receive the incoming call. It sends the requested mobile phone number to the cell-sites.

Step 4. The base station transmits the page containing mobile subscriber phone number on forward control channel.

Step 5. The called mobile subscriber receives the page signal and matches the received mobile subscriber phone number with its own mobile phone number, assuming that the called mobile subscriber is available.

Step 6. The called mobile subscriber acknowledges back the receipt of the mobile subscriber phone number and sends a positive page response including its ESN and Station Class Mark on the reverse control channel to the cell-site for forwarding it to the MTSO.

Step 7. The cell-site receives the mobile subscriber phone number, ESN, and Station Class Mark and passes the information to the MTSO.

Step 8. The MTSO verifies that the called mobile has a valid mobile subscriber phone number and ESN pair.

Step 9. The MTSO requests the cell-site controller to move the called mobile to the available pair of forward and reverse voice channels.

Step 10. The cell-site controller assigns an idle voice channel for the called mobile subscriber and the cell-site transmits the data message on the forward control channel for the called mobile subscriber to move to the specified voice channel.

Step 11. The called mobile subscriber receives the data messages on forward control channel to move to the specified voice channel and sends verification of designated voice channel to the cell-site.

Step 12. The cell-site controller sends an audible call progress tone to the called mobile subscriber, causing it to ring. The MTSO connects the called mobile subscriber with the calling landline phone on the PSTN. At the same time, a ring-back signal is sent back to the landline-calling telephone subscriber by PSTN.

Step 13. The called mobile subscriber answers back, the MTSO terminates the call-progress tones, and the two-way voice conversation begins on the forward voice channel and reverse voice channel between the calling telephone subscriber and the called mobile subscriber.

Once a call is in progress, the MTSO adjusts the transmitted power of the mobile subscriber and changes the channel of the mobile subscriber and cell-site in order to maintain call quality as the subscriber moves in and out of range of each cell-site. This is called hand-off procedure. Special control signaling is applied to the voice channels so that the cell-site may control the mobile subscriber while a call is in progress.

9.3.4 Call Termination

When either the calling subscriber (cellular mobile or landline) or the called subscriber (cellular mobile or landline) engaged in conversation terminates the call, the MTSO is informed and the traffic channels at the cell-site(s) are released. When the mobile subscriber terminates the call, a particular message signal is transmitted to the cell-site. The voice channel is released. The mobile subscriber resumes monitoring page messages through the strongest forward control channel. During a connection, if the base station cannot maintain the minimum required signal strength for a certain period of time because of interference or weak signal spots in certain areas, the voice channel assigned to the mobile subscriber is dropped and the MTSO is informed. This situation is termed as *call drop*, not call termination.

9.3.5 Hand-off Procedure

When the mobile subscriber moves out of the coverage area of its cell-site during the call, the received signal level becomes weak. The present cell-site requests a hand-off to MTSO. The MTSO switches the ongoing call to a new voice channel in a new cell-site without either interrupting the call or alerting the engaged mobile subscriber. The call continues as long as the conversation is on. The mobile subscriber does not notice the hand-off occurrences. Hand-off occurrence depends on the size of the cell, radio coverage boundary, received signal strength, fading, reflection and refraction of signals, and man-made noise. Assuming that the

> **Facts to Know!**
>
> Hand-off basically involves change of radio resources from one cell to radio resources in another adjacent cell. From a hand-off perspective, it is important that a free voice channel is available in a new cell whenever hand-off takes place, so that uninterrupted communication service is available at all times.

MSUs are uniformly distributed in each cell, the probability of a voice channel being available in a new cell depends on the number of channels per unit area. The number of channels per unit area increases if the number of channels allocated per cell is increased or if the area of each cell is decreased. But the radio resources and the number of assigned channels are limited.

The radio coverage area of the cell could be decreased for a given number of channels per cell. This leads to a smaller cell size and may be good for the availability of free channel perspectives. However, this would cause more frequent hand-offs, especially for MSUs with high mobility and vehicle speed. Hand-off can be initiated by the cell-site on its own or assisted by the mobile subscriber.

Cellular systems provide a service called *roaming*. This allows mobile subscribers to operate in service areas other than the one from which the service is subscribed. When a mobile subscriber enters another geographic area that is different from its home service area, it is registered as a roamer in the new service area. This is accomplished over the forward control channel, since each roaming mobile subscriber is stationed on a forward control channel at all times.

After a pre-defined time interval, the MTSO issues a broadcast command over each forward control channel in the cellular system, requesting all mobile subscribers, which are previously unregistered to report their identities such as mobile phone number and ESN over the reverse control channel. New unregistered mobile subscribers in the system periodically report back their subscriber information upon receiving the registration request. The MTSO uses the received data to request billing status from the home location register for each roaming mobile subscriber.

If a particular roaming mobile subscriber has roaming authorisation for billing purposes, the MTSO registers the mobile subscriber as a valid roamer. Once registered, roaming mobile subscribers are allowed to receive and place calls from that service area, and billing is routed automatically to the subscriber's home service provider.

EXAMPLE 9.1 Operation of a basic cellular system

Suppose there are two mobile subscribers in the nearby location. Draw a functional diagram showing the route signals if the cellphones are

(a) operating on the same MTSO

(b) operating on different MTSOs (one each on system A and system B), served by different service providers

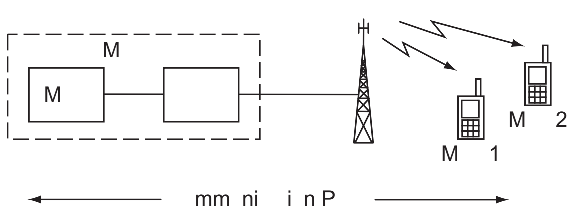

Fig. 9.4 *MSUs operating on the same MTSO*

Solution

(a) Each mobile subscriber communicates with the other mobile subscriber through the same serving cell-site, cell-site controller, and MTSO.

Figure 9.4 depicts a functional diagram showing the communication path between MSU1 and MSU2, both being served by the same MTSO.

The call initiated by MSU1 is routed to MSU2 via a communication link such as

MSU1→CSE→BSC→MSC→BSC→CSE→MSU2.

After the establishment of the link, two-way conversation between MSU1 and MSU2 takes place using this communication path.

(b) Each mobile subscriber communicates through its serving cell-site, cell-site controller, and MTSO respectively. Two $MTSO_s$ are interconnected through PSTN.

The call initiated by MSU1 is routed to MSU2 via a communication link such as

MSU1→CSE1→MTSO-A→PSTN→ MTSO-B→CSE2→MSU2.

Figure 9.5 depicts a functional diagram showing the communication path.

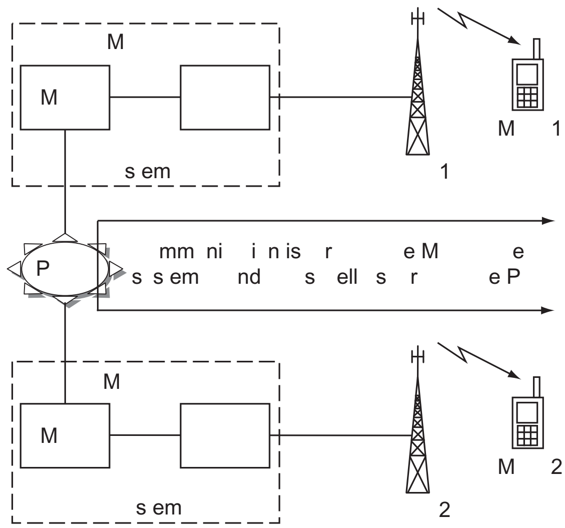

Fig. 9.5 MSUs operating on different MTSOs

9.4 PERFORMANCE CRITERIA

There are mainly five categories for specifying performance criteria in a cellular system:

- Voice quality
- Trunking and Grade of service
- Spectral efficiency
- Radio capacity
- Service quality and special features

9.4.1 Voice Quality

For any wireless communication system, the voice quality is based upon the criterion that some per cent of subscribers rate the system voice quality as good or excellent on the perceived quality scale. The average of the circuit merit level scores obtained from all the active subscribers in the system is called Mean Opinion Score (MOS). Table 9.1 depicts the meaning of different circuit merit levels and MOS.

Facts to Know!

Toll-grade voice quality refers to the quality of the international wired telephone system.

Table 9.1 *Circuit merit level and MOS*

Circuit merit level	*Mean Opinion Score (MOS)*	*Quality scale*	*Physical significance*
CM5	5	Excellent	Speech perfectly understandable
CM4	4	Good	Speech easily understandable
CM3	3	Fair	Speech understandable with slight effort
CM2	2	Poor	Speech understandable with considerable effort, frequent repetitions needed
CM1	1	Unsatisfactory	Speech not understandable

Usually, the telephone voice quality is around MOS $\geq$ 4. As the percentage of subscribers choosing the voice quality level CM4 and CM5 increases, the cost of building the infrastructure for the wireless communication system rises.

9.4.2 Trunking and Grade of Service

The grade of service is a benchmark used to define the desired performance of a particular cellular communication system by specifying a desired probability of a mobile subscriber obtaining channel access given a specific number of channels available in the system. Cellular communication systems rely on trunking to accommodate a large number of mobile subscribers in a limited radio spectrum. Trunking exploits the statistical behavior of mobile subscribers so that a fixed number of channels may accommodate unpredictably large mobile subscribers. The concept of trunking allows a large number of mobile subscribers to share the relatively small number of available channels in a cell by providing access to each mobile subscriber, on demand, from a pool of available channels.

Cellular communication systems are examples of trunked radio systems in which each mobile subscriber is allocated a channel on a per-call request basis. Upon termination of the call, the previously occupied channel is immediately returned to the pool of available channels. When a mobile subscriber requests service and in case all of the radio channels are already busy, the incoming subscriber call is blocked, or denied access to the system. In some communication systems, a queue may be used to hold the requesting mobile subscribers until a channel becomes available.

Facts to Know!

For example, the busy hours for cellular communication systems typically occur during rush hours between 10 a.m. to 11 a.m. in the morning and 4 p.m. to 5 p.m. in the evening on any working day.

The Grade Of Service (GOS) is a measure of the ability of a mobile subscriber to access a cellular system during the busiest hour. The busy hour is based upon the subscriber's demand for the service from the system at the busiest hour during a week, month, or year.

It is necessary to estimate the maximum required capacity in terms of available channels and to allocate the proper number of channels in order to meet the GOS. GOS is typically specified as the probability that a call is blocked, or the probability of a call experiencing a delay greater than the predefined queuing time. A call which cannot be completed at the time of call request made by a mobile subscriber is referred to as a *blocked call* or *lost call*. This may happen due to channel congestion or non-availability of a free channel. In other words, grade of service is a measure of channel congestion which is specified as the probability of a call being blocked, or the probability of a call being delayed beyond a specified time.

Facts to Know!

Erlang, the international, dimensionless unit of telephone traffic, is frequently used in capacity computations.

The trunking and queuing theory developed by Erlang is applied to design trunked cellular systems that can handle a desired subscriber capacity at a specified grade of service. One *Erlang* represents the amount of *traffic intensity* carried by a channel that is completely occupied (that is, one call-hour per hour or one call-minute per minute). For example, a radio channel that is occupied for 30 minutes during an hour carries 0.5 Erlang of traffic (30 minutes divided by one hour). *Traffic intensity* is the average channel occupancy measured in Erlangs, and is a measure of channel time utilisation. This is a dimensionless quantity and may be used to measure the time utilisation of single or multiple channels, and denoted by A.

The offered amount of traffic load presented to a system is typically characterised by the following two important random parameters:

λ = the average rate of calls (that is, connection requests) attempted per unit time or average number of MSs requesting the service or average call-arrival rate

H = the average holding time per successful call or average length of time the MSs requiring the service

The traffic intensity offered by each mobile subscriber is equal to the call request rate (the average number of call requests per unit time), λ per seconds, multiplied by the holding time (average duration of a typical call), H seconds. That is, each mobile subscriber generates an average traffic intensity of A_{av} Erlangs given by the expression

$$= \lambda \times \tag{9.1}$$

A_{av} is also referred to as the *offered traffic load* and can be interpreted in several ways. It is the normalised version of λ, that is, A_{av} equals the average number of calls arriving during the average holding period. A servicing channel that is kept busy for an hour is quantitatively defined as one Erlang. For example, in a cell with 100 MSs on an average, if 30 call requests are generated during an hour, with an average holding time H = 360 seconds, the average request rate or average call-arrival rate is

$$\lambda = A_{av} / H = (30 \text{ call requests}) / (1 \text{ hour or } 3600 \text{ seconds}) = 0.008 \text{ requests/s}$$

Hence, the offered traffic load in Erlang is

$$A_{av} = \lambda \times H = (0.008 \text{ requests/s}) \times 360 \text{ s} = 3 \text{ Erlangs}$$

As another example, if the calling rate averages 20 calls per minute and the average holding time is 3 minutes, then $A_{av} = 60$. A cell switch with a capacity of 120 channels would be about half utilised at any given time. A cell switch with a capacity of 50 channels would clearly be inadequate. A capacity of 60 would meet the average demand but because of fluctuation around the average rate λ, this capacity would at times be inadequate.

EXAMPLE 9.2 | Average traffic intensity

Calculate the average traffic intensity for the traffic data given in Table 9.2, depicting the pattern of activity in a cell of 10-channel capacity over a period of one hour.

Solution

Step 1. *To determine the average rate of calls,* λ

Total number of calls in all 10 channels = 97 per hour (from the given data)

Therefore, rate of calls per minute, $\lambda = 97/60 = 1.62$

Table 9.2 | *Traffic data in a cell with 10-channel capacity*

Channel	*Number of calls per hour*	*Total occupied channel time (minutes)*	*Approximate occupied channel time (hour)*
1	17	51	0.85
2	16	47	0.78
3	14	43	0.72
4	12	39	0.65
5	11	34	0.57
6	10	28	0.47
7	7	22	0.37
8	5	15	0.25
9	3	9	0.15
10	2	6	0.10

Step 2. *To determine the average holding time per call, H*

Total system occupied time = 294 minutes (from the given data)

Therefore, average holding time per call, H = 294/97 = 3.03 minutes

Step 3. *To calculate the average traffic intensity, A_{av}*

The average traffic intensity, $A_{av} = \lambda \times \text{H}$

Hence, the average traffic intensity, $A_{av} = 1.62 \times 3.03 = 4.9$ Erlangs

Comments on the results Thus, on an average 4.9 channels out of available 10 channels in the given system are busy. So, A_{av} can be interpreted as the average number of calls in progress. However, it is true only in the non-blocking system because the parameter λ has been defined as the rate of calls attempted, not carried traffic.

It is generally unreasonable to decide the number of channels in a system for the highest peak of traffic anticipated. Typically, a blocking system has adequate number of channels so as to deal with some upper limit of traffic intensity. However, the common practice is to fix the number of channels in the system to meet the average call rate encountered during a busy hour. Usually, the busy hour is the 60-minute period during daytime when the traffic is expected to be maximum in the long run. The average of the busy hour traffic is usually taken on the 10 or 30 busiest days of a calendar year to determine the acceptable number of channels in the system. It may be noted that these measurements are typical of the carried traffic rather than offered traffic. This can only be used to estimate the true load. For a system containing number of mobile subscribers and an unspecified number of channels, the total offered traffic intensity A_t, is given as

$$A_t = \text{Z} \times A_{av} \tag{9.2}$$

If the numbers of channels are specified, say N, in a trunked cellular mobile system then assuming the traffic is equally distributed among the channels, the traffic intensity per channel, A_c, is given as

$$A_c = A_t / N \tag{9.3}$$

EXAMPLE 9.3 Traffic intensity per subscriber

In a trunked cellular mobile system, each mobile subscriber averages two calls per hour at an average call duration of three minutes. Determine the traffic intensity per mobile subscriber.

Solution

Average number of calls per hour per user, $\lambda = 2$ (given)

Average duration of call, $H = 3$ minutes (given)

Or, average duration of call, $H = 3/60$ hours $= 0.05$ hours

Therefore, the average traffic intensity per user, A_{av} is given by the expression

$A_{av} = \lambda \times H$

Or, $A_{av} = 2 \times 0.05 = 0.1$ Erlangs

When the offered traffic exceeds the maximum capacity of the system in terms of the allocated number of channels, the carried traffic becomes limited due to the limited number of channels. The maximum possible carried traffic is the total number of channels in Erlangs. For example, the AMPS cellular system is designed for a GOS of 2% blocking. This implies that the channel allocations for cell-sites are designed in such a way so that 2 out of 100 calls requested by mobile subscribers will be blocked due to channel congestion during the busiest hour.

EXAMPLE 9.4 Traffic intensity for a cellular system

A cellular system is allocated a total bandwidth of 30 MHz and each simplex channel of 25 kHz. The hexagonal cell configuration is given in Fig. 9.6.

(a) If each channel is shared among 8 mobile subscribers, how many calls can be simultaneously processed by each cell if only 10 channels per cell are reserved for signalling and control purpose?

(b) If each mobile subscriber keeps a traffic channel busy for an average of 5% time and an average of 60 call requests per hour are generated, compute the offered traffic load.

(c) During a busy hour, the number of calls per hour for each of the 12 cells of a cellular cluster is 2220, 1900, 4000, 1100, 1000, 1200, 1800, 2100, 2000, 1580, 1800 and 900 respectively. Assuming that 75% of the mobile subscribers in this cluster are using the system during this period and that one call is made per subscriber, find the number of mobile subscribers per cluster in the system. Assuming the average holding time of 60 seconds, what is the total offered traffic load of the system in Erlangs?

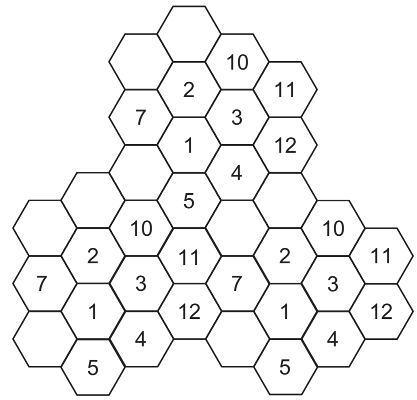

Fig. 9.6 | *A cellular configuration*

Solution

Total allocated bandwidth = 30 MHz (given)
Bandwidth of one simplex channel = 25 kHz (given)

Step 1. *To determine total number of duplex channels*

Total number of simplex channels = 30 MHz / 25 kHz = 1200 channels
Total number of duplex channels = 1200/ 2 = 600 channels

(a) *To determine number of calls per cell*

Number of control channels/cell = 10 channels/cell (given)
Number of cells/cluster = 12 (from given figure)

Step 2. *To determine number of traffic channels/cell*

Total number of control channels = $10 \times 12 = 120$ channels
Total number of traffic channels = $600 - 120 = 480$ channels
Number of traffic channels/cell = $480/12 = 40$ channels/cell

Step 3. *To determine number of calls per cell*

Number of users/channel = 8 (given)
Number of maximum calls/channel = 8
Hence, Total number of calls per cell = $8 \times 40 = 320$ calls/cell

(b) *To compute the offered traffic load*

Step 4. *To determine λ and H*

Average number of call requests per hour = 60 (given)
Or, Average call request rate, $\lambda = 60/3600 = 1/60$ requests/second
Call-holding time, $H = 5\%$ (given)
Or, Call holding time, $H = 0.05 \times 3600 = 180$ seconds

Step 5. *To compute the offered traffic load*

The average offered traffic load, $A_{av} = \lambda \times H$
Or, average offered traffic load, $A_{av} = (1/60) \times 180$
Hence, average offered traffic load, $A_{av} = 3$ Erlangs

(c) *To find the total offered traffic load of the system*

Step 6. *To find the total number of calls made in the cluster*

Number of cells in a cluster = 12 (from given figure)

Total number of calls made in the cluster = 2220 + 1900 + 4000 + 1100 + 1000 + 1200 + 1800 + 2100 + 2000 + 1580 + 1800 + 900 (given data)
Hence, Total number of calls made in the cluster = 21,580

Step 7. *To find the number of mobile subscribers busy in the cluster*

Percentage of busy mobile subscribers = 75% (given data)
Number of busy mobile subscribers = 0.75 × 21,580
Hence, Number of busy mobile subscribers = 16 185 users

Step 8. *To determine average call-request rate, λ*

Average number of calls in each cell = 21,580 / 12
Or, Average number of calls in each cell = 1800
Average call-request rate, $\lambda = 1800/3600$
Hence, Average call request rate, $\lambda = 0.5$ calls/second

Step 9. *To compute total offered traffic load*

Average call-holding time, $H = 60$ seconds (given)
The average total offered traffic load, $A_{av} = \lambda \times H$
Or, the average total offered traffic load, $A_{av} = 0.5 \times 60$
Hence, the average total offered traffic load, $A_{av} = 30$ Erlangs

Practically, there are two types of trunked cellular systems which are commonly used. The first type of trunked cellular system offers no queuing for call requests, popularly known as *Erlang B system*. This means that for every mobile subscriber making a service request, it is assumed that there is no set-up time (the time required to allocate a trunked radio channel) to a requesting mobile subscriber. The mobile subscriber is given immediate access to a channel if it is available. If no channels are available, the requesting mobile subscriber is blocked without access to the system and is free to try again later. In other words, when all channels are busy, an arriving call is not serviced. This type of trunking is called *blocked calls cleared* or *blocked calls lost*. It assumes that calls arrive as determined by a Poisson distribution. Furthermore, the following assumptions are made:

– There are an infinite number of mobile subscribers in the system.
– There are memoryless arrivals of requests, implying that all subscribers, including blocked subscribers, may request a channel at any time with no previous record of call requests.
– The probability of a mobile subscriber occupying a channel is exponentially distributed, so that longer duration calls are less likely to occur as described by an exponential distribution.
– There are a finite number of channels available in the trunked pool.

This leads to the *Erlang B* formula (also known as the blocked-calls cleared formula). The Erlang B formula provides a conservative estimate of the GOS for an infinite number of mobile subscribers, as the results of the finite number of mobile subscribers always predict a smaller probability of blocking. The capacity or total offered traffic intensity A_t of a trunked cellular system where blocked calls are cleared is tabulated for various values of blocking probability or grade of service (GOS) and the given number of channels. Table 9.3 is an extract from Erlang B table for ready reference here.

Given the offered load and number of channels, the grade of service can be determined from the table. More often, it is needed to determine the amount of traffic that can be handled by a given number of channels or capacity to produce a given grade of service or to determine the capacity required to handle a given amount of traffic at a given grade of service.

Table 9.3 *Average offered traffic load of an Erlang B system*

Number of channels	*Offered load in Erlangs for given GOS or blocking probability of*				
	0.1 %	*0.2%*	*0.5%*	*1.0%*	*2.0%*
1	0.001	0.002	0.005	0.01	0.02
4	0.439	0.535	0.701	0.869	1.09
5	0.762	0.900	1.130	1.360	1.66
10	3.090	3.430	3.960	4.460	5.08
20	9.410	10.100	11.100	12.100	13.19
24	12.240	13.010	14.21	15.27	16.64
40	24.5	25.7	27.3	29.0	31.0
70	49.2	51.0	53.7	56.1	59.13
100	75.2	77.4	80.9	84.1	87.97

EXAMPLE 9.5 Number of channels required

In a cellular system, the average calls per hour in one cell is 3000 and an average calling (call holding) time is 1.76 minutes. If the blocking probability is 2%, find the offered traffic load and the maximum number of channels needed in the system. If the average number of calls per hour in one cell increases to 28,000, find the maximum number of channels required in the system.

Solution

Step 1. *To determine λ and H*

The average calls per hour in one cell = 3000 (given)

Average call-request rate, $\lambda = 3000/3600$

Hence, average call-request rate, $\lambda = 0.833$ calls/second

Average call-holding time, $H = 1.76$ minutes (given)

Or, Average call-holding time, $H = 1.76 \times 60 = 105.6$ seconds

Step 2. *To compute the offered traffic load*

The offered traffic load, $A_{av} = \lambda \times \mathrm{H}$

Or, the offered traffic load, $A_{av} = 0.833 \times 105.6$

Hence, the offered traffic load, $A_{av} = 87.97$ Erlangs

Step 3. *To find maximum number of channels, N*

Blocking probability = 2% or 0.02 (given)

Corresponding to 0.02 blocking probability, using Erlang B Table, we get

Maximum number of channels, $N = 100$ channels/cell

Step 4. *To find new λ for increased average calls per hour in one cell*

The average calls per hour in one cell = 28,000 (given)

New average call-request rate, $\lambda = 28{,}000/3600$

Hence, average call-request rate, $\lambda = 7.78$ calls/second

Step 5. *To compute the new offered traffic load*

Average call holding time, $H = 105.6$ second (As in Step 1)

The offered traffic load, $A_{av} = \lambda \times H$

Or, the offered traffic load, $A_{av} = 7.78 \times 105.6$

Hence, the offered traffic load, $A_{av} = 821.6$ Erlangs

Step 6. *To find the maximum number of channels required for increased* λ

Blocking probability = 2% or 0.02 (given)

Corresponding to 0.02 blocking probability and $A_{av} = 821.6$ Erlangs, using Erlang B Table, we get

Maximum number of channels, $N = 820$ channels/cell

Example 9.6 Number of calls per hour per cell

If there are 50 channels in a cell to handle all the calls and the average call-holding time is 100 s per call, how many calls can be handled in this cell with a blocking probability of 2 percent?

Solution

Number of channels in a cell, $N = 50$ (given)

Blocking probability = 2% or 0.02 (given)

Step 1. *To find offered traffic load,* A_{av}

Corresponding to blocking probability of 0.02 and $N = 50$, the offered load, A_{av} from Erlang B Table is found to be 40.3 Erlangs.

Therefore, offered traffic load, $A_{av} = 40.3$ Erlangs

Step 2. *To determine average call-holding time, H*

The average call-holding time per call = 100 s (given)

Therefore, average call-holding time, $H = 100/3600$ hours = 0.0278 hours

Step 3. *Number of calls handled in the cell*

We know, the offered traffic load, $A_{av} = \lambda \times H$

Or, Average call request rate, $\lambda = A_{av} / H$

Or, average call request rate, $\lambda = 40.3/0.0278$

Hence, number of calls handled in the cell = 1450 calls per hour

Two important points can be deduced from the Erlang B Table.

– A large-capacity cellular system is more efficient than a smaller capacity cellular system for a given grade of service. For example, consider two cells, each having a capacity of 10 channels. The joint capacity of the two cells taken together is 20 channels and they can handle a combined offered traffic intensity of 6.86 Erlangs at a grade of service of 0.002. However, a single cell of 20-channel capacity can handle a traffic intensity of 10.07 Erlangs at a grade of service of 0.002. This shows that a large capacity cellular system is more efficient in terms of offered traffic intensity than a smaller capacity cellular system for a given grade of service.

– A large capacity cellular system is more susceptible to reduction of the grade of service. For example, consider a cell with a capacity of 10 channels, giving a grade of service of 0.002 for an offered load of 3.43 Erlangs. A 30% increase in traffic (that is, for an offered load of 4.46 Erlangs) reduces the grade of service to 0.01. However, for a cell capacity of 70 channels, only a 10% increase in traffic reduces the grade of service from 0.002 to 0.001.

EXAMPLE 9.7 Number of users in an Erlang B system

In a blocked-calls-cleared (Erlang B) system having 10 trunked channels, each mobile subscriber generates 0.1 Erlangs of traffic. Compute the number of mobile subscribers that can be supported for 0.5% blocking probability. Repeat the above for 100 trunked channels.

Solution

Average traffic intensity per user, $A_t = 0.1$ Erlangs (given)

Blocking probability $= 0.5\%$ or 0.005 (given)

Number of trunked channels, $N = 10$ (given)

Step 1. *To find offered traffic load, A_{av}*

Corresponding to blocking probability of 0.005 and $N = 10$, the offered load, A_{av} from the Erlang B Table is found to be 3.96.

Therefore, offered traffic load, $A_{av} = 3.96$ Erlangs

Step 2. *To compute total number of mobile users*

The total number of mobile users that can be supported in the given system can be computed by dividing the offered traffic load, A_{av} with the given average traffic intensity per user, A_t. That is,

Total number of mobile users $= A_{av} / A_t$

Hence, total number of mobile users $= 3.96 / 0.1 = 39$ users

Step 3. *To find offered traffic load, A_{av} for increased N*

Number of trunked channels, $N = 100$ (given)

Corresponding to blocking probability of 0.005 and $N = 100$, the offered load, A_{av} from Erlang B Table is found to be 80.9 Erlangs.

Total number of mobile users $= A_{av} / A_t$

Hence, total number of mobile users $= 80.9 / 0.1 = 809$ users

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EXAMPLE 9.8 | Number of users in an Erlang B system

If a cellular system has 20 cells to cover the specified service area, with each cell having 40 channels, how many mobile subscribers can the cellular system support if a blocking probability of 2% is required? Assume that each mobile subscriber makes an average of three calls/hour and each call duration is an average of three minutes (use Erlang B values).

Solution

Number of channels in each cell = 40 channels (given)

Blocking probability $= 2\%$ or 0.02 (given)

Step 1. *To find offered traffic load, A_{av}*

Corresponding to blocking probability of 0.02 and $N = 40$, the offered load, A_{av} from Erlang B Table is found to be 31.0.

Therefore, offered traffic load, $A_{av} = 31$ Erlangs

Step 2. *To determine number of users per cell*

Number of calls per hour per user = 3 (given)

Let there be   number of users requesting calls per hour. Then, average number of calls per hour, λ = number of users × number of calls per user

Therefore, $\lambda = \quad \times 3$

Average call-holding time, $H = 3$ minutes or 3/60 hours (given)

Using the expression, $A_{av} = \lambda \times H$, we get

$31 = \lambda \times 3/60$

Substituting $\lambda = \quad \times 3$, we get

$31 = \quad \times 3 \times 3/60$

Or, $\quad = 31 \times 60 / 9 = 206.67$ users per cell

Step 3. *To determine number of mobile subscribers in the system*

Total number of cells in the system = 20 (given)

Therefore, number of mobile subscribers in the system = 206.67 × 20

Hence, number of mobile subscribers in the system = 4133 users

The overall system capacity in a trunked cellular system depends largely upon the allocation of channels. Trunking efficiency is a measure of the number of mobile subscribers that can be offered a particular GOS or blocking probability with a particular configuration of fixed channels. The way in which channels are grouped can substantially alter the number of mobile subscribers handled by a trunked system. Trunking efficiency decreases for less number of trunked channels in an Erlang B system. Since cell sectoring breaks up the available trunked channel pool into several smaller pools of channels to be used in each sector, overall trunking efficiency decreases.

Facts to Know!

System capacity is the number of subscribers accommodated by the system, and is a useful measure of efficient spectrum utilisation. The system capacity is proportional to the number of subscribers in a unit area provided the geographical distribution of traffic is uniform within the service area.

Example 9.9 Trunking efficiency

In an Erlang B system having 24 trunked channels, show that 10 channels trunked together can support 60% more traffic at a specified GOS than two five-channel trunks grouped individually.

Solution

Number of trunked channels = 24 (given)

Let the given system have a GOS of 0.01 or 1% blocking probability.

Step 1. *To find offered traffic load, A_{av} for $N = 10$*

Corresponding to blocking probability of 0.01 and $N = 10$, the offered load, A_{av} from Erlang B Table is found to be 4.46.

Therefore, offered traffic load, $A_{av} = 4.46$ Erlangs

Step 2. *To find offered traffic load, A_{av} for $N = 5$*

Corresponding to blocking probability of 0.01 and $N = 5$, the offered load, A_{av} from Erlang B Table is found to be 1.36.

Therefore, offered traffic load, $A_{av} = 1.36$ Erlangs

Step 3. *To find offered traffic load, A_{av} for two groups of 5 trunked channels each*

Offered traffic load, A_{av} for one group = 1.36 Erlangs (From Step 2)

Offered traffic load, A_{av} for two groups = 2 × 1.36 Erlangs

Or, offered traffic load, A_{av} for two groups = 2.72 Erlangs

Step 4. *Comparison of offered traffic load for $N = 10$ and $2 \times (N = 5)$*

Offered traffic load for ($N = 10$) = 4.46 Erlangs (As in Step 1)

Offered traffic load for $2 \times (N = 5)$ = 2.72 Erlangs (As in Step 3)

Clearly, 10 channels trunked together can support more traffic at a specific GOS (say 0.01) than two five-channel trunks individually do.

Step 5. *The extent of traffic supported by $N = 10$ system*

The extent of more traffic supported by a 10-channel trunked system as compared to two five-channel trunked systems = 2.72 / 4.46 = 0.6 or 60%

EXAMPLE 9.10 Trunking efficiency in sectored systems

Consider a seven-cell reuse cellular system having a total of 395 traffic channels. In this system, an average call lasts for three minutes, and the probability of blocking is to be no more than 1%. Let every mobile subscriber make one call per hour, on average. Assume that blocked calls are cleared so the call blocking is described by the Erlang B distribution. Determine the following:

(a) The average number of calls made by a mobile subscriber per hour if the system is configured as an omnidirectional system.

(b) The average number of calls made by a mobile subscriber per hour if the system is configured as a 3-sectored antenna configuration. Show that the decrease in trunking efficiency from that of an omnidirectional configuration is 24%.

(c) The average number of calls made by a mobile subscriber per hour if the system is configured as a 6-sectored antenna configuration. Show that the decrease in trunking efficiency from that of an omnidirectional configuration is 44%.

Comment on the above results.

Solution

Total number of allocated traffic channels = 395 (given)
Number of cells per cluster = 7 (given)
Probability of blocking (GOS) < 1% (given)

Step 1. *To find number of channels per cell*

Number of channels per cell, N = Total channels / Cells per cluster
Therefore, number of channels per cell, $N = 395 / 7 = 57$ traffic channels

Step 2. *To find call holding time, H*

Average duration of a call by a user = 3 minutes (given)
Therefore, call-holding time, $H = 3/60$ hours = 0.05 hours

(a) *Omnidirectional configuration cellular system*

Step 3. *To find offered traffic load, A_{av}*

Corresponding to blocking probability of 0.01 and $N = 57$, the offered load, A_{av} from Erlang B Table is found to be 44.2.

Therefore, offered traffic load, $A_{av} = 44.2$ Erlangs

Step 4. *To determine average number of calls, λ*

Average number of calls made in a cell, $\lambda = A_{av} / H$
Or, Average number of calls, $\lambda = 44.2$ Erlangs / 0.05 hours
Hence, Average number of calls, $\lambda = 884$ calls per hour

(b) *3-sector configuration cellular system*

Step 5. *To find number of channels per sector*

Number of sectors per cell = 3 (given)
Thus, Number of channels per sector = 57/3 = 19 traffic channels

Step 6. *To find offered traffic load, A_{av} per sector*

Corresponding to blocking probability of 0.01 and $N = 19$, the offered load, A_{av} per sector from Erlang B Table is found to be 11.2.

Therefore, offered traffic load per sector, $A_{av} = 11.2$ Erlangs

Step 7. *To determine average number of calls made in a sector*

Average number of calls made in a sector = A_{av} / H
Average number of calls made in a sector = 11.2 Erlangs / 0.05 hours
Hence, Average number of calls in a sector = 224 calls per hour

Step 8. *To determine average number of calls made in a cell, λ*

Average number of calls in a cell, $\lambda = 3 \times 224$

Hence, Average number of calls in a cell, $\lambda = 672$ calls per hour

Step 9. *To determine decrease in trunking efficiency*

Decrease in trunking efficiency = (884 – 672) / 884

Hence, decrease in trunking efficiency = 0.24 or 24%

Comments on the result *Thus, cell sectoring decreases the trunking efficiency while improving the signal-to-interference ratio for each user in the system.*

(c) *6-sector configuration cellular system*

Step 10. *To find number of channels per sector*

Number of sectors per cell = 6 (given)

Thus, Number of channels per sector = 57/6 = 9.5 traffic channels

Step 11. *To find offered traffic load, A_{av} per sector*

Corresponding to blocking probability of 0.01 and $N = 9.5$, the offered load, A_{av} per sector from Erlang B Table is found to be 4.1.

Therefore, offered traffic load per sector, $A_{av} = 4.1$ Erlangs

Step 12. *To determine average number of calls made in a sector*

Average number of calls made in a sector = A_{av} / H

Average number of calls made in a sector = 4.1 Erlangs / 0.05 hours

Hence, Average number of calls in a sector = 82 calls per hour

Step 13. *To determine average number of calls made in a cell, λ*

Average number of calls in a cell, $\lambda = 6 \times 82$

Hence, Average number of calls in a cell, $\lambda = 492$ calls per hour

Step 14. *To determine decrease in trunking efficiency*

Decrease in trunking efficiency = (884 – 492) / 884

Hence, decrease in trunking efficiency = 0.44 or 44%

Comments on the result Thus, the degradation in trunking efficiency in a 6-sector configuration amounts to 44% when compared to the unsectored (omnidirectional) case. It is also a fact that using 6-sectors per cell reduces the number of cochannel interferers from six to only one in the first tier that results into significant improvement in signal-to-interference ratio for each user for a seven-cell system. This enables a four-cell reuse system. Of course, using six sectors per cell reduces the trunking efficiency and increases the number of necessary hand-offs even more.

The second type of trunked cellular system is called *blocked calls delayed* and its measure of grade of service is defined as the probability that a call is blocked after waiting a specific length of time, say *t*, in the queue. In this system, a queue is provided to hold the calls requested which are blocked. If a channel is not available immediately, the call request may be delayed until a channel becomes available. To find the grade of service, it is first necessary to find the probability that a call is initially denied access to the system. The probability of a call not having immediate access to a channel is determined by the *Erlang C formula*. If no channels are immediately available, the call is delayed. The probability that the delayed call is forced to wait for more than *t* seconds is given by the probability that a call is delayed, multiplied by the conditional probability that the delay is greater than *t* seconds.

EXAMPLE 9.11 Number of users in an Erlang C system

A cellular system using a four-cell reuse pattern has hexagonal cells, each having a radius of 1.4 km. A total number of 60 channels are used within the entire system. If the average traffic load per user is 0.029 Erlangs, and $\lambda = 1$ call/hour, compute the approximate number of users per square kilometre that will be supported for an Erlang C system which has a 5% blocking probability of a delayed call. (Use the traffic intensity as 9.0 Erlangs for 5% blocking probability with 15 channels per cell from Erlang C Table.)

Solution

Step 1. *To find area covered by hexagonal cell, A_{cell}.*

Radius of the cell, $R = 1.4$ km (given)

Area covered by hexagonal cell, $A_{cell} = 2.6 \times R^2$

Therefore, area covered by a cell, $A_{cell} = 2.6 \times (1.4)^2 = 5.1$ km^2

Step 2. *To find number of channels per cell, n_{cell}.*

Number of cells per cluster, $K = 4$ (given)

Total number of channels, $n_{total} = 60$ (given)

Number of channels per cell, $n_{cell} = 60 / 4 = 15$ channels

Step 3. *To find total number of users supported in a cell*

The average traffic load per user = 0.029 Erlangs (given)

Specified traffic intensity = 9.0 Erlangs (given)

Blocking probability = 5% or 0.05 (given)

Total number of users supported by the system in a cell is given by total traffic intensity / average traffic load per user, that is,

Total number of users supported in a cell = 9.0 Erlangs / 0.029 Erlangs

Total number of users supported in a cell = 310 users

Step 4. *To compute total number of users per km^2 area*

Number of users per km^2 area = Number of users in a cell / cell area

Therefore, number of users per km^2 area = 310 users / 5.1 km^2

Hence, number of users per km^2 area = 60 users/ km^2 (approx.)

9.4.3 Spectral Efficiency

The spectrum of the wireless channel is interference limited because of the severe channel impairments. In a mobile communication system, the spectrum utilisation can be enhanced by using various techniques such as choice of multiple access method, channel assignment, bandwidth reduction, and data compression to reduce the transmission rate. The overall spectral efficiency of a mobile communication system can be estimated based on the spectral efficiency component, η_1 (expressed as channels/MHz/km^2), due to the system parameters such as cell area in km^2, frequency reuse factor, channel spacing in kHz, and type of modulation scheme used, as well as the spectral efficiency component, η_2 (expressed as Erlangs/MHz/km^2), due to the multiple access method used. Spectral efficiency may be defined as channels/MHz/km^2 or Erlangs/MHz/km^2, where Erlang is a measure of the traffic load, in order to capture the frequency reuse in the service coverage area of the system. That is, spectral efficiency defined in terms of channels/MHz/km^2 can be expressed as

$$\eta_1 = (\text{Total number of channels}) / (\text{system bandwidth})(\text{total coverage area})$$

And, spectral efficiency defined in terms of Erlangs/MHz/km^2 can be expressed as

$$\eta_2 = (\text{Total traffic load}) / (\text{system bandwidth})(\text{total coverage area})$$

Thus, overall system spectral efficiency for a mobile communication system, η, can then be expressed as $\eta = \eta_1 \times \eta_2$.

9.4.4 Spectral Efficiency of FDMA Systems

In a FDMA/FDD cellular systems such as AMPS, a single user occupies two simplex channels (which are frequency duplexed) during a call. Let B_t be the total frequency spectrum bandwidth available for transmission in one direction (uplink or downlink), B_c be the channel bandwidth, and B_g be the guard band used at each edge of the allocated spectrum (as shown in Fig. 9.7). Then the total number of channels, N_t, that can be simultaneously supported in the system is given by

Fig. 9.7 Channel spacing and guard band in FDMA uplink or downlink

$$_t = (\;_t - 2 \times \;)\,/ \tag{9.4}$$

Or,
$$B_t = N_t \times B_c + 2 \times B_g \tag{9.5}$$

In a cellular system, all available radio channels are allocated to the cells in a single cluster, and the same radio channels are reused in each and every cluster. Then N_t represents the total number of available channels per cluster. Let N_c be the number of allocated control channels and N_d be the number of user traffic/data in the system (or one cluster). The maximum number of simultaneous users in each cluster is equal to the number of available user data channels, N_d. Then the total number of available channels is

$$N_t = N_c + N_d \tag{9.6}$$

Therefore,
$$B_t = (N_c + N_d) \times B_c + 2 \times B_g$$

Or,
$$B_t = N_c \times B_c + N_d \times B_c + 2 \times B_g$$

Or,
$$B_t \;\; > N_d \times B_c \tag{9.7}$$

where $N_d \times B_c$ is the total bandwidth available for user data transmission.

The spectral efficiency of FDMA, in general, is defined as the ratio of total bandwidth available for user data transmission and allocated system bandwidth in a cluster, that is,

$$\eta_f = (N_d \times B_c) / B_t < 1 \tag{9.8}$$

If K is the cluster size (the number of cells in a cluster) then the number of user data channel per cell, $N_{d/cell}$, is given by

$$N_{d/cell} = N_d / K \tag{9.9}$$

where N_d is the number of data channels per cluster.

If A_{cell} is the geographical area covered by a cell in km^2, then the overall spectral efficiency in terms of channels/MHz/km^2 is expressed as

$$\eta_1 = \text{(Number of channels per cell)} / \text{(system bandwidth)(cell area)}$$

Or,
$$\boldsymbol{\eta}_1 = \;_{/e}\, / (\;_t \times \;_e) \tag{9.10}$$

EXAMPLE 9.12 | Spectral efficiency of FDMA system in channels/MHz/km²

In an analog FDMA cellular system, the allocated system bandwidth is 12.5 MHz, the channel spacing is 30 kHz, and the guard spacing at each edge of the allocated system bandwidth is 10 kHz. The number of channels allocated for control signaling is 21. Determine the following:

(a) The total number of available channels per cluster

(b) The number of channels available for user data transmission per cluster

(c) The number of channels available for user data transmission per cell if the cluster size or frequency reuse factor is 7

(d) The overall system spectral efficiency in units of channels/MHz/km², assuming the cell area as 6 km²

Solution

Allocated system bandwidth, $B_t = 12.5$ MHz or 12,500 kHz (given)

Channel spacing, $B_c = 30$ kHz (given)

Guard spacing, $B_g = 10$ kHz (given)

(a) *To determine total number of available channels per cluster, N_t*

The total number of available channels per cluster, N_t, is given by the expression

$$N_t = (B_t - 2 \times B_g) / B_c$$

Or, $N_t = (12500 - 2 \times 10) / 30 = 416$ channels

(b) *To determine the number of user data channels per cluster, N_d*

Number of control channels, $N_c = 21$ control channels (given)

So, the number of channels available for user data transmission per cluster, N_d, is given by the expression

$$N_d = N_t \quad N_c$$

Or, $N_d = 416 - 21 = 395$ data channels per cluster

(c) *To determine the number of user data channels per cell, $N_{d/cell}$*

Number of cells in a cluster, $K = 7$ (given)

So, the number of channels available for user data transmission per cell, $N_{d/cell}$, is given by the expression

$$N_{d/cell} = N_d / K = 395 / 7 \approx 56 \text{ data channels per cell}$$

(d) *To determine overall system spectral efficiency, η_1*

Area of the cell, $A_{cell} = 6$ km² (given)

The overall spectral efficiency of the system in terms of channels/MHz/km² is given by the expression

$$\eta_1 = N_{d/cell} / (B_t \times A_{cell})$$

Hence, $\eta_1 = 56 / (12.5 \times 6) = 0.747$ channels/MHz/km²

In any multiple access system, there is a finite probability that some of the traffic access is blocked. Let η_{tr} be the trunking efficiency in each cell, which is a function of the blocking probability and the total number of available user data channels per cell, $N_{d/cell}$. Hence, the total traffic carried in a cluster, in Erlang, is $\eta_{tr} \times N_d$, where N_d is the number of data channels per cluster. Therefore, spectral efficiency defined in terms of Erlangs/MHz/km² can be expressed as

$$\eta_2 = (\eta_t \times \quad _{/e}) / (\quad _t \times \quad _e) \quad \text{Erlangs/MHz/km}^2 \tag{9.11}$$

EXAMPLE 9.13 | Spectral efficiency of FDMA System in Erlangs/MHz/km²

In an analog FDMA cellular system configured with a cluster size of 7, the allocated system bandwidth is 12.5 MHz, the channel spacing is 30 kHz, and the guard spacing at each edge of the allocated system bandwidth is 10 kHz. The number of channels allocated for control signaling is 21. The cell area is 6 km² and the area of the entire cellular system is 3024 km². The average number of calls per user during a busy hour is 1.5 and the average channel holding time of a call is 180 seconds. If the trunk efficiency is 0.95, determine the following:

(a) The number of cells in the system
(b) The number of calls per hour per cell
(c) The average number of users per hour per cell
(d) The system spectral efficiency in units of Erlangs/MHz/km²

Solution

(a) *To determine the number of cells in the system, N_{cells}*

The area of the entire cellular system, $A_{cellular} = 3024$ km^2 (given)

The area of a cell, $A_{cell} = 6$ km^2 (given)

The number of cells in the system, $N_{cells} = A_{cellular} / A_{cell}$

Hence, $N_{cells} = 3024 / 6 = 504$ cells

(b) *To determine the number of calls per hour per cel, $N_{call/hr/cell}$*

Step 1. *To find number of data channels per cluster, $N_{d.}$*

Allocated system bandwidth, $B_t = 12.5$ MHz or 12,500 kHz (given)

The channel spacing, $B_c = 30$ kHz (given)

The guard spacing, $B_g = 10$ kHz (given)

Number of control channels, $N_c = 21$ control channels (given)

Number of data channels per cluster, $N_d = (B_t - 2 \times B_g) / B_c \quad N_c$

Therefore, $N_d = (12500 - 2 \times 10) / 30 - 21$

$= 395$ channels per cluster

Step 2. *To find number of data channels per cell, $N_{d/cell}$*

Number of cells in a cluster, $K = 7$ (given)

Number of data channels per cell, $N_{d/cell} = N_d / K$

Therefore, $N_{d/cell} = 395 / 7 \approx 56$ channels per cell

Step 3. *To determine the number of calls per hour per cell, $N_{call/hr/cell}$*

Average channel holding time of a call, $H = 180$ seconds or 1/20 hours (given)

The number of calls per hour, $N_{call/hr} = 1/H = 20$

The trunk efficiency, $\eta_{tr} = 0.95$ (given)

The number of calls per hour per cell, $N_{call/hr/cell} = N_{d/cell} \times \eta_{tr} \times N_{call/hr}$

Hence, $N_{call/hr/cell} = 56 \times 0.95 \times 20 = 1064$ calls/hour/cell

(c) *To determine the average number of users per hour per cell*

The average number of calls per user per hour, $N_{call/user/hr} = 1.5$ (given)

The average number of users per hour per cell, $N_{users} = N_{call/hr/cell} / N_{call/user/hr}$

Hence, $N_{users} = 1064 / 1.5 \approx 709$ users/hour/cell

(d) *To determine the spectral efficiency, η*

The spectral efficiency in terms of channels/MHz/km^2, η_1, is given by

$$\eta_1 = N_{d/cell} / (B_t \times A_{cell})$$

$$\eta_1 = 56 \text{ channels} / (12.5 \text{ MHz} \times 6 \text{ km}^2) = 0.747 \text{ channels/MHz/km}^2$$

Hence, the system spectral efficiency in units of Erlangs/MHz/km^2, $\eta = \eta_{tr} \times \eta_1$

Hence, $\eta = 0.95 \times 0.747 = 0.710$ Erlangs/MHz/km^2

9.4.5 Spectral Efficiency of TDMA Systems

In TDMA-based cellular systems, the total available frequency spectrum is divided into a number of frequency sub-bands, with each frequency sub-band operating as a TDMA system being partitioned into contiguous time slots. Each time slot is long enough to transmit one information unit.

An individual user only uses the allocated time slot of a frequency sub-band, and can only transmit data at a rate governed by the allocated frequency sub-band. The channel time in each sub-band is divided into time slots, numbered 1, 2, , N_s. In this way, any one of the numbered time slots, say slot 1, can be used by any user.

The TDMA format is illustrated in Fig. 9.8.

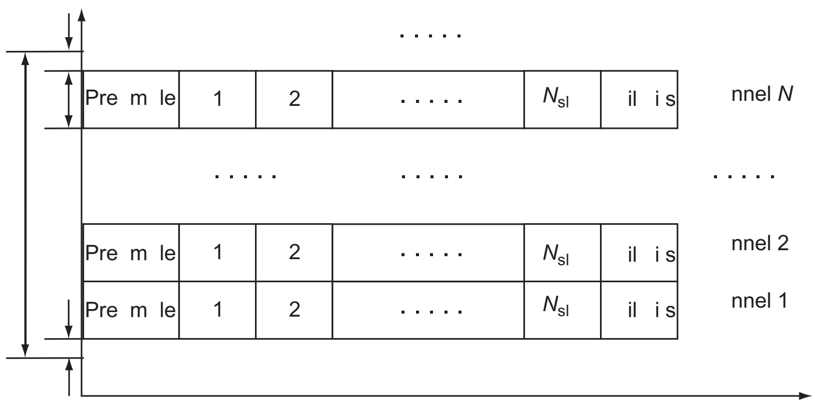

Fig. 9.8 *TDMA format*

Let B_t be the total frequency spectrum bandwidth available for transmission in one direction (uplink or downlink), B_c be the channel bandwidth, and B_g be the guard band used at each edge of the allocated spectrum. Then the total number of sub-bands, N_u that can be supported in the system is given by

$$= (\ _t - 2 \times \quad) / \tag{9.12}$$

Or,
$$N_u \times B_c = B_t - (2 \times B_g) \tag{9.13}$$

where $N_u \times B_c$ represents the actual usable bandwidth for information transmission. Thus, the sub-band efficiency, $\eta_{t(subband)}$, can be expressed as the ratio of the available information transmission bandwidth to the system bandwidth, that is,

$$\eta_{t(subband)} = (N_u \times B_c) / B_t \tag{9.14}$$

Since $(N_u \times B_c) \leq B_t$, then $\eta_{t(subband)} \leq 1$.

TDMA operates on a frame-by-frame basis. A TDMA frame comprises of a group of uniform time slots plus preamble in the beginning of the frame and tail bits at the end of the frame. In addition to carrying the information data, a time slot also includes other fields, such as tail bits, sync (synchronisation) bits, guard bits, etc. A time slot can be used by one subscriber to transmit or receive during one use of the transmission channel. The spectral efficiency per frame of a TDMA system, $\eta_{t(frame)}$, is defined as the product of time slot efficiency, $\eta_{t(slot)}$, and the information data symbols efficiency in a time slot, $\eta_{t(data)}$, that is,

$$\eta_{t(frame)} = \eta_{t(slot)} \times \eta_{t(data)} \tag{9.15}$$

Let T_f be the time duration of a TDMA frame in any frequency subband, T_p be the time duration for the preamble in a frame, and T_t be the time duration for the tail bits in a frame. Then the time slot efficiency, $\eta_{t(slot)}$, is given by the expression

$$\eta_{t(slot)} = (T_f \quad T_p \quad T_t) / T_f \tag{9.16}$$

Let L_s be the total number of symbols in each time slot, and L_d is the number of information data symbols in each time slot (a time slot also includes other fields, such as trail bits, sync (synchronisation) bits, guard bits, etc., in addition to information data field). Then the information data symbols efficiency, $\eta_{t(data)}$, is given by the expression

$$\eta_{t(data)} = L_d / L_s \tag{9.17}$$

Therefore, the spectral efficiency per frame of a TDMA system, $\eta_{t(frame)}$, is given by

$$\eta_{t(\quad e)} = \eta_{t(\quad t)} \times \eta_{t(\quad t\ )} = (\quad - \quad_p - \quad_t) / \quad \times (\quad / \quad) \tag{9.18}$$

The first term on the right-hand side takes into account the overhead at the frame level (due to frame preamble/tail bits), and the second term takes into account overhead at the time-slot level (due to tail bits, sync bits, and guard bits). The overhead corresponds to the control signaling data needed to coordinate the multiple access. In fact, $\eta_{t(frame)}$ signifies the spectral efficiency of TDMA system in a frequency sub-band. The spectral efficiency of TDMA system is given by

$$\eta_{t(sys)} = \eta_{t(frame)} \times \eta_{t(subband)} \tag{9.19}$$

Or,
$$\eta_{t(sys)} = (T_f \quad T_p \quad T_t) / T_f \times (L_d / L_s) \times (N_u \times B_c) / B_t \tag{9.20}$$

The overall spectral efficiency of a TDMA system is also a function of digital modulation scheme used in the system and the frequency reuse factor. It is defined in terms of bps/Hz/cell, and can be expressed as

$$\eta_t = \eta_{t(\quad)} \times \eta_{t(\quad)} / K \quad \text{bps/Hz/cell} \tag{9.21}$$

where $\eta_{t(mod)}$ is the modulation efficiency, defined as the maximum transmission rate in bits per second that the modulation can accommodate over one hertz channel bandwidth. K is the frequency reuse factor, defined as the number of cells in a cluster in which the entire allocated frequency spectrum is available for use.

EXAMPLE 9.14 | Spectral efficiency of TDMA cellular system

Consider a TDMA/FDD cellular system that uses an uplink or downlink spectrum of 25 MHz. The system bandwidth is divided into radio channels of 30 kHz, and uses guard spacing of 20 kHz. The frame duration is 40 ms (assuming preamble and tail bit duration as 0), consisting of 6 time slots. A single radio channel supports 3 full-rate speech channels, each using 2 time slots in a frame. Each time slot consists of 324 bits, among which 260 bits are for information data and the remaining 64 bits are overhead for access control. The speech codec rate is 7.95 kbps, which corresponds to a gross rate of 13.0 kbps with channel encoding. If the frequency reuse factor is 7, determine

(a) the number of simultaneous users that can be accommodated in each cell, N_u
(b) the spectral efficiency per frame of a TDMA system, $\eta_{t(frame)}$
(c) the spectral efficiency of the TDMA system, $\eta_{t(sys)}$.
(d) the overall spectral efficiency, η_t, in terms of bps/Hz/cell

Solution

***(a)** To determine the number of simultaneous users in each cell, N_u*

Allocated system bandwidth, $B_t = 25$ MHz or 25,000 kHz (given)

The channel bandwidth, $B_c = 30$ kHz (given)

The guard spacing, $B_g = 20$ kHz (given)

We know that $N_u = (B_t - 2 \times B_g) / B_c$

Hence, $N_u = (25{,}000 - 2 \times 20) / 30 = 832$ users/cluster

***(b)** To determine the spectral efficiency per frame of a TDMA system, $\eta_{t(frame)}$*

We know that $\eta_{t(frame)} = (T_f \quad T_p \quad T_t) / T_f \times (L_d / L_s)$

Time duration of a TDMA frame, $T_f = 40$ ms (given)

Time duration for the preamble in a frame, $T_p = 0$ ms (given)

Time duration for the trailer in a frame, $T_t = 0$ ms (given)

Number of data symbols in a time slot, $L_d = 260$ bits (given)
Total number of symbols in a time slot, $L_s = 324$ bits (given)
Hence, $\eta_{t(frame)} = (40 - 0 - 0) / 40 \times (260 / 324) = 0.8$

***(c)** To determine the spectral efficiency of the TDMA system, $\eta_{t(sys)}$*

We know that $\eta_{t(sys)} = \eta_{t(frame)} \times (N_u \times B_c) / B_t$
Hence, $\eta_{t(sys)} = 0.8 \times (832 \times 30) / 25{,}000 = 0.798$ or ≈ 0.8

***(d)** To determine the overall spectral efficiency, η_t, in terms of bps/Hz/cell*

***Step 1.** To find modulation efficiency, $\eta_{t(mod)}$*

Speech codec rate, $R_s = 7.95$ kbps (given)
Channel bandwidth, $B_c = 30$ kHz (given)
Modulation efficiency, $\eta_{t(mod)}$ is given by

$$\eta_{t(mod)} = R_s / B_c = 7.95 \text{ kbps} / 30 \text{ kHz} = 0.265$$

***Step 2.** To determine overall spectral efficiency, η_t*

Frequency reuse factor, $K = 7$ (given)
We know that $\eta_t = \eta_{t(sys)} \times \eta_{t(mod)} / K$ bps/Hz/cell
Hence, $\eta_t = (0.8 \times 0.265) / 7 \approx 0.303$ bps/Hz/cell

9.4.6 Radio Capacity and *C/I*

There is a tradeoff between various system parameters such as the radio capacity, voice quality, and dropped call rate. All these parameters are based on a common signal quality parameter *C/I*. The radio capacity in a cell, J is expressed as the number of available channels per cell. In a cellular system, if B_t is the total allocated spectrum bandwidth, B_c is the channel bandwidth, and K is the cluster size, then radio capacity, J can be expressed as

$$= (\ \ _t / \ \ ) \times (1 / K) \tag{9.22}$$

where B_t / B_c is the total number of available voice channels.

Or, $$J \times K = (B_t / B_c) \tag{9.23}$$

Considering the case of a fully developed hexagonal cellular system, having six equidistant cochannel interfering cells in the first tier, the voice quality, usually expressed as *C/I* ratio, is given by (assuming a mobile radio environment, $\gamma = 4$).

$$\frac{C}{I} \approx \frac{1}{6}.(q)^4 \tag{9.24}$$

where q is the cochannel interference reduction factor and is related with the cluster size K as given by the expression:

$$q = (3K) \tag{9.25}$$

Substituting Eq. (9.25) into Eq. (9.24), we get

$$\frac{C}{I} \approx \frac{1}{6}.\left(\sqrt{3K}\right)^4$$

Or, $$\frac{C}{I} \approx \frac{1}{6}.(3K)^2$$

Or, $$\frac{C}{I} \approx \frac{3}{2}.(K)^2$$

Or,
$$K = \sqrt{\frac{2}{3} \cdot \frac{C}{I}} \tag{9.26}$$

Substituting Eq. (9.26) into Eq. (9.23), we get

$$J\sqrt{\frac{2}{3} \cdot \frac{C}{I}} = \left(\frac{B_t}{B_c}\right)$$

Or,
$$\frac{C}{I} = \frac{3}{2}\left(\frac{B_t}{B_c}\right)^2 \left(\frac{1}{J}\right)^2 \tag{9.27}$$

The C/I ratio given by Eq. (9.27) is the required value for designing a system, which specifies the achievable voice quality. The following two facts can be concluded from this derivation.

- When the specified C/I is reduced (that is, lower voice quality is acceptable), the radio capacity increases.
- When the measured C/I is less than the specified C/I, both poor voice quality and dropped calls can occur.

When the radio capacity is based on the frequency reuse, the cochannel interference level is high, the size of the cells is small, and coverage is not an issue. Then, the call-dropped rate totally depends on interference. Using Eq. (9.23),

$$J = \left(\frac{B_t}{B_c}\right)\left[1 \Big/ \sqrt{\frac{2}{3} \cdot \frac{C}{I}(\min)} \right] \tag{9.28}$$

It implies that the maximum radio capacity occurs when $(C/I)_{min}$ and B_c are minimised. In order to provide the same voice quality, $(C/I)_{min}$ may be lower in a digital system when compared to an analog system. Typically, the minimum required C/I is 18 dB for narrowband analog FM cellular systems and about 12 dB for narrowband digital cellular systems, although exact values are determined by subjective listening tests in actual operating propagation conditions. Each digital cellular standard has a different $(C/I)_{min}$, and in order to compare different systems, an equivalent C/I must be used. Lower $(C/I)_{min}$ values imply more capacity. If B_t and J are kept constant, then it is clear that B_c and $(C/I)_{min}$ are related by the expression

$$(C/I)_{eq} = (C/I)_{min}\,(B_c / B_c')^2 \tag{9.29}$$

where B_c is the bandwidth of a particular system, $(C/I)_{min}$ is the acceptable value for the same system, B_c' is the channel bandwidth for a different system, and $(C/I)_{eq}$ is the minimum acceptable C/I value for the different system when compared to the $(C/I)_{min}$ for a particular system. It is implied that $(C/I)_{min}$ and B_c are inversely related.

When expressed in dB,

$$(\quad)_{eq} \text{ in dB} = (\quad)_{\;i} \text{ in dB} + 20\log(B_c / B_c') \tag{9.30}$$

If the channel bandwidth is halved, $(C/I)_{min}$ value has to be increased four times in order to maintain the same voice quality for a constant number of users per radio channel.

EXAMPLE 9.15 | Radio capacity in analog cellular system

Compare the performance of the following four different analog cellular systems in terms of C/I. Which system offers the maximum capacity? Assume path propagation constant, $\gamma = 4$ in each case.

System I: $B_c = 30$ kHz, $(C/I)_{min} = 18$ dB
System II: $B_c = 25$ kHz, $(C/I)_{min} = 14$ dB
System III: $B_c = 12.5$ KHz, $(C/I)_{min} = 12$ dB
System IV: $B_c = 6.25$ KHz, $(C/I)_{min} = 9$ dB

Solution Let channel bandwidth, $B_c' = 6.25$ kHz for each system, and $(C/I)_{eq}$ is the minimum acceptable C/I value for each of given systems with changed value of channel bandwidth which is required to be computed when compared to the $(C/I)_{min}$ for a particular system.

Step 1. *To determine $(C/I)_{eq}$ for System I*

The channel bandwidth, $B_c = 30$ kHz (given)

$(C/I)_{min} = 18$ dB (given)

Then $(C/I)_{eq}$ corresponding to $B_c' = 6.25$ kHz can be computed as

$$(C/I)_{eq} \text{ in dB} = (C/I)_{min} \text{ in dB} + 20 \log(B_c / B_c')$$

Or, $$(C/I)_{eq} = 18 + 20 \log(30/6.25) = 31.62 \text{ dB}$$

Step 2. *To determine $(C/I)_{eq}$ for System II*

The channel bandwidth, $B_c = 25$ kHz (given)

$(C/I)_{min} = 14$ dB (given)

Then $(C/I)_{eq}$ corresponding to $B_c' = 6.25$ kHz can be computed as

$$(C/I)_{eq} \text{ in dB} = (C/I)_{min} \text{ in dB} + 20 \log(B_c / B_c')$$

$$(C/I)_{eq} = 14 + 20 \log(25/6.25) = 26 \text{ dB}$$

Step 3. *To determine $(C/I)_{eq}$ for System III*

The channel bandwidth, $B_c = 12.5$ kHz (given)

$(C/I)_{min} = 12$ dB (given)

Then $(C/I)_{eq}$ corresponding to $B_c' = 6.25$ kHz can be computed as

$$(C/I)_{eq} \text{ in dB} = (C/I)_{min} \text{ in dB} + 20 \log(B_c / B_c')$$

$$(C/I)_{eq} = 12 + 20 \log(12.5/6.25) = 18 \text{ dB}$$

Step 4. *To determine $(C/I)_{eq}$ for System IV*

The channel bandwidth, $B_c = 6.25$ kHz (given)

$(C/I)_{min} = 9$ dB (given)

Then $(C/I)_{eq}$ corresponding to $B_c' = 6.25$ kHz can be computed as

$$(C/I)_{eq} \text{ in dB} = (C/I)_{min} \text{ in dB} + 20 \log(B_c / B_c)$$

$$(C/I)_{eq} = 9 + 20 \log(6.25/6.25) = 9 \text{ dB}$$

Maximum radio capacity occurs when the required C/I and B_c are minimum. Based on comparison, the smallest value of $(C/I)_{eq}$ should be selected for maximum capacity. Hence, the system IV offers the best capacity.

The capacity of FDMA and TDMA cellular systems is limited by available bandwidth. FDMA and TDMA cellular systems having the equivalent radio parameters have the same radio capacity and consequently the same spectrum efficiency. For example, an FDMA system with three channels, each having a bandwidth of 10 kHz and a transmission rate of 10 kbps, and a TDMA system having three time slots in a given channel bandwidth of 30 kHz, and a transmission rate of 30 kbps have the same radio capacity. However, the required peak power for TDMA is 10 log (k) higher than FDMA, where k is the number of time slots in a TDMA system of equal bandwidth. In practice, TDMA systems improve radio capacity by a factor of three to six times as compared to analog cellular radio systems by virtue of its ability to employ powerful error control and speech-coding techniques which result into better link performance in high-interference environments. By employing speech activity and adaptive channel-allocation methods, some TDMA systems are able to utilise each radio channel in a much better way.

EXAMPLE 9.16 Capacity of a 2G GSM digital cellular system

Compute the capacity of a 2G GSM digital cellular system, assuming the allocated spectrum for uplink and downlink as 12.5 MHz each.

Solution

The allocated bandwidth, $B_t = 12.5$ MHz (given)
The carrier channel bandwidth, $B_c = 200$ kHz (GSM Standard)
Number of users per channel, $N_s = 8$ (GSM Standard)

Step 1. *To find total number of available carrier channels, N*

Total number of available channels, $N = B_t / B_c$

Therefore, $N = 12.5$ MHz / 200 kHz = 62.5 channels

Step 2. *To find total number of users, N_U*

Number of users per channel, $N_s = 8$ (GSM Standard)

We know that total number of users, $N_U = N \times N_s$

Therefore, $N_U = 62.5 \times 8 = 500$ users

Step 3. *To compute the capacity of the system*

Frequency reuse factor, $K = 4$ (commonly used in GSM system)

Total number of users are available in a cluster. Assuming uniform traffic distribution, the number of users in each cell would be the same.

Hence, the system capacity per cell = 500/4 = 125 users per cell

Step 4. *Alternate method to compute the capacity of the system*

Alternately, user capacity per cell, M, is given by the expression

$$M = (B_t / B_c) \times N_s \times (1 / K)$$

Or, $M = (12.5 \text{ MHz} / 200 \text{ kHz}) \times 8 \times (1/4) = 125$ users per cell

Table 9.4 compares key parameters of analog FM based AMPS cellular system to other digital TDMA-based cellular systems leading to relative estimated capacity gains.

Table 9.4 *Comparison of analog AMPS with digital TDMA cellular systems*

Key parameter	*AMPS*	*USDC*	*GSM*	*PDC*
RF bandwidth (MHz)	25	25	25	25
No. of voice channels	833	2500	1000	3000
Cluster sizes	7	7 or 4	4 or 3	7 or 4
No. of channels/site	119	357 or 625	250 or 333	429 or 750
Traffic (Erlangs/km^2)	11.9	41 or 74.8	27.7 or 40	50 or 90.8
Capacity gain	1.0	3.5 or 6.3	2.3 or 3.4	4.2 or 7.6

Thus, it is seen that various cellular standards such as the US digital cellular (USDC), GSM, and Pacific Digital Cellular (PDC) use digital TDMA for high capacity as compared to AMPS using FDMA.

The capacity of CDMA systems is interference limited. Therefore, any reduction in the interference will cause a linear increase in the capacity of CDMA. In other words, the link performance for each mobile user increases as the number of users decreases. There are various other means of increasing CDMA capacity in addition to those of inherently provided by reuse of the available frequency spectrum in each cell and effective mobile transmitter power control. These techniques include use of multi-sectorised antennas in each cell that results in spatial isolation of mobile subscribers, and operating in a discontinuous transmission mode (DTX) which exploits the intermittent nature of speech.

9.4.7 Dropped-Call Rate

The dropped call is defined as the drop of an on-going call after the call is established but before it is properly terminated. If there is a possibility of a call drop due to the poor signal of assigned voice channel, this is considered a dropped call. This situation can occur when the mobile subscriber establishes a call using a strong control channel but changes to a weak voice channel. One reason could be due to the frequency-selective fading phenomenon involving different control channels and voice channels. In the mobile radio environment, the dropped calls can also occur due to rapid signal loss in the shadow region. This clearly means a dropped call can happen only after the successful establishment of the call using control channel.

Calls can also be dropped if a mobile phone equipment at the other end of the call loses battery power and stops transmitting abruptly. Transmission problems or a faulty transceiver at the base station are also the common cause of dropped calls. Cochannel and adjacent channel interference can also be responsible for dropped calls in a wireless communication network. Neighbour cells having adjacent channel frequencies can interfere with each other, deteriorating the quality of service and producing dropped calls.

If there is a possibility of a call drop due to non-availability of voice channel, this is considered as a blocked call, not a dropped call. The perception of dropped call rate by mobile subscribers can be higher due to many other reasons such as improper functioning of mobile equipment, very high speed of the mobile, worst (highly noisy) operating environment, etc. The dropped call rate and the specified voice quality (usually measured as C/I ratio) are inversely proportional. If the desired voice quality is compromised, dropped call rate can be set very low.

The dropped-call rate, measured in terms of dropped-call probability, is one of the most important quality of service indexes for monitoring the performance of cellular networks. The cellular service providers apply many optimisation procedures on several service aspects for the reduction of dropped-call rate. As an example, traffic load balancing is maintained in different cells, the service coverage area is maximised, the cellular network usage is optimised; interference and traffic congestion conditions are minimised. The specified cochannel and adjacent channel interference levels should be maintained in each cell during a busy hour, that is, worst interference case.

Hand-off is considered one of the main probable reasons of call dropping. The signaling of the hand-off and the mobile-assisted hand-off algorithm also impact the dropped-call rate. The response time for a handoff request has to be shorter in order to reduce the dropped-call rate. The dropped-call rate and call-blocking probabilities are also affected by subscriber mobility, different patterns for movements of mobile subscribers. It is explicitly assumed that an appropriate frequency planning has been implemented. Network equipment outages and mobile equipment failure are also ignored. These assumptions lead to consider that calls are dropped mainly due to propagation conditions, irregular user behavior, and the failure of the hand-off procedure. That is, the connection of an active mobile subscriber changing cell during the call several times is terminated (cell drop) due to the lack of radio resources in the new cell.

> **Facts to Know!**
>
> In order to minimise the dropped-call rate while maintaining a certain voice quality level, the signal coverage should be based on the probability (say 90%) that all the received signals will be above the specified threshold received signal level.

If α_n is the weight value for those calls having 'n' number of hand-offs, and μ is the probability that the signal is below the specified cochannel interference level in an interference-limited system then the probability of a dropped call when the call has gone through 'n' number of hand-offs can be approximately given as

$$P_d = \Sigma\, \alpha_n \quad 1 \;\; (1 \;\; \mu\,)^n \tag{9.31}$$

9.4.8 Service Quality and Special Features

For an FDMA system, the capacity of a cell is equal to the number of frequency channels allocated to it. Ideally, the number of available channels in a cell would equal the total number of mobile subscribers active at any time. In practice, it is not feasible to have the capacity to handle any possible load at all times. Incidentally, not all mobile subscribers place calls at the same time and so it is reasonable to allocate the number of channels in the cellular network to be able to handle some expected level of load.

Consider a cell which is able to handle N simultaneous mobile subscribers (capacity of N channels) that have L potential mobile subscribers. If $L > N$, the system is referred to as nonblocking; all calls can be handled all the time. If $L < N$, the system is blocking; a subscriber may attempt a call and find the capacity fully in use and therefore be blocked. For a blocking system, the fundamental performance criterion is the degree of blocking that is the probability that a call request will be blocked. Alternatively, what capacity (N) is needed to achieve a certain upper bound on the probability of blocking. If blocked calls are queued for service, then the performance criterion is the average delay. Alternatively, how much capacity is needed to achieve a certain average delay is the main concern.

To achieve desired service quality, there are three main criteria:

(a) Coverage The cellular system should serve an area as large as possible. However, with radio coverage, radio system usually tries to cover 90% of an area in flat terrain and 75% of an area in hilly terrain.

(b) Required Grade of Service It means less blocking probability (< 0.02).

(c) Number of Dropped Call It must be kept low. A high drop rate could be caused by either coverage problem or hand-off problem related to inadequate channel availability.

Special Features A cellular system would like to provide as many special features as possible, such as call forwarding, call waiting, automatic roaming, SMS, etc. For some special services, the customers might have to pay extra charges.

9.5 PLANNING A CELLULAR SYSTEM

The objective of the cellular system planning is to find the optimal network solution taking into account the existing infrastructure, scalability in terms of capacity demand and future technology evolution, as well as the technical and geographical constraints.

The cellular system planning can be broadly categorised into three main stages: regulations and market situations, engineer's role, and finding solutions.

The first requirement is to become fully familiar with rules and regulations administered by the central and state government agencies applicable throughout the country/state; city/town regulations regarding its building codes and zoning laws. Sometimes, waivers need to be obtained in advance to implement certain clauses. One must be sure that the plan is workable and can be executed with success. The marketing department must work upon the tasks such as prediction of gross revenue, understanding competitors in the area of interest, and geographic coverage. Gross revenue can be predicted based on various aspects like the population, average income, business types, and business zones. Understanding competitors means knowing about the competitor's financial situation, area coverage, system performance, and the number of subscribers. Any system should provide a unique and outstanding service to overcome the competition. Decision of geographic coverage needs to be implemented depending upon the type of service to be provided.

An engineer's role is quite vital in successful implementation of an overall plan. The cellular architecture should be planned in such a way so as to use minimum numbers of cell-sites to cover the given area to initiate a cellular mobile service. Then the number of voice channels required to handle the traffic load at busy hours should be determined. The interference problems, such as cochannel and adjacent channel interference, and the intermodulation products generated at the cell-site should be critically examined. The blocking probability of each call at each cell-site should be properly estimated. There should be a contingency plan to absorb more new mobile subscribers in future depending on service charges, system performance and seasons of the year.

In order to find solutions to various issues, practical design tools should be used to create a plan that uses minimum number of cell-sites, to determine the number of voice channels required to handle the traffic load at the busy hours, to find way to reduce interference problems, to try to minimise blocking problems and develop new technologies to utilise fully the allocated spectrum. A large spectrum should be requested, if necessary, after the analysis of spectrum efficiency due to the natural limitations and meet the increasing demand.

Facts to Know!

A good plan is necessary not only to ensure a good service to the customers but also to maximise the utilisation of resources.

9.5.1 Network Planning Process

Business planning forecasts future subscriber base and traffic volumes over several years, that is, the roll-out period. In the network-planning phase, the business plan is used as input for making a cellular network plan. Commercial launch of a cellular network takes place after the implementation phase. After the commercial launch, customer training and technical support starts along with network management.

Figure 9.9 depicts the various activities involved in a network life cycle in a flow-chart form.

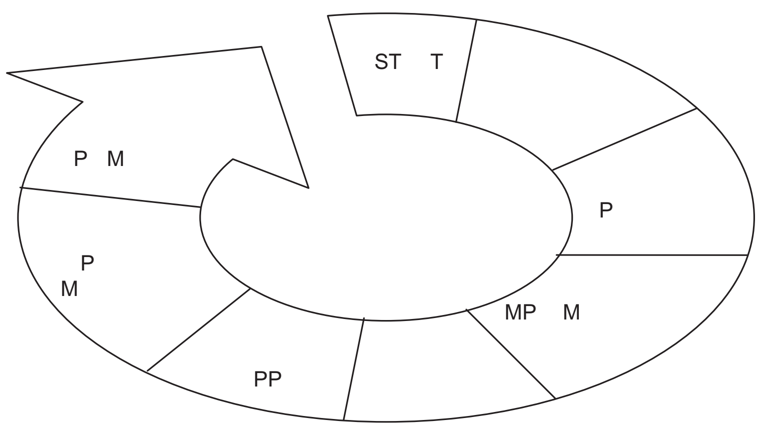

Fig. 9.9 *Network life cycle*

The planning process for a cellular system can be well described with the help of the Network Life Cycle (NLC). It starts with the initial business decisions, covers the business planning and implementation phases, the network operations after commercial launch and it extends over the network maintenance, expansions and optimisation activities. After a decision about the network establishment or expansion has been done, the network life cycle begins again with business planning, or forecasting. The service provider, sometimes in consultation with the infrastructure supplier, usually does business forecasting.

After the network has been running for some period of time, a network optimisation is done with the objective to detect possible places of low utilisation level of bandwidth. The basic problem in managing the NLC-activities arises from many uncertainties, which degrade the business and network planning. The dynamic nature of the planning environment prevents finalising the network roll-out plan over an extended time horizon in advance. All assumptions of the plan have to be continually checked and the plans upgraded accordingly to their variations and relevance.

The aim of business forecasting is to characterise how the network will look like in the future, how it will expand with the traffic growth, and how it will evolve with technology.

9.5.2 Nature and Scope of Planning Areas

Cellular system planning requires several different kinds of planning expertise, each of them binding a number of people. In this kind of network planning project, all the planning tasks demanding different kind of expertise must be divided between separate planning teams. Some of the major planning areas include

- Radio network planning
- Transmission network planning
- Signalling network planning
- External connectivity network planning

A *radio-network planning* project covers air-interface related planning tasks, for example, a cellular-system coverage plan, frequency plan, routing-area plan, user-capacity dimensioning and parameter plan. The purpose of the *transmission network planning* is to perform a system capacity plan and to define the best way to utilise the existing transmission solutions. Transmission network planning expertise is required in interface planning, where increase of signalling traffic may require more signalling link capacity. The scope of the *signalling network planning* is to integrate the operator's existing signalling functionality to the required signalling functions as per the intended application. It also involves planning the re-dimensioning of the existing signalling links if it is required due to the expansion of additional services. *External connectivity network planning* includes interface planning, dimensioning and backbone planning.

Issues like network topology, element and link capacities, IP addressing schemes, routing protocols, redundancy and security are planned in detail, wherever applicable. When planning connections to external networks, security considerations are very important. Security solutions are based on advanced router and firewall technologies.

9.5.3 Timing and Information Flow

Network planning is teamwork between several network planners and engineers, and between several planning teams. Output of one planning team is input for another planning team. In a big planning project, good project management is essential for successful network plan. The customer provides the business plan containing information about data and subscriber volume estimations, future growth estimation and plans.

The cellular-network planning will be based on the operator's business plan. Figure 9.10 gives a short description of how a cellular-network planning project could proceed.

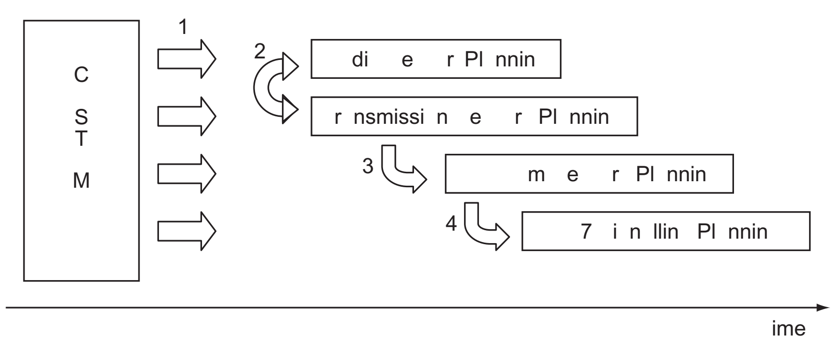

Fig. 9.10 *Timing and information flow in a cellular-network planning project*

Radio and transmission network planning proceed hand in hand, that is, dimensioning and planning results of radio planning are needed by transmission planning and vice versa. Radio and transmission planning provide information to signalling network planning; containing data traffic per cell-site, existing transmission solutions, and provides information to external connectivity planning covering backbone network topology.

9.5.4 A Typical Roll-out Plan

An annual roll-out plan needs to be made for a cellular network to be built in several phases in which the cellular network operator invests into new infrastructure. The planning starts from designing the target network in the end of the estimated roll-out period. It then continues by stepping backwards in order to create a consistent growth path from the very beginning in such a way that the investments made in the first years could be utilised in later phases. The cellular operator may either re-use certain network nodes, as they run out of capacity, in other parts of the network, or the operator invests in the first few years into higher capacity equipment, into those required in later phases, to save higher costs in a long run. A roll-out network plan covers the increasing traffic generated by existing users (for example, due to new services), or it extends the network geographically into new areas (to cover new cities, business areas).

Key Terms

- Bandwidth
- Cell-site
- Downlink
- Full-duplex transmission
- Grade of service
- Mobile Telecommunications Switching Office (MTSO)
- Public Switched Telecommunications Network (PSTN)
- Quality of Service (QoS)
- Spectral Efficiency
- Trunking

Summary

In this chapter, the essential components of a basic cellular system including the design considerations are presented. Since a cellular system is designed to establish calls between one mobile user to another mobile user as well as a landline subscriber to a mobile user, the call procedures are described in detail. The performance-determining parameters such as voice quality, service quality, grade of service, radio capacity, and spectrum efficiency under different situations are also presented here. In order to provide a satisfying service to the customers, a good system plan is necessary. An overview of main aspects of cellular system planning leading to optimum utilisation of network resources is given. There are a number of different types of wireless communication systems which have been evolved across the world over the time depending upon the availability of support technology and customer requirements. An account of various wireless communication systems with a special emphasis on cellular mobile communication systems is the main focus of attention in forthcoming chapters.

Important Equations

- ❑ $A_{av} = \lambda \times H$ (9.1)
- ❑ $N_t = (B_t - 2 \times B_g) / B_c$ (9.4)
- ❑ $\eta_1 = N_{d/cell} / (B_t \times A_{cell})$ (9.10)

- ❑ $\eta_2 = (\eta_{tr} \times N_{d/cell}) / (B_t \times A_{cell})$ Erlangs/MHz/km^2 (9.11)
- ❑ $N_u = (B_t - 2 \times B_g) / B_c$ (9.12)
- ❑ $\eta_{t(frame)} = \eta_{t(slot)} \times \eta_{t(data)} = (T_f \quad T_p \quad T_t) / T_f \times (L_d / L_s)$ (9.18)
- ❑ $\eta_t = \eta_{t(sys)} \times \eta_{t(mod)} / K$ bps/Hz/cell (9.21)
- ❑ $J = (B_t / B_c) \times (1 / K)$ (9.22)
- ❑ $(C/I)_{eq}$ in dB $= (C/I)_{min}$ in dB $+ 20 \log(B_c / B_c)$ (9.30)

Short-Answer Type Questions with Answers

A9.1 What are the major operational limitations of conventional mobile telephone systems

The conventional mobile systems deploy a high transmitter power base station in a large autonomous geographical service area providing limited service capability. The inefficient frequency spectrum utilisation, low subscriber capacity, poor service performance, high blocking probability during busy hours, and no continuation of call between different service areas are the major concerns in conventional mobile systems.

A9.2 List the major functions of MTS in a cellular system.

MTSO is the central coordinating element for all the cell-sites connected to it. The MTSO provides a centralised control, administration and maintenance center for the entire cellular network including interfaces with the public telephone network over voice trunks and data links. The MTSO controls allocation of radio-frequency channels, signaling, call set-up, switching, call processing, and call supervision functions.

A9.3 Distinguish between forward control channel and reverse control channel.

On forward control channel, the signaling and control information is transferred from the base station to the mobile subscribers and used to continuously broadcast all of the traffic requests for all the mobile subscribers in the system. Forward control channels are continuously monitored by mobile subscribers when they are not engaged in voice call. The signaling and control information from the mobile subscribers to the base station is transferred on the reverse control channel. Control channels carry data messages for call initiation and service requests.

A9.4 Differentiate between vehicle-installed and handheld mobile phone equipments.

Vehicle-installed and handheld mobile phone equipments are essentially identical except that vehicle-installed mobile phone equipments have relatively large output power, have an omnidirectional gain antenna mounted on the rooftop of the vehicle, and can operate from vehicle batteries. Handheld mobile equipments have a lower output power, have a less efficient antenna, and operate exclusively on batteries.

A9.5 Why is it necessary for a mobile subscriber to get registered?

Each and every mobile subscriber must be registered at one of the MTSOs or MSCs. This is necessary for authentication and identity verification, access privileges, and also for billing purpose. Moreover, the cellular system needs to know whether the mobile subscriber is currently located in its own home service area or is visiting some other service area. This enables incoming calls meant for roaming mobile subscribers to be routed to an appropriate cell location and assures desirable support for mobile-originated calls.

A9.6 What types of messages are carried in the broadcast forward control channels?

The cell-site periodically broadcasts control signals to determine and test nearby mobile subscribers. This is done by exchanging signals known as handshake signals between the cell-site and the mobile subscribers. Each mobile subscriber listens for broadcast control signals transmitted by the cell-site. Some of the information contained in the broadcast forward control signals includes cellular network identifier, timestamp, ID (identification) of the paging area, gateway MSC address, and other system parameters of the cell-site. The mobile subscriber uses this information to locate the nearest cell-site and establish an appropriate communication link with the cellular system.

A9.7 List the major factors for occurrence of hand-off.
The occurrence of hand-off depends on the cell size, radio signal coverage boundary, received signal strength, fading effect, mobility of the subscriber, type of operating terrain and surroundings, and man-made noise.

A9.8 Define the terms Erlang traffic intensity and trunking efficiency.
e represents the amount of traffic intensity carried by a channel that is completely occupied (that is, one call-hour per hour or one call-minute per minute).
i i te it is the average channel occupancy measured in Erlangs, and is a measure of channel time utilisation. This is a dimensionless quantity and may be used to measure the time utilisation of single or multiple channels.
i e i ie is a measure of the number of mobile subscribers that can be offered a specified grade of service or blocking probability with a particular configuration of fixed channels.

A9.9 Distinguish between Erlang B system and Erlang C system.
The Erlang B system of a trunked cellular system offers no queuing for call requests. If no channels are available, the incoming call request is blocked without access to the system. This type of trunking is also called *blocked calls cleared* or *blocked calls lost*. The Erlang C system, also called *blocked calls delayed*, is defined as the probability that a call is blocked after waiting a specific length of time in the queue.

A9.10 Suggest some techniques other than frequency reuse to enhance the spectrum utilisation in a mobile communication system.
In a mobile communication system, the spectrum utilisation can be enhanced by various techniques such as choice of multiple access method, channel-assignment schemes, bandwidth reduction, and data compression to reduce the transmission rate.

A9.11 What is meant by spectral efficiency of FDMA
The spectral efficiency of FDMA, in general, is defined as the ratio of total bandwidth available for user data transmission and allocated system bandwidth in a cluster. It is apecified in terms of channels/MHz/km^2 or Erlangs/MHz/km^2.

A9.12 Mention some important functions of telecom engineers in planning of a cellular system.
A telecom engineer's role is quite vital in successful planning and implementation of a cellular system. The cellular architecture should be planned in such a way so as to use minimum numbers of cell-sites to cover the given area to initiate a cellular mobile service. Then the number of voice channels required to handle the traffic load at the busy hours should be determined. The interference problems, and the blocking probability of each call at each cell-site should be properly estimated. There should be a contingency plan to absorb new mobile subscribers in future depending on service charges, system performance and seasons of the year.

A9.13 What are the major planning areas in cellular system design and implementation
The major planning areas include radio network planning, transmission network planning, signalling network planning, and external connectivity network planning. Radio network planning is concerned with coverage plan, frequency plan, routing area plan, and user-capacity dimensioning plan. The transmission network planning is to perform a system capacity. The signalling network planning is to integrate the operator's existing signalling functionality to the required signalling functions as per the intended application. External connectivity network planning includes interface planning, and backbone planning.

Self-Test Quiz

S9.1 In a cellular system, the air interface is between
(a) mobile subscriber and cell-site
(b) cell-site and mobile telephone switching office
(c) two mobile telephone switching offices
(d) mobile telephone switching office and PSTN

S9.2 The cell-site equipment is usually located of a cell.
(a) at the centre of the coverage region
(b) at the edge of the coverage region
(c) anywhere in the coverage region
(d) either at the center or edge of the coverage region

S9.3 The radio transmitting equipment at the cell-site operates at RF power than do the mobile equipments.
(a) considerably higher
(b) considerably lower
(c) almost same
(d) either higher or lower

S9.4 Generally, are used at each cell-site or each sector of a cell in a cellular system.
(a) one Tx antenna and one Rx antenna
(b) one Tx antenna and two Rx antennas
(c) two Tx antennas and one Rx antenna
(d) two Tx antennas and two Rx antennas

S9.5 Two Rx antennas provide space diversity to counteract the effects of
(a) fading
(b) multipath interference
(c) cochannel interference
(d) adjacent channel interference

S9.6 In a typical cellular system, the control channels use modulation scheme.
(a) FM (b) FSK
(c) PSK (d) FSK or PSK

S9.7 A(n) is used to separate the transmit and receive signals at the mobile subscriber unit.
(a) antenna (b) duplexer
(c) transceiver (d) control unit

S9.8 When a mobile subscriber originates a call, a call initiation request is sent on the
(a) forward control channel
(b) reverse control channel
(c) forward voice channel
(d) reverse voice channel

S9.9 Usually the telephone voice quality is around
(a) $MOS = 5$ (b) $MOS \geq 4$
(c) $MOS \approx 3$ (d) $MOS \leq 3$

S9.10 The is a measure of the ability of a mobile subscriber to access a cellular system during the busiest hour.
(a) circuit merit level
(b) mean opinion score
(c) grade of service
(d) service quality

S9.11 The GOS is typically specified as the probability that a call is
(a) blocked (b) dropped
(c) delayed (d) completed

S9.12 The traffic intensity offered by each mobile subscriber is equal to , where λ represents the call request rate, and H denotes the call holding time.
(a) $\lambda \times H$ (b) λ / H
(c) H / λ (d) $\lambda + H$

S9.13 If the calling rate averages 20 calls per minute and the average holding time is 3 minutes then the offered traffic load in Erlang is
(a) 60 (b) 6.66
(c) 0.15 (d) 23

S9.14 If each mobile subscriber averages two calls per hour at an average call duration of three minutes then the traffic intensity per mobile subscriber is
(a) 2 Erlangs (b) 3 Erlangs
(c) 6 Erlangs (d) 0.1 Erlangs

S9.15 Cell sectoring ________ the trunking efficiency while improving the signal-to-interference ratio for each user in the system.
(a) does not change at all
(b) increases drastically
(c) decreases significantly
(d) improves marginally

S9.16 The overall spectral efficiency of a TDMA system is also a function of ________ in the system.
(a) frequency reuse factor
(b) digital-modulation scheme used
(c) either (a) or (b)
(d) both (a) and (b)

S9.17 The overall spectral efficiency of a TDMA system is expressed in terms of
(a) bps/MHz/cluster
(b) bps/Hz/cell
(c) channels/MHz/km^2
(d) Erlangs/MHz/km^2

S9.18 Typically, the minimum required *C/I* is about ________ for narrowband digital cellular systems.
(a) 24 dB (b) 18 dB
(c) 12 dB (d) 9 dB

S9.19 The cellular radio system usually tries to cover ________ radio coverage of a service area in flat terrain.
(a) 100% (b) 90%
(c) 75% (d) 50%

S9.20 Frequency plan is a part of
(a) radio network planning
(b) transmission network planning
(c) signalling network planning
(d) external connectivity network planning

Answers Self-Test Quiz

S9.1 (a); S9.2 (d); S9.3 (a); S9.4 (b); S9.5 (a); S9.6 (d); S9.7 (b); S9.8 (b); S9.9 (b); S9.10 (c); S9.11 (a); S9.12 (a); S9.13 (a); S9.14 (d); S9.15 (c); S9.16 (d); S9.17 (b); S9.18 (c); S9.19 (b); S9.20 (a)

Review Questions

Q9.1 Compare the features of cellular mobile communication systems with that of conventional mobile telephone systems.

Q9.2 List and describe the essential components of a cellular telephone system.

Q9.3 What are the design issues in various components of a cellular system?

Q9.4 Explain the various steps involved in placing a call from
(a) Mobile to a landline phone
(b) Landline phone to a mobile phone

Q9.5 Explain the following performance criteria used to evaluate the performance of communication systems:
(a) Voice quality
(b) Service quality

Q9.6 Why is it essential for the MTSO to control the transmitted power level at both the cell-site and the mobile subscribers?

Q9.7 What are the coordinating measures that have to be taken by two cellular service providers operating in the same frequency band between two adjacent areas to prevent interference between each other? What are the different methods used?

Q9.8 Draw the block diagram of an analog cellular communication system and explain the functions of each block.

Q9.9 Explain the different MTSO Interconnections to wireline networks and cell-sites.

Q9.10 A good plan is necessary to ensure a good service to the customers in a cellular system. Justify this statement, describing the various stages of planning with a special emphasis on the engineers' role.

Analytical Problems

P9.1 A cellular system is capable of coping with hand-offs once every 2 minutes.

(a) What is the maximum cell radius if the system must be capable of working with mobile subscribers installed in a vehicle traveling at highway speeds of 120 km/h in an open area?

(b) Suppose this cellular system provides service in an urban area with maximum vehicle speeds of 60 km/h. What is the maximum cell radius for this urban system?

P9.2 A cell-site and mobile are separated by 5 km. What is the propagation time for a signal traveling between them

P9.3 Consider three different cellular systems that share some characteristics such as the frequency bands are 825 MHz to 845 MHz for mobile unit transmission and 870 MHz to 890 MHz for base-station transmission, and a duplex circuit consists of one 30-kHz channel in each direction. The cellular systems are distinguished by the reuse factor, which is 4, 7, and 12 respectively.

(a) Suppose that in each of the three cellular systems, the cluster of cells (4, 7, and 12) is duplicated 16 times. Find the number of simultaneous communications that can be supported by each system.

(b) Find the number of simultaneous communications that can be supported by a single cell in each system.

(c) What is the area covered, in terms of cells, by each system?

(d) Suppose the cell size is the same in all three cellular systems and a fixed area of 100 cells is covered by each system. Find the number of simultaneous communications that can be supported by each system.

P9.4 How many number of users can be supported with 40 channels at 2% blocking? Assume $H = 105$ s, $\lambda = 1$ call/hour.

P9.5 How many users can be supported for 0.5% blocking probability for the following number of trunked channels in a blocked-calls cleared system?

(a) 4 Use total offered traffic load = 0.701 from Erlang B Table

(b) 20 Use total offered traffic load = 11.1 from Erlang B Table

(c) 40 Use total offered traffic load = 27.3 from Erlang B Table

Assume that each user generates 0.1 Erlangs of traffic.

P9.6 In a particular cellular service area, there are three competing trunked cellular systems *A*, *B*, and *C*. System *A* has 49 cells with 100 channels each, System *B* has 98 cells with 57 channels each, and System *C* has 394 cells with 19 channels each. Find the number of users that can be supported by each system as well as in total service area at 2% blocking of probability if each user averages two calls per hour at an average call duration of three minutes.

P9.7 A geographical area of 3,000 km^2 is covered by a cellular system using a seven-cell hexagonal reuse pattern, with each cell having a radius of 6 km. The system is allocated 40 MHz of spectrum with a full duplex channel bandwidth of 60 kHz. Assumeing a blocking probability of 2% is specified for an Erlang B system, compute

(a) the number of cells in the service area and the number of channels per cell

(b) the theoretical maximum number of users that could be served at one time by the system

(c) traffic intensity of each cell

(d) the maximum carried traffic

(e) the total number of users if the offered traffic per user is 0.03 Erlangs

(f) the number of users per channel (considering channel reuse)

P9.8 For a $K = 7$ system with a blocking probability of 1% and average call length of two minutes, find the traffic-capacity loss due to trunking for 57 channels when going from omnidirectional antennas to 60 sectored antennas. (Assume that blocked calls are cleared and the average call rate per user is one per hour.)

P9.9 A cellular system has a total of 395 allocated voice channels. The system has a frequency reuse factor of 4. Assume that the traffic distribution is uniform with an average call holding time of 2 minutes and the call blocking during the system busy hour is 2%. Determine the number of calls per cell per hour.

P9.10 Repeat P9.9 for $K = 7$ and $K = 12$.

P9.11 A certain area is served by a cellular radio system with a total number of 84 cells and a cluster size K. A total number of 300 traffic channels are available for the system. The mobile users are uniformly distributed over the area covered by the cellular system, and the offered traffic per user is 0.04 Erlang. Assume that blocked calls are cleared and the designated blocking probability is 1%.

(a) Determine the maximum carried traffic per cell if the cluster size $K = 4$ is used. Repeat for cluster sizes $K = 7$ and 12.
(b) Determine the maximum number of mobile users that can be served by the system for a blocking probability of 1% and cluster size $K = 4$. Repeat for cluster sizes $K = 7$ and 12.

P9.12 In the AMPS system, the one-way bandwidth available to a single operator is 12.5 MHz with a channel bandwidth of 30 kHz and 21 control channels. Given the system parameters as cell area = 8 km^2; total coverage area = 4000 km^2; frequency reuse factor, $K = 7$; average number of calls per user during the busy hour, $\lambda = 1.2$; average call holding time, $H = 100$ s; call-blocking probability = 2%.

(a) How many voice channels are there per cell?
(b) Determine the total traffic carried per cell, in Erlangs/cell and Erlangs/km^2 using Erlang B Table and a simple straight-line interpolation if needed.
(c) Calculate the number of calls/hour/cell.
(d) Calculate the number of calls/hour/km^2.

P9.13 Consider a 7-cell hexagonal pattern cellular system covering an area of 3100 km^2. The system consists of a total of 395 channels. The traffic in the seven cells is as follows:

Cell number	1	2	3	4	5	6	7
Traffic (Erlangs)	30.8	66.7	48.6	33.2	38.2	37.8	32.6

Each user generates an average of 0.03 Erlangs of traffic per hour, with an average holding time of 120 s. The system is designed for a grade of service of 0.02. Determine the following:

(a) The number of subscribers in each cell
(b) The number of calls per hour per subscriber
(c) The number of calls per hour in each cell
(d) The number of channels required in each cell using Erlang B Table
(e) The total number of subscribers
(f) The average number of subscribers per channel
(g) The subscriber density per km^2
(h) The total traffic in Erlangs
(i) The total traffic per km^2

P9.14 Consider a geographical service area of 1500 km^2 to be covered by a hexagonal cellular system with a 7-cell reuse pattern. Suppose each cell has a radius of 5 km and the system is allocated a 25-MHz spectrum, with a full-duplex channel bandwidth of 30 kHz and a total 40 of kHz guard bands. The system uses FDMA with 14 control channels. Determine

(a) the number of cells in the service area
(b) the number of channels without frequency reuse
(c) the cell capacity
(d) the system spectral efficiency in channels/MHz/km^2
(e) the system spectral efficiency in Erlangs/MHz/km^2, if each user makes two calls in a busy hour on the average and each call lasts 2 minutes. Assume that the trunking efficiency is 0.9.

P9.15 In a GSM cellular system, the total system bandwidth (50 MHz) is divided into 25 MHz each for uplink and downlink. There are a total of 125 duplex RF carriers, each having a bandwidth of 2 × 200 kHz with a duplex separation of 45 MHz between uplink and downlink channels. Each carrier supports 8 users using TDMA with a frame duration of 4.615 ms. Using GMSK modulation; the bandwidth efficiency is 1.35 bits/s/Hz. The speech codec rate is 13 kbps, and the channel coding results in a coded bit rate of 22.8 kbps. Consider only the normal speech frames, each frame consisting of 8 time slots. And each time slot has 156.25 bits—3 start bits, 116 coded speech bits, 26 training bits, 3 stop bits, and 8.25 bits for guard period. Assume a frequency reuse factor of 4. Determine

(a) the number of simultaneous users that can be accommodated in each cluster,
(b) the cell capacity,
(c) the spectral efficiency of TDMA,
(d) the overall spectral efficiency in bits/s/Hz/cell, and
(e) the overall system spectral efficiency in Erlangs/MHz/km^2, if each user makes two calls in a busy hour on the average and each call lasts 5 minutes. Assume that the trunking efficiency is 0.85. The system has a service area of 4500 km^2, and each cell has a coverage area of 7.5 km^2.

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10

Wireless Communication Systems

Many wireless and mobile radio communication systems have been developed throughout the world, and more standards are available for implementation. This chapter covers some of the recent past and current wireless communication systems. The class of systems include Paging and Messaging Systems, Cellular Telephone Systems, Cordless Telephone Systems, Wireless Local Loop and Local Multipoint Distribution Service. The design aspects of first-generation analog cellular systems, followed by second-generation US digital cellular systems are covered in details so as to get acquainted with various aspects of cellular system implementations. A brief overview of personal digital cellular is given in the end for comparison purpose.

10.1 PAGING AND MESSAGING SYSTEMS

Paging and messaging systems are wireless mobile communication systems that send brief text or numeric or alphanumeric messages to a pager subscriber. Paging and messaging systems are typically used to inform a pager subscriber that there is an urgent need to call a particular landline telephone number or move to a known location to receive further instructions. The traditional paging system uses widely spaced high-power pager transmitters, each covering a considerable large geographical service area. In addition, all transmitters in a given paging system operate on the same frequency, either in the VHF range (at about 152 MHz or 158 MHz band) or in the UHF range (454.025 MHz–454.650 MHz). All pages (messages) are sent by all the pager transmitters in the system. The paging system is not spectrum efficient at all but it works because short paging messages require much smaller bandwidth.

A message (also called a page) is sent to a paging subscriber via the unique paging system access number. The paging transmission system then broadcasts the requested page on a radio carrier via several base station transmitters installed throughout the service area.

The traditional subscriber pager receiver is simply a fixed-tuned wireless radio receiver that uses a transmitted code to identify received messages meant for it. The earliest page

Facts to Know!

Paging is the simplest radio communication concept of all the major organised mobile radio systems. In 1956, the first paging systems were installed in a hospital in London. The wide-area paging system were developed in US and Canada in the early 1960s and introduced in Europe in 1964.

Facts to Know!

Paging systems are wireless communication systems that are designed to send brief numeric / text / alphanumeric messages to its subscriber. It's a one-way messaging system in which the base station sends messages to all subscribers in its coverage area. The paging system transmits the message also known as page, along with the paging system access number, throughout the service area using base station, which broadcasts the page on a radio link.

receivers used to beep a tone whenever they received a message. That was simply an indication to the user of the pager to make a telephone call to the central office of the pager system to find out further details of the page. However, most modern pager systems transmit a short text or numeric or alphanumeric message to the pager receiver. The numeric message is usually a telephone number for the pager subscriber to make a call, but may have numerical codes for standard pre-defined messages.

The advantages of pagers are their extremely small size, low cost, and long battery life (typically several weeks compared to few hours for cellular mobile phones and couple of days for PCS mobile phones). The extended battery life of one-way pagers results due to no radio transmitter or audio receiver amplifier. Moreover, pagers have the ability to go into a sleep mode, waking up at short intervals to check for any messages. Simple paging systems may cover a particular building or limited range of 2 to 5 km. Wide-area paging systems can provide worldwide coverage. Though pager receivers are simple, small and cheap, the pager transmission system required is quite complex and costly.

10.1.1 A Basic Paging Network

A basic paging network comprises of several high-power transmitters connected to a central paging control system. Paging systems are designed to provide reliable communication to pager subscribers wherever they are. This necessitates high transmitter powers of the order of hundreds of watts. All pages are sent from all transmitters simultaneously and there is no need for the system to know the location of any pager. The use of multiple transmitters has the added advantages of reducing the number of weak signal spots and completely avoiding any need for hand-offs. Figure 10.1 shows a basic paging network.

Paging messages are relatively infrequent and usually require only a few seconds to transmit. Small RF bandwidths are used to maximise the signal-to-noise ratio at each paging receiver. Thus, low transmission data rates of the order of 6400 bps or less are used for maximum coverage from each paging transmitter. Many pagers can share a common assigned frequency using TDMA technique.

Each pager receiver has a unique identification address called a *capcode.* This capcode is transmitted by pager transmitters every time it is paged to decode it and receive the message. Transmissions addressed to other pagers are simply ignored. The pagers are receive-only devices with no transmission capability. Hence, there is no way for the system to know whether the page has been received by the desired pager or not.

10.1.2 A Wide-Area Paging Network

A wide-area paging network consist of the paging control centre, several wireline (coaxial cable or optical fiber) or satellite communication links, paging terminals, base station transmitters, and Tx antennas mounted on tall towers. The base-station transmitters simultaneously broadcast a page in a large geographical service area. Figure 10.2 shows a block diagram of a wide-area paging network.

The pages are received by the paging control system either from the PSTN via dedicated trunk lines or from the Internet via e-mail. The paging control system then dispatches the received pages throughout several paging terminals and pager transmitters, using point-to-point microwave radio links or wireline links or even

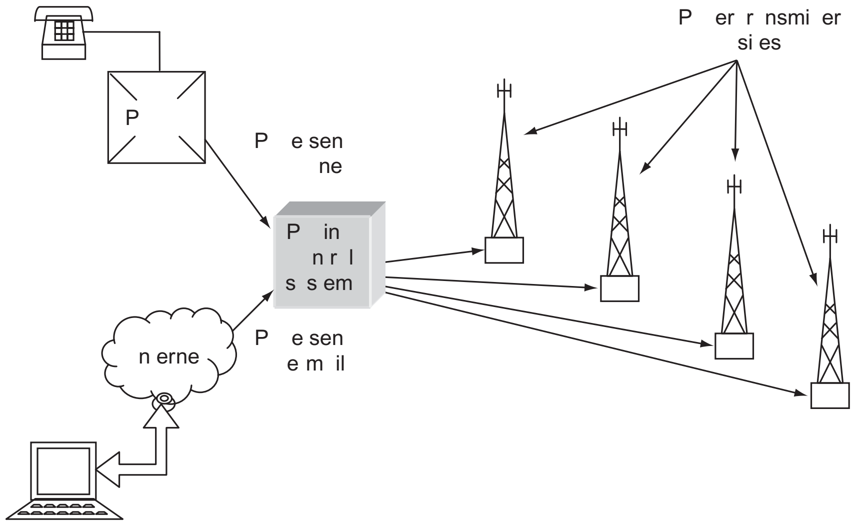

Fig. 10.1 *A basic paging network*

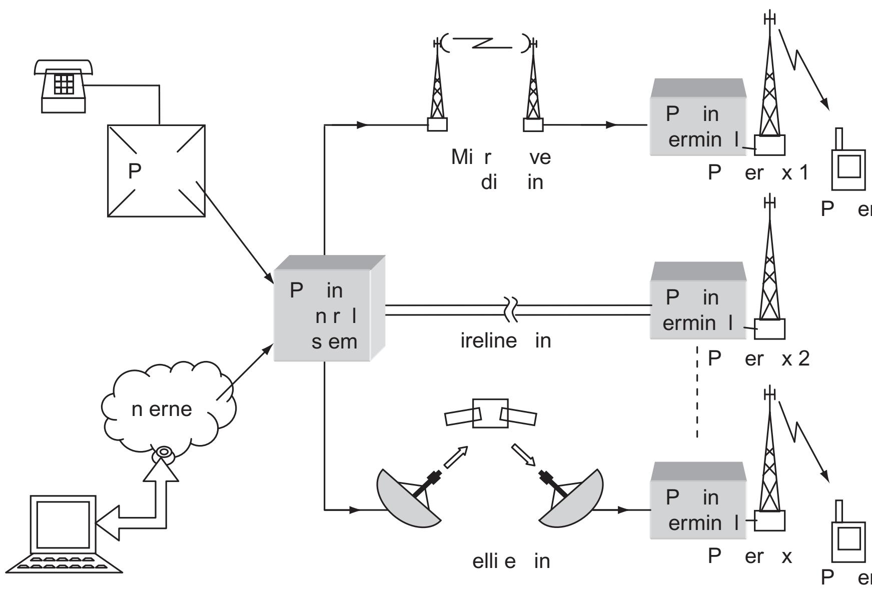

Fig. 10.2 *A wide-area paging network*

via satellite communication links at the same time. Satellite communication links are often used to transmit pages covering very large distances to local paging systems. In addition, there are low-earth orbit (LEO) satellite systems that can send paging messages directly to individuals called pagers. These systems are useful in remote areas without terrestrial paging systems due to high operating costs involved.

Roaming feature may be used in wide-area paging network. The pager service provider informs the system about the movement of the particular pager receiver away from the local paging network. This does not require sending all the pages by all the pager transmitters over a very large area, thereby minimising the need of resources.

Facts to Know!

The paging systems can be of two types: manual paging system and automatic paging system. In a *manual paging system*, a message is sent to the paging operator through telephone call by the calling subscriber. The message is then delivered to the pager through the paging network by the operator. In an *automatic paging system*, the incoming requests are automatically processed by the paging terminal and then this information is delivered to the pager. Automatic paging systems are mostly used.

10.1.3 Types of Messages in Paging Systems

Typically, there are four distinct types of messages which can be transmitted by different paging systems.

- Alert tone message
- Text string message
- Digital string message
- Voice message

Alert Tone Message In the alert tone message, a dedicated pager number is assigned to the pager receiver, which is also known as tone pager. To generate tone-type messages, the pager is triggered by dialing this number. In addition to simplicity, the main advantage of tone paging is that it utilises a small amount of airtime.

Digital String Message In digital string message, the receiver is a numeric pager. The string can be the telephone number of the caller or a coded message. This coded message is generated on request of the caller by the paging system and is decoded by a code book built into the pager. This type of paging takes less amount of airtime.

Text String Message In the text string message, the receiver is an alphanumeric pager, which has a large screen to display the text strings. This type of messaging is becoming more popular than numeric messaging.

Voice Message In the voice message, a voice message can be transmitted in special types of tone paging systems after the beep.

10.1.4 Paging Systems Standards

Table 10.1 lists the most common paging system standards along with some of the major parameters.

The Post Office Code Standard Advisory Group (POCSAG) paging standard is the most common protocol for one-way paging systems. The system employs Frequency Shift Keying (FSK) modulation scheme in an RF channel bandwidth of 25 kHz. It can support the transmission data rates of 512 bps, 1200 bps, and 2400 bps.

Table 10.1 *Major paging systems standards*

Paging systems standard	*requency band*	*Multiple access technique*	*Channel bandwidth*	*Modulation scheme*	*Region of operation*
POCSAG, GSC	Several	Simplex	12.5/25 kHz	FSK	North America
FLEX, ReFLEX	Several	Simplex	15/25 kHz	4-FSK	North America
ERMES	Several	FDMA	25 kHz	4-FSK	Europe
NEC	Several	FDMA	10 kHz	FSK	Japan
NTT	280 MHz	FDMA	12.5 kHz	FSK	Japan

EXAMPLE 10.1 Spectral efficiency of POCSAG paging system

Define spectral efficiency. Calculate the spectral efficiency of the POCSAG paging system in terms of bps per hertz of RF bandwidth at each of its three specified data rates.

Solution

Spectral efficiency is defined as the ratio of the permissible source data rate in bits per second to the available channel bandwidth in Hz . It is measured in bits/s/Hz. It is a measure of performance of intense practical importance in the design of wireless communication systems.

Channel bandwidth of POCSAG paging system = 12.5 kHz (standard)

Step 1. *To calculate the spectral efficiency at 512 bps*

Transmission data rate 1 = 512 bps (standard)

The spectral efficiency = Data rate/Channel bandwidth

Or, Spectral efficiency = 512 bps/12.5 kHz

Hence, Spectral efficiency at 512 bps = **0.04 bps Hz**

Step 2. *To calculate the spectral efficiency at 1200 bps*

Transmission data rate 2 = 1200 bps (standard)

The spectral efficiency = Data rate/Channel bandwidth

Or, Spectral efficiency = 1200 bps/12.5 kHz

= **0.096 bps Hz**

Step 3. *To calculate the spectral efficiency at 2400 bps*

Transmission data rate 3 = 2400 bps (standard)

The spectral efficiency = Data rate/Channel bandwidth

Or, Spectral efficiency = 2400 bps/12.5 kHz

= **0.192 bps Hz**

POCSAG messages are sent in the form of data frames, also called *batches*. Several frames may be sent together in one transmission. Simple beep pagers do not need any message frames; an address frame is sufficient. Alphanumeric pagers require more message frames than do the simpler numeric pagers.

Figure 10.3 shows an illustration of a POCSAG signal protocol. Each page transmission begins with a preamble of 576 bits of the known bit sequence 1010101010. Each frame consists of 544 bits, out of which

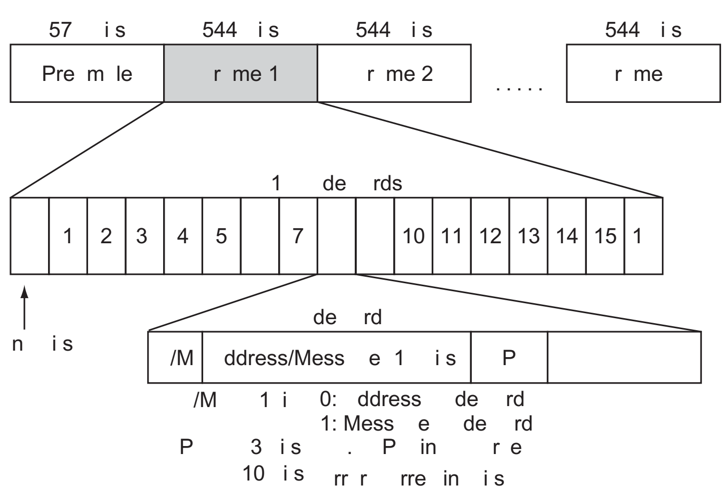

Fig. 10.3 *POCSAG signal protocol*

the first 32-bits are synchronising bit patterns. This is then followed by 16 code words which belong to 8 subframes, each having two code words. Each code word consists of 32 bits.

The code word can be defined either as an address code word (if the first bit is 0) or a message code word (if the first bit is 1). The address has an 18-bit address field, followed by 3 bits that allow for 8 different paging sources, and 10 error-correction bits per code word. The number of possible addresses that can be transmitted in one frame is $2^{18} = 262{,}144$. For a large paging system, the number of possible addresses can be increased. Each pager is programmed with a pre-defined 3-bit code. This code indicates the exact position of its address out of the eight frames transmitted together. The possible number of addresses now increases to $2^{21} = 2{,}097{,}152$. Code words in a batch that are not used for addresses are used for messages. There are 20 message bits per code word, and as many code words may be used as necessary to transmit the message.

EXAMPLE 10.2 Number of pages in POCSAG paging system

Suppose the POCSAG paging system is used with simple tone pagers, which require only an address field. If all the frames are used for addresses, how many pages could be transmitted by this system in 60 seconds if it operates at the POCSAG rate of 1200 bps? Assume that only one preamble is needed.

Solution

Transmission data rate of POCSAG = 1200 bps (given)

Time for transmission of pages = 60 seconds (given)

Step 1. *To find number of usable bits in one minute*

Total number of data bits in 60 seconds = $1200 \times 60 = 72{,}000$ bits

Number of bits used in the preamble = 576 bits (Standard)

Total number of usable bits in one minute = $72{,}000 - 576 = 71{,}424$ bits

Step 2. *To find number of bits in one batch*

As specified in POCSAG paging standard, there is one synchronising code word and sixteen address code words in each batch.

Number of bits in one synchronising code word = 32 bits

Number of bits in one address code word = 32 bits

Number of bits in sixteen address code words = $16 \times 32 = 512$ bits

Total number of bits in one batch = $32 + 512 = 544$ bits

Step 3. *To find number of batches in one minute*

Number of batches in one minute = (Total usable bits/minute)/(Bits/batch)

Number of batches in one minute = 71,424/544

Number of batches in one minute = 131.294 batches/minute

Step 4. *To determine number of pages transmitted in one minute*

Number of pager addresses per batch = 16 (Standard)

Number of pages that can be transmitted in one minute means number of different addresses per minute, which is given by

Number of pages transmitted per minute = batches/minute $\times$ addresses/batch

Number of pages transmitted per minute = 131.294×16

Number of pages transmitted per minute = 2100 pages

Paging systems such as FLEX, ReFLEX, and ERMES can support signaling data rate up to 6400 bps by using 4-level FSK modulation scheme. The Motorola FLEX system is the standard for one-way paging system. The FLEX system uses four-level FSK to allow operation at 1600 bps, 3200 bps, and 6400 bps.

The possible frequency deviations from the carrier frequency are ±800 Hz and ±2400 Hz. Two information bits can be transmitted per symbol, allowing a data rate of 6400 bps.

EXAMPLE 10.3 FLEX vs POCSAG paging system

Suppose a POCSAG paging system operating at 1200 bps is replaced with a FLEX paging system operating at 6400 bps. Compare the spectral efficiency of these two paging systems. Give an estimate of the increase in capacity.

Solution

The channel bandwidth in POCSAG system = 25 kHz (standard)

The channel bandwidth in FLEX system = 25 kHz (standard)

Step 1. *To determine spectral efficiency in POCSAG paging system*

Transmission data rate = 1200 bps (given)

The spectral efficiency = Data rate/Channel bandwidth

Or, Spectral efficiency = 1200 bps/25 kHz

Hence, the spectral efficiency = 0.048 bps/Hz

Step 2. *To determine spectral efficiency in FLEX paging system*

Transmission data rate = 6400 bps (given)

The spectral efficiency = Data rate/Channel bandwidth

Or, Spectral efficiency = 6400 bps/25 kHz

Hence, the spectral efficiency = 0.256 bps/Hz

Step 3. *To estimate an increase in capacity*

The approximate increase in capacity in a FLEX paging system over the POCSAG paging system is given by the ratio of their respective data rates, that is,

Capacity increase = 6400 bps / 1200 bps = 5.3 times

The increase in capacity is just an estimate because the two protocols differ in the amount of redundancy included in the protocols.

10.1.5 Two-Way Paging Systems

The Motorola ReFLEX system is the standard for two-way alphanumeric pagers. ReFLEX paging systems operate in the frequency ranges of 928 MHz–932 MHz and 940 MHz–941 MHz for the forward channel (base station to pager transceiver) and 896 MHz–902 MHz for the reverse channel (pager transceiver to base station). One forward paging channel can be carried in a 25-kHz channel, or three paging channels in one 50-kHz channel. The available forward data transmission rates are 1600 bps, 3200 bps, and 6400 bps. Reverse channel communication uses a 12.5-kHz channel and can support up to a 9600 bps data rate.

The two-way paging system is much more complex and expensive than the more common one-way paging system. Every pager needs a transceiver, with a power output of the order of one watt. The pager transceiver is able to acknowledge the receipt of pages without having to make a call using a landline telephone for response to the received page. It is also possible for a two-way paging system to employ a frequency reuse plan. Pagers can be located within a particular cell. Pager transceivers can respond to queries from the paging system.

Facts to Know!

A pager receiver includes four basic elements—a receiver (to receive and demodulate the paging signals, the receiver is tuned to the same radio frequency as the base station); a decoder (to decode the binary information); a control logic (to provide control for duplicate, locking and freeze messages); and a display to read messages.

10.1.6 Voice Paging

Voice paging basically allows a pager to function in a manner similar to a standard telephone answering machine. The Motorola InFLEXion system is the most popular voice-paging protocol standard. It uses AM SSB analog modulation scheme with analog compression technique to transmit voice messages from the base station to the pagers. Both upper and lower sidebands are used but each sideband constitutes a separate voice channel. In this way, the system allows two voice messages to be transmitted simultaneously on a channel that is 6.25 kHz wide. A pilot carrier is also transmitted for reliable demodulation at the receiver. Time-division multiplexing is also used so that many pagers can share a single voice channel. InFLEXion voice pagers normally allow a text (not voice) reply in a manner similar to two-way alphanumeric pagers. In that respect they are less flexible.

Facts to Know!

Paging systems have ultra-reliable coverage, high infrastructure, high complexity and high hardware cost.

10.2 CORDLESS TELEPHONE SYSTEMS

A cordless telephone system provide subscribers with mobility within a residence or small office by separating the telephone handset from the rest of the landline telephone and providing a wireless link between them. Cordless telephone systems are full-duplex wireless communication systems that use transceivers to connect a portable cordless handset to a dedicated base station called fixed wireless port. The fixed wireless port is connected to a dedicated telephone line with a specific telephone number on the public Switched Telephone Network (PSTN). Thus, a cordless telephone system provides wireless extension to the landline telephone for voice communications within a limited radio range, while retaining the standard features and call procedure as that of landline calls.

Figure 10.4 illustrates a block diagram of the basic cordless telephone system.

A cordless telephone or portable telephone is a telephone with a wireless handset that communicates via radio waves with a base station connected to a fixed telephone line, usually within a limited range of its base station (which has the handset cradle). The base station is on the subscriber premises, and attaches to the telephone network the same way a corded telephone does. The cordless handset is powered by a rechargeable battery, which is charged when the handset sits in its cradle.

Table 10.2 lists the most common cordless telephone system standards used in North America, Europe, and Japan.

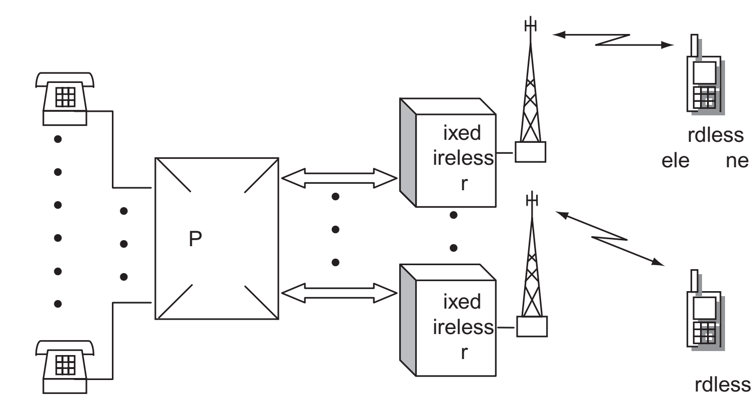

Fig. 10.4 *A basic cordless telephone system*

Table 10.2 *Major cordless systems standards*

Cordless systems standard	*requency band*	*Multiple access technique*	*Channel bandwidth*	*Modulation scheme*	*Region*
PWT	1.91–1.92 GHz	TDMA/ FDMA	1250 kHz	π/4-DQPSK	North America
PACS	1.85–1.99 GHz	TDMA/ FDMA	300 kHz	π/4-DQPSK	North America
CT2	864–868 MHz	FDMA	100 kHz	GFSK	Europe
DECT	1.88–1.90 GHz	TDMA	1728 kHz	GFSK	Europe
DCS-1800	1.71–1.88 GHz	TDMA	200 kHz	GMSK	Europe
PHS	1.895–1.907 GHz	TDMA	300 kHz	π/4-DQPSK	Japan

Facts to Know!

A cordless telephone or portable telephone is a telephone with a wireless handset. It communicates via radio waves with a base station which is on the subscriber premises, and attaches to the telephone network the same way a wired telephone does. The base station which also has the handset cradle, is connected to a fixed telephone line.

In first-generation cordless telephone systems, the portable cordless telephone unit communicates only to the dedicated base station or fixed wireless port. The distance between the fixed wireless port and the cordless handset is limited up to a few tens of metres. Therefore, cordless telephone systems operate exclusively as extension telephones to a transceiver connected to a subscriber line on the PSTN.

In North America, the PACS standard is mostly used inside office buildings as a wireless voice and data telephone system or radio local loop (cordless system). The European cordless telephone standards CT2 system makes use of microcells which cover distances up to 100 metres. The CT2 system use fixed wireless ports with omnidirectional antennas mounted on lamp posts or on the sides of buildings. The CT2 system uses the frequency-shift-keying modulation scheme along with a 32-kbps Adaptive Differential Pulse Code Modulation (ADPCM) speech coder for high-quality voice transmission. The DECT system accommodates voice as well as data transmissions for office and business telephone subscribers.

Modern cordless telephone systems are integrated with paging receivers so that a subscriber may first be paged and then response to the page can be given using the cordless telephone. These systems allow subscribers to use their handsets at many outdoor locations in urban regions. Typical second-generation base stations provide coverage ranges up to a few hundred metres. Cordless telephone systems provide the subscriber with limited radio range and mobility as it is usually not possible to maintain a call if the subscriber travels outside the range of the base station.

10.2.1 CT2 Cordless Telephony

CT2 is a cordless telephony standard that was used in the early 90s to provide short-range prototype-mobile phone service in some countries in Europe. It is one of the first digital cordless pan-European standard in the second-generation digital cordless technologies, providing improved speech quality, greater immunity from interference and a wider range of user facilities. CT2 is a digital FDMA system that uses TDD to share carrier frequencies between handsets and base stations.

Features of CT2 Systems

– Standardised on 864 MHz–868 MHz frequency band
– 100 kHz carrier channel spacing
– GFSK modulation technique
– 32 kbps ADPCM voice channel compression technique
– 500 frames/second (alternately base station and handset)

– 10mW maximum power output
– Up to 100-metre range

Facts to Know!

It is the base station on subscriber premises is what differentiates a cordless telephone from a cellular mobile telephone. However, current cordless telephone standards, such as PHS and DECT, have also implemented cell handover, data-transfer features and even limited international roaming. In these systems, base stations are maintained by a commercial cellular mobile network operator and users subscribe to the service.

CT2 is primarily a voice-only system. Although, like any minimally compressed voice system, users could deploy analog modems to transfer data. CT2 is a microcellular system fully capable of supporting hand-off, but it does not support forward hand-off, meaning that it has to drop its previous radio link before establishing the subsequent one, leading to a dropout in the call during the hand-off process.

Typical CT2 users have a base station and a handset, which can be connected to an existing home telephone system. Calls via the home base station would be routed via the home telephone line and in this configuration; the system was identical to a standard cordless phone. The user could receive incoming calls within the radio coverage range of its base station. Once out of coverage range, the CT2 user could find another network base station in the area, and make outgoing calls (but not receive calls) using the network base station. Base stations were located in a variety of public places. In this configuration, callers would be charged a per-minute rate higher than if they made calls from home.

The advantages to the user of a cordless telephone are that the call rates are generally lower than that of cellular phone-call charges, and that the same handset could be used at home and away from home. Compared to cellular services, the disadvantages are that many networks did not deliver incoming calls to the phones, and their operational areas are also very much limited.

10.2.2 DECT

Digital Enhanced Cordless Telecommunications (DECT), also known as Digital European Cordless Telephone, is an ETSI standard for digital portable phones or cordless home telephones, commonly used for domestic or corporate purposes. In Europe, the allocated frequency band is 1880 MHz–1900 MHz and outside Europe, the allocated frequency band is 1920 MHz–1930 MHz. These channels are reserved exclusively for voice communication applications and therefore are less likely to experience interference from other wireless devices used in wireless networks.

The DECT standard fully specifies a means for a portable unit such as a cordless telephone to access a fixed telecoms network via radio. Connectivity to different types of fixed networks is done through a base station to terminate the radio link, and a gateway to connect calls to the fixed network. In most cases the base station connection is to the public switched telephone network or telephone jack, although connectivity with newer technologies such as Voiceover IP has become available.

Features of DECT Systems

– Multiple handsets to one base station and one phone line socket. The additional handsets do not require additional telephone sockets or additional transceivers. This allows several cordless telephones to be placed around the house, all operating from the same telephone jack. Additional handsets usually have a battery charger station instead of a base station.
– Interference-free wireless operation to around 100 metres outdoors. Operates clearly in common congested domestic radio traffic situations. For instance, generally immune to interference from Wi-Fi networks, Bluetooth technology, and other wireless devices.
– Intercom facility. Ability to make internal calls between handsets.

– An extended range between the telephone and base station, allowing greater physical distance between the two devices.
– Extended battery talk-time up to 24 hours.

DECT Protocol Architecture Figure 10.5 depicts the protocol architecture that supports DECT operation. The DECT physical layer uses FDMA/TDMA multiple access technique with TDD. DECT operates in the 1880–1900 MHz band and defines 10 channels from 1881.792 MHz to 1897.344 MHz with a channel spacing of 1728 kHz. Each base station frame provides 12 duplex speech channels with each time slot occupying any of the RF channels. The modulation method is GFSK with a nominal deviation of 288 kHz. The data rate is 1.152 Mbps and the bandwidth efficiency is 2. The maximum allowed power for portable equipment as well as base stations is 250 mW. A portable device radiates an average of about 10 mW during a call as it is only using one of 24 time slots to transmit.

The DECT media access control layer is the layer which selects the physical channel and provides connection-oriented, connectionless and broadcast services to the higher layers. It also provides encryption services with the DECT Standard Cipher (DSC), using a 35-bit initialisation vector and encrypting the voice stream with 64-bit encryption. The DECT data link layer uses LAPC (Link Access Protocol Control), a specially designed variant of the ISDN data link protocol, called LAPD, based on HDLC.

The DECT network layer contains protocol entities such as call control, mobility management, call-independent supplementary services, connection-oriented message service, and connectionless message service. The call-control protocol is derived from the ISDN DSS1 protocol. The architecture presumes that linkages between the operation of the mobility management and call control is designed into the interworking unit that connects the DECT access network to mobility-enabled fixed network. By keeping the entities separate, the handset is capable of responding to any combination of entity traffic, and this creates great flexibility in fixed network design without breaking complete interoperability.

Applications of DECT Standard

– Domestic cordless telephony, using a single base station to connect one or more handsets to the public telecoms network.
– Enterprise premises cordless PABXs and wireless LANs, using many base stations for coverage. Calls continue as users move between different coverage cells, through hand-off.

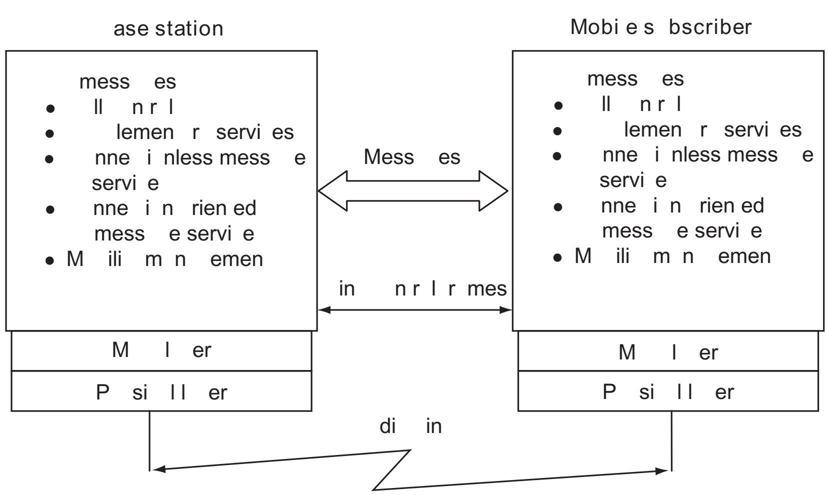

Fig. 10.5 *DECT protocol architecture*

– Public access, using large numbers of base stations to provide indoor or urban area coverage as part of a public telecoms network.
– Fixed wireless access by using directional antennas up to over 10 km range.

10.2.3 PHS

Personal Handy-phone System (PHS) was developed in Japan as a cordless telecommunication system operating within the frequency band of 1895 MHz to 1918.1 MHz. PHS supports seventy-seven carrier channels, each having a 300-kHz bandwidth. The system uses $\pi/4$-DQPSK digital modulation scheme at a channel data rate of 32 kbps on forward as well as reverse links. Out of a total of seventy-seven channels, thirty-seven channels are designated for home or office use in the 1895 MHz to 1906.1 MHz band, and the remaining forty channels are designated for public use in the 1906.1 MHz to 1918.1 MHz band. PHS cells are small in size, with transmission power of the base station up to 500 mW.

Facts to Know!

Unlike a wired telephone, a cordless telephone needs ac mains electricity to power the base station. The cordless handset is powered by a rechargeable battery, which is charged when the handset rests in its cradle.

PHS uses TDMA/TDD for its radio channel access method, and 32 kbps ADPCM with CRC error detection for its voice codec. Each TDMA frame is of 5 ms duration. Four full-duplex data channels are provided on each carrier using TDMA/TDD technique. The system uses dynamic channel assignment strategy. Base stations allocate channels based on the RF signal strength and interference levels observed at both the base station and cordless phone. When cordless phone subscribers are free, they lock on to dedicated control channels. A modern PHS phone can also support many value-added services such as high-speed wireless data/Internet connection at 64 kbps nominal, WWW access, e-mailing, text messaging and even colour image transfer. PHS technology is also used for providing a wireless local loop. A PHS base station has a compatibility with, and is often connected directly to an ISDN telephone exchange equipment.

PHS is a lightweight portable wireless telephone that functions as a cordless phone at home and as a mobile phone elsewhere. The Personal Handy-phone also handles voice, fax, and video signals. PHS is a cordless telephone with the capability to hand-off from one cell to another at pedestrian speed only because PHS is designed for microcell or indoor applications. Its typical range is in tens or at most hundreds of metres (or up to about 2 kilometres in line-of-sight conditions). This makes PHS suitable for dense urban areas, but impractical for rural areas, and the small cell size also makes it difficult to make calls from fast-moving vehicles.

The primary focus of the current PHS evolution is to develop packet data capabilities. A number of new data applications for PHS using small handheld PDAs are emerging. PHS handsets with implementations for PCMCIA are also available and will greatly enhance connection of PCs to the PHS service.

10.2.4 PACS

The Personal Access Communication System (PACS) radio interface is the leading low-tier cordless telephony standards for residential and office applications in North America. This radio interface was originally conceived to serve fixed-distribution and pedestrian applications; there has been significant development in extending this technology into high-mobility environments. In such environments, rapid channel variations and the effects of time-delay spread significantly degrade the performance of the PACS radio link. The system integrates all forms of wireless local loop communications to support voice, data, and video images for microcell and indoor applications.

The system operates in 1850 MHz–1910 MHz for uplink and 1930 MHz–1990 MHz for downlink transmissions. A large number of channels, each having a bandwidth of 300 kHz, may be frequency division multiplexed with 80 MHz duplex spacing between a pair of channels, or time division multiplexed with eight time slots provided in 2.5 ms TDMA frame on each carrier channel, with the time offset of exactly two time slots between the forward and reverse time slot for each user. PACS uses $\pi/4$-DQPSK modulation technique with a raised cosine filter having a roll-off factor of 0.5. The system uses ADPCM encoding scheme at the rate of 32 kbps. The PACS subscriber units employ adaptive power control to optimise the transmitted power so as to reduce cochannel interference as well as to save battery power.

Facts to Know!

Most cordless telephones, irrespective of frequency band or transmission method used, will hardly ever exactly match the voice quality of a high-quality wired telephone attached to a good telephone line. This constraint is caused by a number of issues such as side tone (hearing one's own voice echoed in the receiver speaker), a noticeable amount of constant background noise (This is noise inherent to cordless telephone system rather than interference from external source), and frequency response characteristics.

The PACS radio interface standards supports a variety of mobile data and messaging services which include circuit-mode data service, packet-mode data service, interleaved speech/data service, and individual messaging service. The *circuit-mode data service* is a nontransparent data service that uses the link access protocol for radio to provide data flow control, error control, and data ciphering. The *packet-mode data servic*e uses the data sense multiple access protocol over the radio interface for contention resolution. In the interleaved speech/data service, both speech and data information can be transmitted using a single 32-kbps time slot. The individual messaging service can support maximum message lengths of 16 Mbytes, with a provision for message encryption.

An Interworking Function (IWF) is provided to PACS to provide end-to-end flow of data. It also converts digital data on the radio interface to a form suitable for transmission over the intermediate and destination networks. The IWF also provides special functions such as protocol conversion and adaptation, address translation and management, and hand-off management.

10.2.5 Comparison of Cordless Telephone Standards

Table 10.3 summarises the air-interface specification parameters of major digital cordless standards,

Table 10.3 *Digital cordless air interface parameters*

Parameter	*CT2*	*DECT*	*PHS*	*PACS*
Frequency band (MHz)	864–868	1880–1900	1895–1918	1850–1910 (U) 1930–1990 (D)
Multiple access	FDMA	FDMA/TDMA	FDMA/TDMA	FDMA/TDMA
Duplexing	TDD	TDD	TDD	FDD or TDD
Carrier spacing (kHz)	100	1728	300	300
Number of carriers	40	10	77	400 or 32
Channels/carrier	1	12	4	8
Modulation	GFSK	GFSK	$\pi/4$ DQPSK	$\pi/4$ QPSK
Speech coding	32 kbps	32 kbps	32 kbps	32 kbps
Channel bit rate (kbps)	72	1152	384	384
Frame duration (ms)	2	10	5	2.5 or 2.0
Peak handset Tx power (mW)	10	250	80	100
Average handset Tx power (mW)	5	10	10	25
Region of operation	Europe	Europe	Japan	US

10.3 WIRELESS LOCAL LOOP

Traditionally, the provision of voice and data communication services to the end subscribers, over the subscriber loop or local loop, has been provided by wired systems. For residential subscribers, twisted pair cable has been and continues to be the standard means of connection. For business and government subscribers, twisted pair cable, coaxial cable, and optical fiber cable are in use. As subscribers have demanded greater capacity and data-rate capabilities, particularly to support Internet use, traditional twisted-pair cable technology has become inadequate.

Telecommunications providers have developed a number of technologies to meet the need, including Integrated Services Digital Network (ISDN), and a family of digital subscriber loop technologies, known as xDSL. In addition, cable operators have introduced two-way high-speed service using cable modem technology. Thus, wired technologies are responding to the need for reliable, high-speed access by residential, business, and government subscribers.

Facts to Know!

WLL is a system that connects subscribers to the Public Switched Telephone Network (PSTN) using radio signals as a substitute for copper for all or part of the connection between the subscriber and the switch. This includes cordless access systems, proprietary fixed radio access, and fixed cellular systems.

However, there are competing wireless technologies for end subscriber access. These approaches are generally referred to as Wireless Local Loop (WLL), or fixed wireless access. WLL alternatives are narrowband, which offer a replacement for existing telephony services, and broadband, which provide high-speed two-way voice and data service. Only limited-number of WLL systems is usually found in remote areas where provision for fixed-line usage is difficult.

Modern WLL systems use CDMA technology. A WLL service provider serves one or more cells. Each cell includes a base station antenna mounted on top of a tall building or tower. Individual subscribers have a fixed antenna mounted on a building or a high pole that has an unobstructed line of sight to the base-station antenna. From the base station, there is a wired or wireless link to a switching centre. The switching centre is typically a telephone exchange, which provides connections to the local and long-distance landline telephone networks. An Internet Service Provider (ISP) may be connected to the switching centre by a high-speed data link.

The WLL has a number of advantages over a wired approach to subscriber loop support, such as the following.

(a) WLL systems are less expensive than wired systems.
(b) WLL systems typically can be installed rapidly.
(c) Subscriber radio units are installed only for those willing subscribers who want the service at a given time. With a wired system, typically a cable is laid out in anticipation of serving every potential subscriber in a local area.
(d) A large geographical area is still not covered with landline telephone service or not covered for high-speed data transmission applications.
(e) WLL has become cost-competitive with wired local loops, and new requirements are preferred to be met with WLL approach.
(f) Cellular systems are quite expensive and do not provide sufficient facilities to act as a realistic alternative to broadband WLL.
(g) A major advantage of WLL over the cellular mobile system is that the fixed subscriber can use a directional antenna pointed at the base-station antenna, providing improved signal quality in both directions.

WLL has been allocated a frequency band of 2 GHz to 40 GHz, especially the unused frequency bands available above 25 GHz. Note that these frequencies are considerably higher than those used for cellular systems. At these frequencies, often referred to as millimetre wave frequencies, propagation characteristics are quite different from those in the MHz ranges. For example, free-space propagation path loss increases with the square of the frequency; thus signal losses are much higher in WLL than in the traditional microwave radio systems. Generally, above 10 GHz, attenuation effects due to rainfall and atmospheric or gaseous absorption are large. Multipath losses due to reflection, diffraction and scattering can be quite high because of much smaller wavelength than the size of an obstacle. Because of these negative propagation characteristics, WLL systems can only serve cells of a limited radius, usually of the order of a few kilometres. Also, obstructions including dense foliage must be avoided along the line-of-sight radio path.

WLL applications include Local Multipoint Distribution Service (LMDS) and Multichannel Multipoint Distribution Service (MMDS). LMDS is a relatively new WLL service capable of providing video, telephony, and high data rates of the order of several Mbps, within the short range from the base station, requiring a relatively large number of base stations to service a given area. MMDS signals operate at lower band of millimetre range and can operate in considerably larger cells within a radius of 50 km, but subscriber antennas must be in the line of sight.

MMDS can be used to support two-way services as well as an alternative for broadband services such as Internet access. But it offers much less bandwidth than LMDS. LMDS is useful for larger companies requiring greater bandwidths whereas MMDS is likely to be used by residential subscribers and small businesses.

Facts to Know!

MMDS has been used to compete with cable TV providers and to provide service in rural areas not reached by broadcast TV or cable. For this reason, MMDS is also referred to as wireless cable.

10.4 LMDS

Local Multipoint Distribution System (LMDS) is the broadband wireless point-to-multipoint communication technology used to provide digital two-way voice, data, Internet, and video services. The system operates in the 20-GHz and above frequency spectrum. The propagation characteristics of signals in this frequency range limit the potential coverage area of a single cell-site up to 8 km. LMDS systems use cellular-like network architecture, though services provided are fixed, not mobile. In the United States, 1.3 MHz of bandwidth has been allocated for LMDS to deliver broadband services in a point-to-point or point-to-multipoint configuration to residential and commercial customers.

Facts to Know!

Local Multipoint Distribution Services (LMDS) is a fixed wireless technology that operates typically in the 28-GHz band and offers line-of-sight coverage over distances up to 3–5 kilometres. It can deliver data and telephony services to 80,000 customers from a single node.

10.4.1 LMDS Network Architecture

LMDS network architecture is configured either as point-to-multipoint wireless access systems, or point-to-point systems, or TV distribution systems. LMDS services are a combination of voice, video, and data. Therefore, both asynchronous transfer mode (ATM) and Internet Protocol (IP) transport methodologies are practical. The LMDS network architecture consists of primarily four parts: Network Operations Centre (NOC), fiber-based infrastructure, Base Station (BS), and Customer-Premises Configurations (CPE).

The *NOC* contains the network management system equipment that manages large regions of the customer network. Multiple NOCs can be interconnected. The *fiber-based infrastructure* typically consists of synchronous optical network optical carrier links; central-office equipment; ATM and IP switching systems; and interconnections with the Internet and PSTNs.

Base-station equipment includes the network interface for fiber termination; and microwave transmission and reception equipment typically located on the rooftop of a building or high on a pole. BSs may include local switching, in which subscribers connected to it can communicate with one another without entering the fiber infrastructure. This function implies that channel access management, registration, authentication, and billing occur locally within the base station. Alternatively, BS architecture provides connection to the fiber infrastructure, routing all traffic to terminate in ATM switches or call office equipment. The base-station digital equipment provide processing, multiplexing, demultiplexing, compression, error detection, encoding, decoding, routing, modulation, demodulation, and ATM switching functions.

Facts to Know!

LMDS is one solution for providing high-bandwidth services to homes and offices within the last mile of wireless connectivity, an area where cable or optical fiber may not be convenient or economical. Data transfer rates for LMDS can be achieved between 1.5 Gbps to 2 Gbps, but a more realistic value may average around 38 Mbps in downstream.

The *CPE* includes outdoor-mounted microwave and indoor digital equipment, and customer-premises interface functionality. For two-way data network applications, a transceiver is used to provide a return path for LMDS services. The antenna may be an integral part of the transceiver. The transceiver may be broadband or channelised. The antenna systems include microstrip design, parabolic, grid-parabolic reflectors, or horn antenna. The CPE may attach to the network using FDMA or TDMA or CDMA techniques. The customer premises interfaces will cover the complete range of DS–0 level, plain old telephone service, 10BaseT, unstructured DS–1, structured DS–1, frame relay, ATM25, serial ATM over T1, DS–3, OC–3, and OC–1. The customer premises locations include large enterprises, in which the microwave equipment is shared between many users, to mall locations and residences, in which single offices requiring 10BaseT and/or two POTS lines are connected.

10.4.2 LMDS Operation

The idealised circular coverage area around the cell-site is divided into 4, 8, 12, 16, or 24 sectors, deploying multiple-sector microwave systems antennas providing service area over a 90-, 45-, 30-, 22.5-, or 15-degree beamwidth. Alternative architectures include connecting the base-station indoor unit to multiple remote microwave transmission and reception systems with analog fiber interconnection. By using remote microwave equipment, there may be a reduced sectorisation requirement at each remote location.

Facts to Know!

Multipoint Microwave Distribution System (MMDS), also known as Multichannel Multi-point Distribution System and wireless cable, is another wireless broadband technology for Internet Access. MMDS is a line-of-sight service, so it does not work well around mountains, but it is suitable in rural areas, where copper wires are not available.

In the uplink, FDMA, TDMA, and CDMA access methods apply to the connection from the customer-premises site to the base station. In the downlink, from base station to customer premises, TDM scheme is used either to a specific user site (point-to-point connectivity) or multiple user sites (a point-to-multipoint system design). MPSK and m-ary QAM modulation methods are used in broadband wireless LMDS systems.

The LMDS systems at 28 GHz are most susceptible to atmospheric behaviour. Rainfall and foliage causes depolarisation of the signals, leading to decreased signal level and decreased interference isolation between adjacent sectors and adjacent cell-sites. The cell coverage distance will vary depending on the rainfall statistics and foliage height in the particular area. Cell sizes are strongly affected by the propagation environment including foliage, rainfall statistics, height of the cell-site transmit antenna, height of the customer-premises antenna, the local obstructions, terrain and topology details. The choice of modulation technique also affects the range. For example, for QPSK and 4–QAM, range may be limited to 10 km, whereas for 16–QAM and 64–QAM, the range is 5 km and 2.5 km respectively.

10.5 CELLULAR TELEPHONE SYSTEM

A cellular telephone system provides a wireless connection to the PSTN for any location of the mobile subscriber within the radio range of the system. Cellular systems accommodate a large number of mobile subscribers over a large geographic area, using a limited frequency spectrum. High capacity is achieved by limiting the coverage of each base-station transmitter to a small geographic area called a cell so that the same radio channels may be reused by another base station located some distance away.

Figure 10.6 shows a basic cellular system that consists of mobile stations, base stations and a Mobile Telephone Switching Office (MTSO).

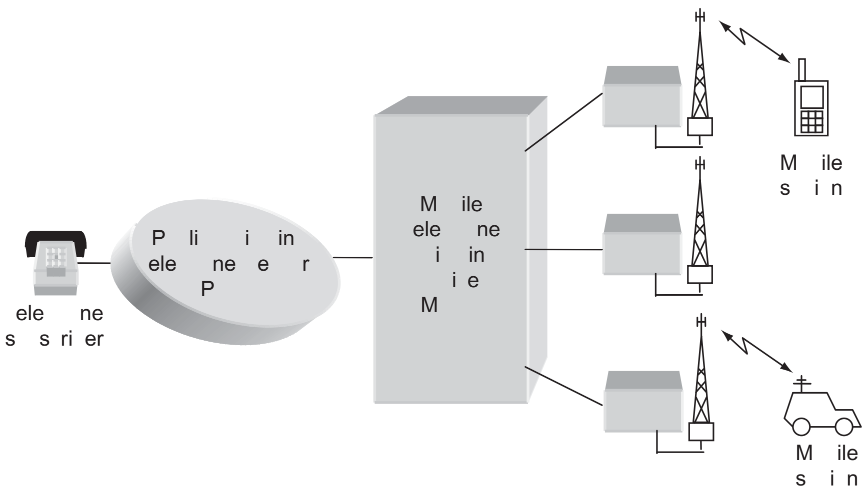

Fig. 10.6 *A cellular telephone system*

The mobile station consists of a transceiver, an antenna, and control circuitry. It may be mounted in a vehicle or used as a portable handheld unit. Each mobile station communicates via wireless link with one of the base transceiver stations and may be handed-off to any number of base stations throughout the duration of a call.

The base stations consist of several transmitters and receivers that simultaneously handle full-duplex communications and generally have towers which support several transmitting and receiving antennas. The base station serves as an intermediator between all mobile subscribers in the cell and connects the simultaneous mobile calls via data links (T1 or E1 lines) or microwave links to the MTSO. The MTSO is responsible for connecting all mobiles to the PSTN in a cellular system. It coordinates the activities of all of the base transceiver stations and connects the entire cellular system to the PSTN.

Cellular radio systems provide high-quality service that is often comparable to that of the landline telephone systems. A sophisticated automatic switching technique called a hand-off enables a call to proceed uninterrupted when the mobile subscriber moves from one cell to another cell.

10.6 ADVANCE MOBILE PHONE SERVICE (AMPS)

The AMPS is the most common first-generation analog cellular radio communication technology developed in North America. Typically, the AMPS system can serve a cell radius ranging from 2 km to 25 km, depending on the subscribers' density in the service area. The AMPS system uses medium-power (about 100 watts) transmitters at cell-sites and low-power (about 4 watts or less) transmitters at the mobile subscribers. The system is aimed to reduce blocking probability to about 2% during busy hours.

> **Facts to Know!**
>
> AMPS is an analog cellular system based on circuit-switched technique and essentially designed for providing voice communication among mobile subscribers. However, low-speed data such as FAX can be transmitted using modems at each end of the communication link.

The system is configured with a seven-cell frequency reuse pattern having provisions for 120 degree three-sector per cell using three directional antennas at the cell-site. The cell-splitting technique can also be deployed in order to increase user capacity if needed. The system operates in the 800-MHz frequency range, with a frequency spectrum of 40-MHz width (increased to 50 MHz in extended spectrum operation). The duplex spacing between the cell-site and mobile transmit frequencies is 45 MHz, which enables the use of simple duplexers to separate transmit and receive signals in the mobile subscriber phones. The system uses a channel spacing of 30-kHz and narrowband analog FM modulation, with frequency deviation of ± 12 kHz. The 30-kHz channel requires a signal-to-interference ratio of 18 dB to ensure satisfactory voice-quality performance to mobile subscribers. In transmitting data, the system uses Manchester coding scheme at the data rate of 10 kbps. Separate channels are used for transmitting signaling/control and voice. Since fewer control messages are exchanged between the cell-site and the mobile subscriber, a smaller number of control channels are reserved than voice channels. Typically, there is one control channel for every eight voice channels.

10.6.1 AMPS Air Interface

In North America, two 25-MHz bands are allocated to AMPS standard, one for transmission from the base station to the mobile subscriber (869 MHz–894 MHz), the other for transmission from the mobile to the base station (824 MHz–849 MHz), with a duplex spacing of 45 MHz. Practically, each of these frequency bands is divided in two equal parts so that two service providers can provide cellular services in the same service area for competition. It means that a service provider is allocated only 12.5 MHz spectrum in each direction for operation of its system. The channel spacing is 30 kHz, which allows a total of 416 channels per service provider. Out of the total 416 channels, 21 channels are used as control channels, and remaining 395 channels are used to carry voice calls. The control channels are basically data channels operating at 10 kbps data rate. The voice channels carry the analog speech signals using frequency modulation. Some control information is also sent on the voice channels in bursts as data.

The cell-site transmits to the mobile subscriber on a forward channel, while transmissions from mobile subscriber to cell-site use a reverse channel. Table 10.4 shows how these channel frequencies are divided between the service providers 'A' and 'B'.

Table 10.4 *AMPS cellular radio frequencies allocation*

Service provider	*Cell-site Tx frequencies (MHz)*	*Mobile Tx frequencies (MHz)*	*Type of channel*
A	869.040–879.360; 890.010–891.480	824.040–834.360 845.010–846.480	Voice
A	879.390–879.990	834.390–834.990	Control
B	880.650–889.980; 891.510–893.970	835.650–844.980 846.510–848.970	Voice
B	880.020–880.620	835.020–835.620	Control

These are the frequency spectrum allocated in extended AMPS spectrum. Note that the frequencies assigned to each service provider are not all contiguous because of the extended spectrum frequencies added subsequently in the system.

An individual cell doesn't use all these channels. Each cell has only one-seventh of the total number of channels assigned to a system since the AMPS system is configured with 7-cell clusters. With a seven-cell frequency reuse pattern, transmitters in the same cell are generally separated by about seven channels or

210 kHz. Each cell in a seven-cell pattern also has three of the 21 control channels. Contiguous frequencies are not used in the same cell in order to reduce adjacent channel interference. To meet receiver selectivity requirements, adjacent channels are not used in adjoining cells also. Therefore, transmitters in adjacent cells are separated in frequency by at least three channels or by 90 kHz.

The mobile subscribers access the system using frequency-division multiple access technique. Simultaneous multiple transmissions are separated in the frequency domain, that is, each channel is allocated a carrier frequency and channel bandwidth within the total allocated frequency spectrum. Simultaneous transmissions from multiple subscribers can occur at the same time without interfering with one another because their transmissions are on different channels and occupy different frequency bands. Mobile subscribers are assigned a pair of voice channels (forward and reverse) for the duration of their call to enable full-duplex communication. Once a voice channel is assigned to a mobile subscriber, no other mobile subscriber can be assigned the same channel in the same cell throughout the duration of the call.

EXAMPLE 10.4 | Air-Interface specifications of AMPS

List the major air-interface specifications of AMPS analog cellular standard.

Solution

Cell-site frequency range	Tx: 869–894 MHz; Rx: 824–849 MHz
Mobile frequency range	Tx: 824–849 MHz; Rx: 869–894 MHz
RF spectrum bandwidth	25 MHz each
Duplexing technique	Frequency-Division Duplex (FDD)
Duplex separation	45 MHz
Multiple access scheme	FDMA
RF channel bandwidth	30 kHz
Total number of full-duplex RF channels	832
Number of full-duplex voice channels	790
Number of full-duplex control channels	42
Voice-channels per RF channel	1
Voice-channel modulation scheme	FM with 12-kHz peak deviation
Control channel modulation scheme	FSK with 8-kHz peak deviation
Data transmission rate	10 kbps
Spectral efficiency	0.33 bps/Hz
Error-control coding scheme	BCH (48, 36, 5) and (40, 28, 5)

EXAMPLE 10.5 | AMPS frequency spectrum

Give a suitable illustration of original Advanced Mobile Phone Service (AMPS) frequency spectrum.

Solution

The original AMPS frequency spectrum = 825 MHz–845 MHz (Uplink); 870 MHz–890 MHz.(Dowlink)
The original AMPS spectrum bandwidth = 20 MHz each in UL and DL
Duplex spacing = 45 MHz
Designation of the system A channels = 1 to 333
Designation of the system B channels = 334 to 666

The original AMPS frequency spectrum including designated channel numbers for forward channels (downlink) is illustrated in Fig. 10.7, and the original AMPS frequency spectrum including designated channel numbers for reverse channels (uplink) is illustrated in Fig. 10.8.

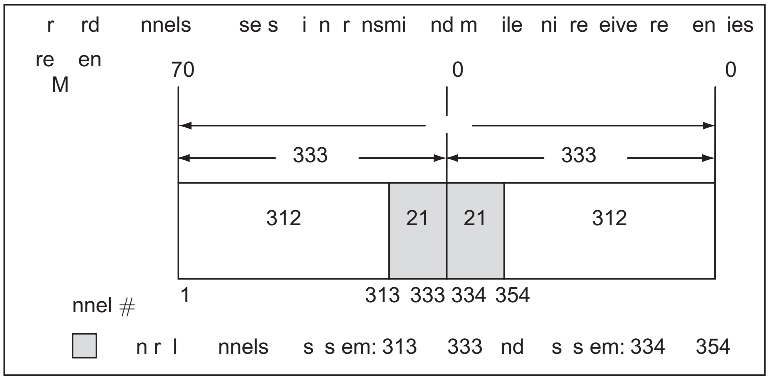

Fig. 10.7 *Original AMPS frequency spectrum—forward channels*

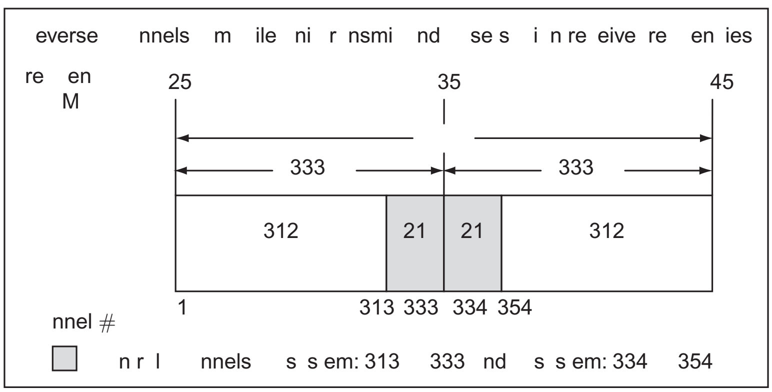

Fig. 10.8 *Original AMPS frequency spectrum—reverse channels*

For mobile subscriber units, Channel 1 has a transmit frequency of 825.03 MHz, and Channel 666 has a transmit frequency of 844.98 MHz.

For base stations, Channel 1 has a transmit frequency of 870.03 MHz, and Channel 666 has a transmit frequency of 889.98 MHz.

The receive frequencies for mobile subscriber units and base stations are, of course, just the opposite to their transmit frequencies respectively.

The forward channel transmit centre frequency, f_{tb} corresponding to a specific channel number, N can be calculated using the following expression:

$$f_{tb} = 870.0 + 0.030\,N \qquad \text{for } 1 \le N \le 799 \tag{10.1a}$$

$$f_{tb} = 870.0 + 0.030\,(N - 1023) \qquad \text{for } 991 \le N \le 1023 \tag{10.1b}$$

The reverse channel transmit centre frequency, f_{tm} corresponding to a specific channel number, N can be calculated using the following expression:

$$f_{tm} = 825.0 + 0.030\,N \qquad \text{for } 1 \le N \le 799 \tag{10.2}$$

$$f_{tm} = 825.0 + 0.030\,(N - 1023) \qquad \text{for } 991 \le N \le 1023 \tag{10.3}$$

The mobile unit's receive carrier frequency, is obtained by simply adding 45 MHz to the transmit frequency, that is,

$$f_{rm}\,(\text{MHz}) = f_{tm}\,(\text{MHz}) + 45\ (\text{MHz}) \tag{10.4}$$

The base station's transmit frequency for any channel is simply the mobile unit's receive frequency, and the base station's receive frequency is simply the mobile unit's transmit frequency, that is,

$$f_{tb} = f_{rm} \quad \text{and} \quad f_{rb} = f_{tm} \tag{10.5}$$

where f_{tb} = transmit carrier frequency (MHz) of the base station

f_{rb} = receive carrier frequency (MHz) of the base station

The extended AMPS frequency spectrum is 25 MHz each in the forward channel and the reverse channel instead of the original 20 MHz each, with a duplex spacing of 45 MHz being the same. The extended AMPS frequency spectrum is 824 MHz–849 MHz for the reverse channel and 869 MHz–894 MHz for forward channel. Figures 10.9 and 10.10 shows the extended AMPS frequency spectrum.

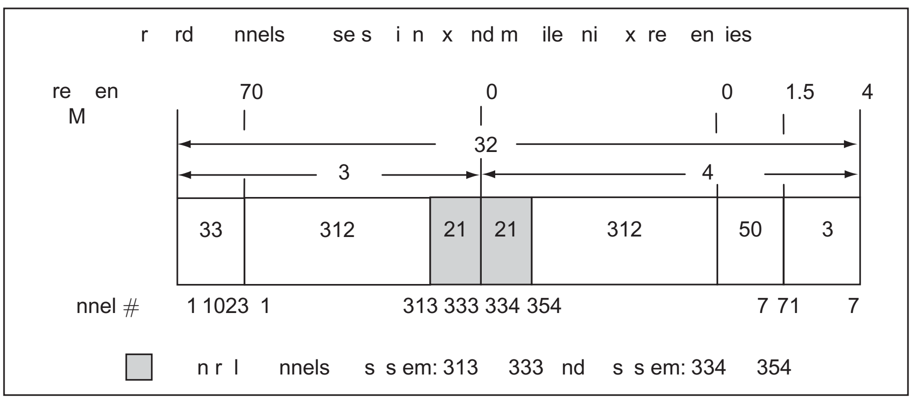

Fig. 10.9 *Extended AMPS frequency spectrum—forward channels*

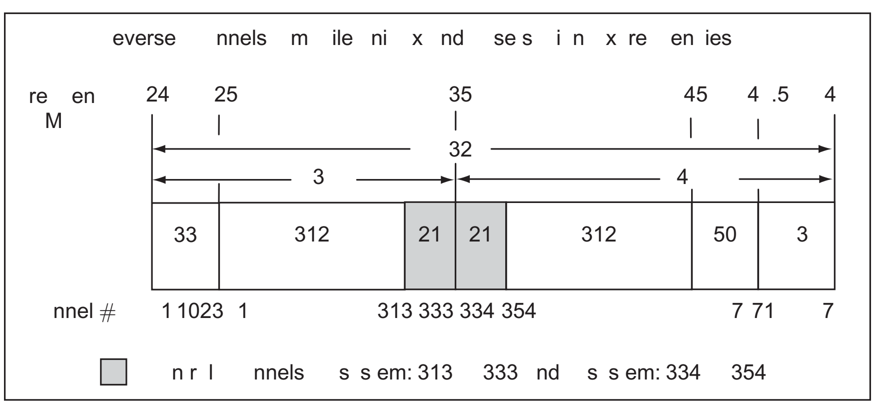

Fig. 10.10 *Extended AMPS frequency spectrum—reverse channels*

It may be noted that 33 channels out of 166 new channels are added below the original frequency spectrum and that the remaining 133 channels are added above the original frequency spectrum.

EXAMPLE 10.6 Number of channels in US AMPS System

A US AMPS cellular service provider is allocated 12.5 MHz for each simplex band. If the guard band is 10 kHz on either end, find the total number of channels available in the system.

Solution

Allocated simplex RF bandwidth, $B_t = 12.5$ MHz (given)

Guard band on one end, $B_g = 10$ kHz (given)

Step 1. *To find the usable simplex RF bandwidth.*

Guard band on both ends, $B_{2g} = 2 \times B_g$

Therefore, $B_{2g} = 2 \times 10 \text{ kHz} = 20 \text{ kHz}$

Available simplex RF bandwidth = 12,500 kHz – 20 kHz

Hence, available simplex RF bandwidth = 12,480 kHz

Step 2. *To find total number of channels in the system, N*

Channel bandwidth in AMPS, $B_c = 30$ kHz (AMPS Standard)

The total number of channels available in the system is given as

N = available simplex RF bandwidth/Channel bandwidth

Hence, $N = 12{,}480/30 = 416$ channels

10.6.2 Control and Voice Channels

The AMPS frequency channels are divided into two basic sets of channels. One set of channels is dedicated for exchanging signaling and control information between mobile subscriber units and base stations and is appropriately termed as control channels. *Control channels* are used exclusively to carry service messages, and cannot carry voice information. Since AMPS mobile subscriber phones contain only one receiver and one transmitter, they cannot receive both a voice channel and a control channel at the same time. Therefore, any control messages that have to be sent during on-going voice communication must use the voice channel. This is done employing in-channel out-of-band signaling consisting of tones above the voice frequency range of 300 Hz to 3400 Hz, as well as with blank-and-burst signaling. During blank-and-burst signaling, the voice signal is muted for about 100 milliseconds while data is sent over the voice channel.

Facts to Know!

To reduce the probability of errors during transmission, the control channel sends each message five times and also uses Hamming error-correction codes. This increases the robustness of the control system but reduces the actual data throughput to 1200 bps only.

Digital signals on the control channel and those sent during blank-and-burst signaling on the voice channel are sent using FSK signaling scheme with ±8 kHz deviation at a channel data rate of 10 kbps. The data is transmitted using Manchester code without any encryption technique.

The base stations continuously transmit FSK data on the forward control channels so that idle mobile subscribers can maintain link with the nearest base station regardless of their location. Forward control channels from base stations may contain overhead data, mobile station control information, or control file information. A mobile subscriber's unit must be locked on to a forward control channel before it can originate or receive calls. Each base station uses a forward control channel to page mobile subscriber units simultaneously so as to alert them of the possibility of incoming calls and to move established calls to a free voice channel.

One of the main functions of the control channels is to allocate voice channels to mobile subscribers. When a calling mobile subscriber places a called subscribers' phone number, it scans all the control channel frequencies to find the strongest control channel usually associated with the nearest cell-site. The mobile subscriber transmits on its reverse control channel, and once the call has been set up, the cell-site assigns it a non-interfering available free voice channel. While the voice communication continues, the adjacent cell-sites monitor the signal strength from the active mobile subscriber. When the received signal strength is greater in one of the adjacent cells, the system transfers the call to that cell, that is, hand-off takes place with a change in frequency channel under control of the system.

A similar procedure takes place for incoming calls (calls from cell-site to mobile subscriber). The mobile subscriber periodically identifies itself to the system automatically whenever it is switched on. Thus, the system usually knows its location. Paging signals are transmitted by the base station periodically on the forward control channel and the mobile subscriber responds on the reverse control channel. This process enables the system to locate the mobile subscriber more precisely.

The major functions and formats of various channels used by the AMPS system are described below.

(a) Forward Control Channel The forward control channel is primarily used by the base station to broadcast, page and locate the mobile subscriber units using the control information. The types of messages transmitted include the mobile subscriber control message and the overhead message sequence. Mobile subscriber control messages command mobile subscriber units to do a particular task such as initial voice channel designation, directed retry, alert, and change power. The overhead message sequence contains system parameter overhead, and control filler messages. Therefore, forward control channel data formats consist of three discrete information streams: the busy-idle stream, stream A, and stream B. These three data streams are multiplexed together.

Figure 10.11 shows the format for an AMPS forward control channel.

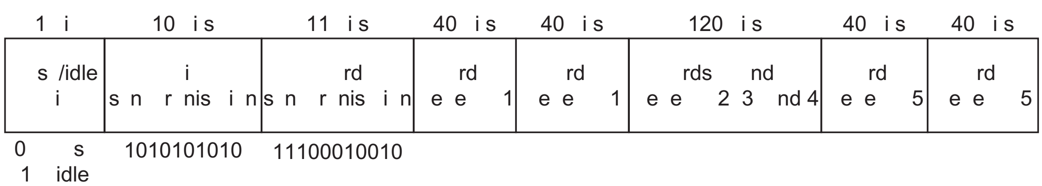

Fig. 10.11 *AMPS forward control channel format*

The busy-idle data stream contains busy-idle bits, which are used to indicate the current status of the forward control channel (0 = busy and 1 = idle). It is followed by a 10-bit dotting scheme (a sequence of alternating 1s and 0s) for bit synchronisation, and an 11-bit word synchronisation with a unique bit sequence. This enables the mobile receiver to acquire synchronisation instantly. The sync word is immediately followed by the message data repeated five times to increase reliability. Each frame contains two words of data – word A and word B. Each word contains 28 data bits and is encoded using 12 check bits BCH error-correction scheme to counter the effects of fading. The received message is taken as correct if three of the five words are received identical. In addition, each frame provides information about the idle or busy status of the corresponding reverse control channel frame through the busy/idle bits that are inserted every tenth bit in the frame. The total frame size is 463 bits. At the 10-kbps signaling rate, the data rate (excluding overhead) is about 1200 bps. The forward cnotrol channel messages include paging messages and frequency assignment messages.

(b) Reverse Control Channel The reverse control channel is used by the mobile subscriber in response to the page sent by the cell-site, or to respond to queries. It follows standard slotted ALOHA packet radio protocol. The control data are transmitted at a 10-kbps data rate and include registration requests, page responses, and access requests.

Figure 10.12 shows the format for the reverse control channel that is transmitted from the mobile subscriber unit to the base station.

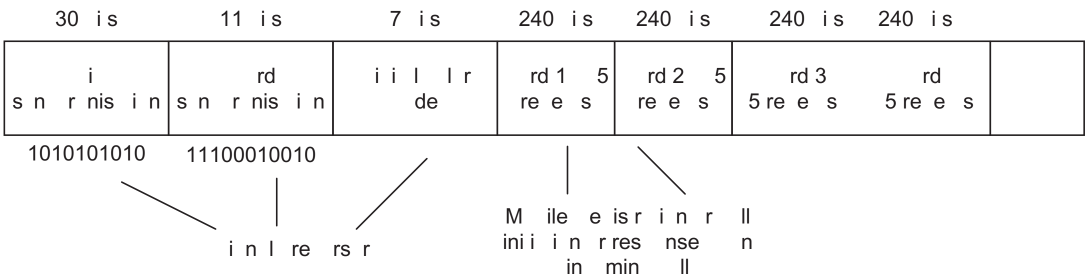

Fig. 10.12 AMPS reverse control channel format

All reverse control channel messages begin with the 48-bit signal precursor comprising of a 30-bit synchronisation sequence for bit sync field of alternating ones and zeros, an 11-bit fixed word synchronisation field (11100010010), and the 7-bit coded digital color code. The digital color code is added so that the control channel is not confused with a control channel from a cochannel that is reusing the same frequency. It is a unique identifier of a base station and acts as a destination address for a reverse control channel frame. The mobile subscriber unit decodes the base station's digital colour code and then returns a coded version of it. This serves as verification to locking of the mobile subscriber to the desired incoming control signal.

The precursor field is followed by one to n words of data. Each message word contains 36 data bits and is encoded using a shortened version of BCH block code $(n, k, t) = (63, 51, 5)$. In this shortened encoded version, 12 check bits are added to the 36 data bits to form a 48-bit word. To further increase reliability, each word is repeated five times for a total word size of 240 bits in the same frame. A majority logic is used to recover the word at the base station.

(c) Forward Voice Channel The forward voice channel is used for one-to-one communication link between the base station and each individual mobile subscriber unit. Forward voice channels carry both voice using FM and digital signaling information using binary FSK. A limited number of messages can be sent on a forward voice channel. The forward voice channel supports two different tones. The continuous supervisory audio tones are used by the base stations to transmit broadcast control signals to check for the active mobile subscriber units in the service area. The discontinuous data stream is used by the base station to send orders or new voice channel assignments to the mobile subscriber unit. When transmitting digital signaling information, voice transmissions are inhibited. This is called blank and burst which means that the voice is blanked for a short duration, and the signaling data is transmitted in a short burst. The bit rate of the digital information is 10 kbps. Figure 10.13 shows the voice-channel signaling format for a forward voice channel.

The digital signaling sequence begins with a 101-bit dotting sequence that alerts the receiver to receive digital information. After the dotting sequence, an 11-bit word synchronisation bit pattern is sent to indicate the start of the message. The forward channel uses 40-bit words, and are repeated 11 times to ensure its integrity.

(d) Reverse Voice Channel A reverse voice channel is used for one-to-one communication link between the individual mobile subscriber unit and the base station during calls in progress and is assigned by the base station to a mobile subscriber unit for its exclusive use. The digital signaling sequence begins with a 101-bit dotting sequence that alerts the base station receiver to receive digital information from the mobile subscriber unit. After the dotting sequence, a synchronisation word is sent to indicate the start of the message. On the reverse voice channel, digital signaling messages are repeated 5 times only. The reverse channel uses 48-bit words.

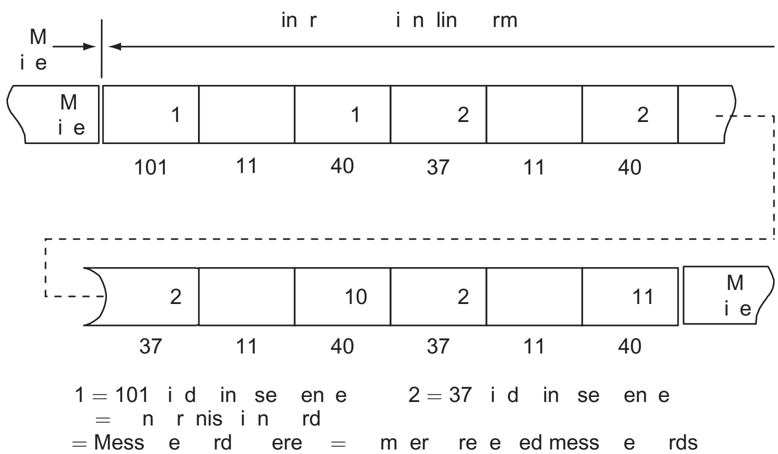

Fig. 10.13 *AMPS forward voice channel format*

Figure 10.14 shows the format for the reverse voice channel.

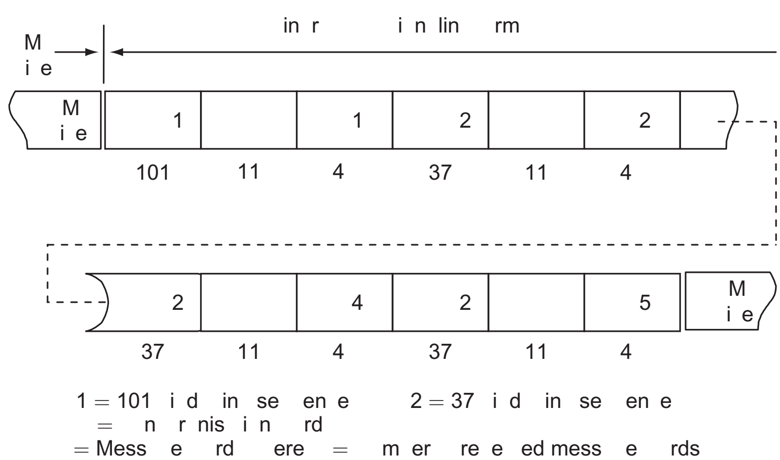

Fig. 10.14 *AMPS reverse voice channel format*

10.6.3 Supervisory Signaling Techniques

When voice channels are busy, there are additional signaling techniques which are used to maintain supervision between the base station and subscriber unit. The supervisory signals are the Supervisory Audio Tone (SAT), the Signaling Tone (ST), and blank-and-burst signaling. The SAT always exists during the use of any voice channel. The AMPS system use three SAT signals at frequencies of either 5970 Hz, 6000 Hz, or 6030 Hz.

A given base station will constantly transmit one of the three SAT tones on each busy voice channel. The SAT signal is superimposed on the voice signal on both the forward and reverse channels and is hardly audible to a mobile subscriber. The particular frequency of the SAT signal denotes the particular base station location for a given voice channel and is assigned by the MTSO for each call.

When a call is established and a voice channel is assigned, the base station starts transmitting the SAT signal on the forward voice channel. The mobile subscriber unit monitors the SAT transmitted by the base

Facts to Know!

A fully equipped cellular system might have as many as three cochannel base stations in the service area, the SAT enables the mobile subscriber unit and the base station to know which of the three cochannel base stations is handling the voice communication.

station on the forward voice channel, it demodulates it and then retransmits back the same tone to the base station on the reverse voice channel. The detection and rebroadcast of SAT must be performed at least every 250 milliseconds at the mobile subscriber unit. This handshake is required by the system to dedicate a voice channel. If the SAT is not present or improperly detected within one second, both the base station and mobile subscriber unit stop transmission on assigned voice channels.

During blank-and-burst data transmissions, the mobile subscriber unit halts transmission of SAT on the reverse channel. Dropped or prematurely terminated voice calls can often be traced to interference or incorrect detection of the SAT either at the base station or the mobile subscriber unit.

When a mobile subscriber terminates a call or turns the mobile phone off during a call, an ST tone is automatically sent by it. The signaling tone (ST) is a 10 kbps data burst which is usually used for termination of on-going voice communication by the mobile subscriber. It is basically an end-of-call message consisting of alternating 1s and 0s. It is transmitted on the reverse voice channel by the mobile subscriber unit for 200 milliseconds. The ST tone is sent simultaneously with the SAT. The ST signal alerts the base station that the mobile subscriber has terminated the on-going call. This allows the base station and the MTSO to know that the call was terminated deliberately by the mobile subscriber.

Wideband data signaling is used on a voice channel to provide brief data messages that allow the mobile subscriber and the base station to adjust transmitted power by the mobile subscriber unit or initiate a handoff request. The system provides supervisory signals during voice channel transmissions that allow each base station and its subscribers to confirm that they are properly connected during a call. The wideband data is provided by using a blank-and-burst signaling for less than 100 milliseconds, where the voice channel audio is muted and replaced with it. The wideband signaling data is sent at 10 kbps using FSK signaling.

10.6.4 AMPS Identification Codes

The AMPS system has an identifying number called the system identification number (SID). It is a unique 15-bit binary number assigned to the system. The SID is stored in all base stations. Every mobile unit knows the SID of the system with whom it is subscribed to. All registered mobile subscribers in the service area use it. A mobile subscriber unit transmits this number before any call can be handled. The SID serves as a check to determine if a particular mobile subscriber unit is registered in its home cell-site or it is a roaming subscriber. Whenever a mobile subscriber unit initialises, it compares its known SID with the SID broadcasted by the nearest base station. If the SIDs are the same, it means that the mobile subscriber unit is communicating with its home system, otherwise it is roaming.

Each base station has a unique two-bit Digital Colour Code (DCC) which is assigned by its service provider. The DCC is one of four binary codes (00, 01, 10, or 11). The DCC enables the mobile subscriber units to distinguish its serving base station from a neighbouring base station. Neighbouring base stations transmit different DCCs. When the mobile detects a change in DCC without a change in frequency, it is an indication that cochannel interference is being received from another cell-site.

Each AMPS mobile subscriber unit has several unique identification numbers—an Electronic Serial Number (ESN), the Mobile Identification Number (MIN), and Station Class Mark (SCM). The mobile equipment has a numeric assignment module in its read-only memory which contains the MIN assigned by the service provider, and the ESN assigned by its manufacturer. The ESN is a unique 32-bit binary number permanently assigned to it. ESN is not supposed to be changeable without rendering the mobile phone inoperable for security reasons. However, there is a provision to store it in an EPROM that can be reprogrammed or replaced by authorised persons only.

The MIN is the 10-digit directory mobile phone number. The MIN comprises of a three-digit area code, a three-digit prefix (mobile exchange number), and a four-digit mobile subscriber (extension) number. The three-digit mobile exchange number is assigned to the service provider. If a mobile subscriber changes service from one service provider to another service provider, the mobile subscriber must be assigned a new cellular mobile phone number. This number is translated into a 34-bit binary number according to a simple algorithm. If a registered MIN appears with the improper ESN, the system will not allow to provide any service. The combination of the ESN and the MIN enables the system to ensure proper billing and to check for fraudulent use.

Facts to Know!

The service providers assign a Supervisory Audio Tone (SAT) to each of their base stations. The SAT helps the mobile subscriber units to distinguish one base station from a neighbouring base station. The SAT is one of three analog frequencies (5970 Hz, 6000 Hz, or 6030 Hz). Neighbouring base stations transmit different SAT frequencies.

The third identification code is the four-bit Station Class Mark (SCM), which indicates whether the mobile subscriber unit has access to all 832 channels or only 666 channels. The SCM also specifies the maximum radiated transmitted power level for the mobile subscriber unit. There are three power classes: Class I, Class II, and Class III, corresponding to mobile phones permanently installed in a vehicle, transportable mobile phones (carried by a person in a bag), and handheld (portable) mobile phones respectively. The maximum power levels, specified as ERP (effective radiated power with respect to a half-wave dipole) of three classes of AMPS mobile phones, are given in Table 10.5.

Table 10.5 *Classes of AMPS mobile phones*

Type of mobile phone	*Type of class*	*Maximum transmitter power level (maximum ERP)*
Vehicle-mounted mobile phone	Class I	+6 dBW (4W)
Transportable mobile phone	Class II	+2 dBW (1.6W)
Handheld (portable) mobile phone	Class III	–2 dBW (600 mW)

In order to minimise the interference and unnecessary radiations, the mobile transmitter power is controlled by the cell-site. The actual transmitted power level is adjusted in 4 dB steps with the lowest power level being –22 dBW by control signals from the cell-site. The mobile transmitter must transmit at within 3 dB of the correct power level within 2 milliseconds of switching on the mobile equipment and must reduce its output to −60 dBm ERP or less within 2 milliseconds of being switched off.

EXAMPLE 10.7 Power levels for AMPS mobile phones

A 3-bit Mobile Attenuation Code (MAC) is transmitted from the cell-site to adjust the power of the Class I, Class II, and Class III AMPS phones. Tabulate the ERP levels in dBW, corresponding to each MAC codes.

Solution

The ERP level of Class I AMPS mobile phones = +6 dBW (standard)

The ERP level of Class II AMPS mobile phones = +2 dBW (standard)

The ERP level of Class III AMPS mobile phones = −2 dBW (standard)

The step size for adjusting these power levels = 4 dB (standard)

The lowest power level = −22 dBW (standard)

Number of bits in Mobile Attenuation Code (MAC) = 3 (given)

Maximum number of different steps 3-bit MAC can have $= 2^3 = 8$.
The ERP levels corresponding to each of 8 MAC codes are tabulated in Table 10.6.

Table 10.6 *Power levels for AMPS mobile phones*

Mobile Attenuation Code (MAC)	*Class I ERP in dBW*	*Class II ERP in dBW*	*Class III ERP in dBW*
000	+6	+2	−2
001	+2	+2	−2
010	−2	−2	−2
011	−6	−6	−6
100	−10	−10	−10
101	−14	−14	−14
110	−18	−18	−18
111	−22	−22	−22

The difference between maximum and minimum mobile power levels for a Class I mobile phone is 28 dB, whereas the power range for Class II and Class III mobile phones is 24 dB and 20 dB respectively.

In large cells, vehicle-mounted mobile phones and transportable mobile phones have better performance than handheld mobile phones under worst propagation conditions. If a handheld mobile phone attenuates the received signal considerably when operated inside a vehicle, the reliable communication can sometimes be established by simply coming out of the vehicle in fringe areas.

EXAMPLE 10.8 AMPS communication range

A handheld mobile phone (Class III) can communicate over a distance of 5 km. Assume that the communication range of a mobile phone is limited by its transmitter power, what would be the maximum communication range for a permanently vehicle-installed mobile phone (Class I) under the same circumstances in

(a) free-space propagation condition

(b) a typical mobile environment condition where the signal attenuation is proportional to the fourth power of the distance

What are the other factors that could limit communication range?

Solution

Maximum ERP for Class I mobile phone $= +6$ dBW (standard)

Maximum ERP for Class III mobile phone $= -2$ dBW (standard)

Step 1. *To find difference in maximum ERP of Class I and Class III phones*

Difference in power levels (dB) $= +6 - (-2) = 8$ dB

Difference in power levels (ratio) $=$ antilog $(8/10) = 6.3$

(a) *To determine communication range in free-space propagation condition*

Step 2. *In free-space propagation condition, the square-law attenuation rule is followed, that is, the communication range is proportional to the square root of power levels. Therefore,*

the communication range for a Class I mobile phone $= 5 \text{ km} \times (6.3)^{1/2}$

Hence, the communication range $= 12.6$ km

***(b)** To determine communication range in a typical mobile environment condition*

In a given mobile environment condition where the signal attenuation is proportional to the fourth power of the distance, the communication range is proportional to the fourth root of power. Therefore,

the communication range with a Class I mobile phone $= 5 \text{ km} \times (6.3)^{1/4}$

Hence, the communication range $= 7.9$ k

Other Factors that could Limit Communication Range The factors such as multipath propagation, shadowing by buildings or hills, more signal attenuation due to operating the handheld mobile phone inside a building or vehicle, sensitivity of mobile phone receiver, etc, could further limit the communication range.

Voice Modulation and Antennas Used in AMPS AMPS cellular system uses Frequency Modulation (FM) with frequency deviation of ± 12 kHz. It is desirable that the transmitted carrier frequency must be within 1 kHz of the specified channel frequency. In order to accommodate a large dynamic range of the speech signals, the input voice signals are compressed with a ratio of 2:1. That is, in the transmitter, a 2-dB change in speech signal level from the microphone causes only a 1-dB change in modulation level. The characteristics are specified such that a nominal 1 kHz reference input signal at a nominal signal level produces a ± 2.9 kHz peak frequency deviation of the transmitted carrier. Signaling data transmission uses FSK with ± 8 kHz frequency deviation.

Class C nonlinear power amplifiers can be used for higher efficiency because FM and FSK modulation schemes are used. The voice transmission is not very secure because of use of FM for voice communication. Any FM receiver tuned to the correct channel frequency can hear the conversations. However, when the mobile subscriber phone is communicating from a moving vehicle and the voice channel is changed due to hand-off in another cell, it is difficult to follow conversations at the new frequency.

Facts to Know!

Companding of speech signals has the advantage of confining the transmitted energy within the 30-kHz channel bandwidth and generates a quieting effect during a speech burst. The output of the compander is filtered with a pre-emphasis filter that has a nominal 6 dB/octave high pass response between 300 Hz and 3400 kHz. This result is an improvement in signal-to-noise ratio for low-level speech signals.

Most handheld mobile subscriber units use a quarter-wave monopole antenna. At 800 MHz frequency of operation, the length of this antenna is about 9.5 cm. The vehicle-mounted mobile phone subscriber units usually use a quarter-wave and a half-wave section, separated by an impedance-transforming coil. The use of more efficient antennas at the vehicle-mounted as well as handheld mobile subscriber units allows transmitter power to be reduced, thereby leading to longer battery life.

10.6.5 Call Procedures in AMPS

The base station sends its system parameters on forward control channels to all the mobile subscribers present in its service area. Each mobile subscriber updates its SID and establishes its paging channels once its SID matches the one transmitted by the base station. Then the mobile subscriber responds to the broadcast control signals and page signals received from the base station.

Similarly, when a mobile subscriber is switched on, it identifies itself to the system. It scans all the forward control channels and locks on to the strongest one. It verifies the SID received from the system to determine whether it is roaming or not. If it does not receive SID within three seconds, it tries the next strongest control channel. After receiving the system information, the mobile subscriber tunes to the strongest paging channel. Paging channels are control channels that carry information about calls that the system is trying to place to mobile subscribers. For an incoming call to the mobile, its phone number will be transmitted on the paging channel.

The mobile subscriber transmits its equipment serial number and mobile phone number to the MTSO via base station. The MTSO maintains a database with information about all mobile subscriber units. The control channel constantly updates the status of its associated reverse control channel. If the mobile subscriber loses the signal and reacquires it or detects that it has moved to another cell, it identifies itself again.

Mobile-Initiated Call Procedure When a call is placed by the mobile subscriber to another mobile subscriber or landline subscriber connected to the system via PSTN, the following sequence of events occurs:

Step 1. The mobile subscriber initiates a call by keying in the called subscriber's phone number and presses the send key. The mobile subscriber unit automatically transmits an origination message on the available reverse control channel to the cell-site. This message includes the mobile unit's ESN and MIN and the called subscriber's phone number.

Step 2. The cell-site passes the information on to the MTSO for processing. The MTSO verifies that the called subscriber number is valid and that the subscriber is authorised to place the call. In some systems, it is required that the mobile subscriber should enter a PIN (Personal Identification Number) as well as the called subscriber number for security reasons.

Step 3. Once authorisation is complete, the cell-site sends a message to the mobile subscriber on the forward control channel, informing it which voice channel to use for the call. It also sends the digital colour code that identifies the cell-site, and a control mobile attenuation code, which sets its transmitter power level to be used.

Step 4. The MTSO sends out a ringing signal to the called subscriber. All of these operations as mentioned in steps 2 through 4 occur within 10 seconds of initiating the call.

Step 5. When the called subscriber answers, the MTSO establishes a circuit between the calling and the called subscriber and initiates billing information. Now the serving cell site as well as the calling mobile subscriber switch from the control channel to a voice channel, but the audio is still muted on the mobile subscriber.

Step 6. The cell-site sends a control message on the forward voice channel confirming the voice channel. It then sends a Supervisory Audio Tone (SAT) on the voice channel to the mobile subscriber. The particular frequency of the SAT denotes the particular cell-site location for a given cannel and is assigned by the MTSO for each call. The mobile retransmits the SAT back to the cell-site, confirming that the desired cell-site and mobile subscriber are connected.

Step 7. The mobile subscriber sends a confirmation message on the reverse voice channel. Then the two-way voice communication begins. During the call, the SAT continues confirming the continuation of the call. Reception of the incorrect SAT by the cell-site indicates an interfering signal and interruption of the tone indicates the lost connection. If the SAT is not restored back within five seconds, the call is terminated. A 10-kHz signaling tone may also be transmitted on the voice channel during a call. It is used to signal hand-offs to another cell and the termination of the on-going call in its serving cell.

Step 8. When either the calling subscriber or the called subscriber hangs up, the MTSO releases the voice channels.

System Originated or Mobile-Terminated Call Procedure An incoming call is routed by the cellular system to the cell-site where the mobile subscriber has identified itself last time. If the mobile subscriber has not identified itself, then a recorded message is given to the calling subscriber informing that the called subscriber is either switched off or out-of-coverage range.

When an incoming call is routed to the mobile subscriber, the following sequence of events occurs:

Step 1. The cell-site sends the MIN on the paging channel along with the voice channel number and power level to be used by the called mobile subscriber.

Step 2. The mobile subscriber acknowledges this message and sends its ESN on the reverse control channel to be matched by the system with the MIN.

Step 3. The cell-site sends its system information on the forward voice channel along with the digital colour code information. The mobile subscriber confirms back the receipt of information on the reverse voice channel.

Step 4. The supervisory audio tone is transmitted on the voice channel and the two-way voice communication begins.

10.6.6 Narrowband AMPS (N-AMPS)

N-AMPS provides access to three mobile subscribers in a 30-kHz AMPS channel by using FDMA and 10-kHz channel bandwidth. This results in three times increase in user capacity. One N-AMPS channel uses the carrier frequency for the existing AMPS channel and, with the other two channels, the carrier frequencies are offset by ± 10 kHz. The FM deviation is also reduced in N-AMPS. Each 10-kHz subchannel is capable of handling its own voice calls. Reducing the channel bandwidth degrades speech quality by lowering the signal-to-interference ratio. With narrower bandwidths, voice channels are more vulnerable to interference.

N-AMPS uses the SAT and ST signalling and blank-and-burst data signaling in exactly the same manner as AMPS, except the signalling is implemented by using sub-audible data signals. SAT and ST signalling is sent using a continuous 200 bps NRZ encoded FSK data, and therefore are called DSAT and DST signaling in N-AMPS. The DST signal is simply the binary inverse of the DSAT. These data signals are sent digitally and repetitively in short predefined code blocks. There are seven different 24-bit DSAT code words, which may be selected by the MTSO. Both the base station and the mobile subscriber constantly repeat the DSAT code word during a call. The seven possible DSATs and DSTs are specially designed to provide a sufficient number of alternating 0s and 1s so that receivers may conveniently implement dc blocking. DSAT and DST signaling data are transmitted without blanking the voice by using 300-Hz highpass filter for each voice channel.

When the voice channel is busy, the voice-channel signalling is carried out with 100 bps Manchester encoded FSK data and is sent in place of DSAT. As with AMPS wideband signaling, there are many messages that may be passed between the base station and subscriber unit. These messages are transmitted in N-AMPS using the same BCH codes as in AMPS with a predefined format of 40 bit blocks on the forward voice channel and 48 bit blocks on the reverse voice channel. N-AMPS systems use standard AMPS control channels for call set-up and termination.

Facts to Know!

Narrowband AMPS systems are capable of providing dual mode operation in the sense that mobile subscriber units are capable of operating either with 30-kHz channels available in standard AMPS or with 10-kHz channels in N-AMPS.

10.7 ENHANCED TOTAL ACCESS COMMUNICATION SYSTEM

The Enhanced or European Total Access Communication System (ETACS), is virtually identical to AMPS, except that it uses 25-kHz channel bandwidth as opposed to 30-kHz channel bandwidth in AMPS in 900-MHz band. ETACS also differ from AMPS in formats of mobile identification number MIN. For ETACS, Area Identification Numbers (AIDs) are used instead of SID, and ETACS mobile subscriber units can access any control or voice channel.

EXAMPLE 10.9 Air-interface specifications of ETACS

List the major air-interface specifications of ETACS analog cellular system.

Solution

Cell-site frequency range	Tx: 935–960 MHz; Rx: 890–915 MHz
Mobile frequency range	Tx: 890–915 MHz; Rx: 935–960 MHz
RF spectrum bandwidth	25 MHz each
Duplexing technique	Frequency-Division Duplex (FDD)
Duplex separation	45 MHz
Multiple access scheme	FDMA
RF channel bandwidth	25 kHz
Number of RF channels	1000
Voice channel per RF channel	1
Voice modulation scheme	FM with peak deviation of ± 10 kHz for voice channels and ± 6.4 kHz for control data
Data rate on control channel	8 kbps
Spectral efficiency	0.33 bps/Hz
Error-control coding scheme	BCH (48, 36, 5) and (40, 28, 5)

ETACS supports forty-two control channels for each of two service providers operating in the same area. Thus, any mobile subscriber phone in the system only needs to scan a limited number of control channels to find the best serving cell-site. It is up to the service provider to ensure that neighbouring cell sites within a system are assigned forward control channels that do not cause adjacent channel interference to the mobile subscribers which monitor different control channels in nearby cell-sites.

10.8 US DIGITAL CELLULAR SYSTEMS (IS-54/136)

Second-generation cellular systems have been developed to provide higher-quality signals, higher data rates for support of digital services, and greater capacity. The key differences between the first-generation analog cellular and second-generation digital cellular systems can be summarised as follows:

(a) Digital Voice Channels The most notable difference between the two cellular generations is that first-generation cellular systems are almost purely analog, whereas second-generation cellular systems are digital. In particular, the first-generation cellular systems are designed to support voice channels using FM; digital data is supported only by the use of an FSK modem that converts the digital data into analog form. Second-generation cellular systems provide digitised voice channels.

(b) Encryption Because it is relatively easy to encrypt digital data to prevent eavesdropping, all second-generation digital cellular systems provide this capability

(c) Error Detection and Correction The digital data stream of second-generation cellular systems allows the use of error detection and correction techniques, such as Block Error correction Codes (BCH codes, Reed-Solomen codes, Block interleaving), Convolution Codes. The results are in the form of very clear voice reception.

(d) Channel Access In first-generation cellular systems, each cell supports a number of frequency channels. At any given time a channel is allocated to only one mobile subscriber. Second-generation cellular systems also provide multiple channels per cell, but each channel is dynamically shared by a number of subscribers using Time Division Multiple Access (TDMA) or Code Division Multiple Access (CDMA).

IS-54 and IS-136 stands for Interim Standard-54 and Interim Standard-136, which are cellular mobile communication standards employing digital technology. IS-54 and IS-136 are known as second-generation (2G) United States Digital Cellular (USDC) systems. The USDC systems use existing AMPS channels and allows for smooth transition between analog and digital cellular systems in the same service area. The USDC systems have been designed to share the same frequencies, frequency reuse plan, and cell-sites as that of AMPS. The USDC standard also uses the 45-MHz FDD scheme. The cell-sites and mobile subscriber units could be equipped with both AMPS and USDC channels within the existing hardware infrastructure. Because USDC maintains compatibility with AMPS in a number of ways, USDC is also known as Digital AMPS (D-AMPS).

Facts to Know!

In addition to limited system capacity, the utility of the first-generation cellular system has been mainly diminished by the proliferation of incompatible standards, which make it impossible for a subscriber to use the same cellphone in different countries. These limitations motivated the development of second-generation digital cellular systems with the objectives of achieving higher capacity and improved compatibility.

USDC is based on Time Division Multiple Access (TDMA) technique which supports three full-rate users or six half-rate users on each AMPS channel. Thus, USDC offers as much as six times the capacity of AMPS. By supporting both AMPS and USDC, cellular service operators have been able to provide new subscribers with USDC mobile phones and have gradually replaced AMPS cell-sites with USDC cell-sites.

Facts to Know!

In IS-136 standard, high-speed digital control channels are added for better security and additional features. IS-136 is used both in the 800-MHz cellular radio band and the 1900-MHz PCS band that is not compatible with the AMPS frequency allocation. One of the most important features of the TDMA digital cellular radio system is its backward compatibility with AMPS analog cellular radio system.

The dual mode USDC/AMPS system was standardised as Interim Standard 54 (IS-54) by the Electronic Industries Association and Telecommunication Industry Association (EIA/TIA) and later was upgraded to IS-136. In IS-54 standard, the AMPS control channels are retained as it is, and only the voice channels are digitised and Time Division Multiple Access (TDMA) scheme is employed.

USDC systems specifies dual-mode operation and backward compatibility with standard AMPS. USDC, like AMPS, divides the total available radio-frequency spectrum into individual 30-kHz cellular channels (that is, FDMA). However, TDMA allows more than one mobile unit to use a channel at the same time by further dividing transmissions within each channel into six time slots, one for each mobile subscriber unit using that channel. The technique of time-sharing channels significantly increases the capacity of a system, allowing more mobile subscribers to use a system at virtually the same time within a given service area.

USDC forward and reverse control channels use exactly the same signaling techniques as AMPS in order to maintain compatibility with AMPS mobile subscriber phones. USDC voice channels use 4-ary $\pi/4$ DQPSK modulation with a channel rate of 48.6 kbps. The IS-136 also uses $\pi/4$ DQPSK modulation for the USDC control channels and provides 4-ary keying instead of FSK on dedicated USDC control channels in order to increase control channel data rate, and to provide specialised services such as paging and short messaging.

10.8.1 IS-54 Specification Standards

The EIA/TIA standardised the dual-mode USDC/AMPS system as IS-54. Dual mode operation specifies that a mobile subscriber complying with the IS-54 standard must be capable of operating in either the analog AMPS or the digital USDC mode for voice transmissions. Using IS-54, a cellular service provider could convert any or all of its existing analog channels to digital channels. The key criterion for achieving dual-mode operation is that IS-54 digital channels cannot interfere with transmissions from existing analog AMPS base stations and mobile subscribers. This objective is achieved by providing digital control channels and both analog and digital voice channels.

IS-54 employs the same 30-kHz channel spacing and frequency bands (824 MHz–849 MHz and 869 MHz–894 MHz) as AMPS. Each 30-kHz channel pair is divided into three time slots (hence time division) and digitally compressing the voice data. This yields three times the call capacity in a single cell. The IS-54 standard specifies 84 control channels, 42 of which are shared with AMPS. To maintain compatibility with the existing AMPS cellular telephone system, the primary forward and reverse control channels in IS-54 cellular systems use the same signaling techniques and modulation scheme (binary FSK) as AMPS. An AMPS/IS-54 infrastructure can support use of either analog AMPS mobile phones or D-AMPS mobile phones. The IS-54 system also incorporates encryption to make calls more secure.

Before a voice channel is assigned, IS-54 Dual-mode mobile subscriber units use AMPS forward and reverse control channels to carry out user authentications and call management operations. When a mobile subscriber unit transmits an access request, it indicates that it is capable of operating in the digital mode; then the base station will allocate an available digital voice channel. The allocation procedure indicates the channel number and the specific time slot within a TDMA frame. IS-54 specifies a 48.6-kbps rate per 30-kHz voice channel divided among three simultaneous mobile subscribers. Each mobile subscriber is allocated 13 kbps, and the remaining 9.6 kbps is used for timing and control overhead.

EXAMPLE 10.10 Air-interface specifications of USDC IS-54/IS-136

List the major air-interface specifications of USDC IS-54/IS-136 digital cellular system.

Solution

Cell-site frequency range	Tx: 869–894 MHz; Rx: 824–849 MHz
Mobile frequency range	Tx: 824–849 MHz; Rx: 869–894 MHz
RF spectrum bandwidth	25 MHz each on uplink and downlink
Duplexing technique	Frequency-Division Duplex (FDD)
Duplex separation	45 MHz
Multiple access scheme	TDMA
RF channel bandwidth	30 kHz
Number of RF channels	832 (3 or 6 users per channel)
Voice channels per RF channel	3 with full-rate @ 7.95 kbps/user; 6 with half-rate @ 3.975 kbps/user
Voice modulation scheme	$\pi/4$ DQPSK
Channel data rate	48.6 kbps
Spectral efficiency	1.62 bps/Hz
Interleaving	2 slot interleaver
Speech coding	VSELP @ 7.95 kbps
Channel coding	7 bit CRC and rate 1/2 convolutional coding of constraint length 6
Type of equalisers	Adaptive
Handheld mobile Tx power	600 mW max.; 200 mW avg.

USDC and AMPS use identical 10-kbps binary FSK with Manchester coding and control channels. On voice channels, the FM modulation is replaced with digital modulation having a gross bit rate of 48.6 kbps. To achieve a transmission bit rate of 48.6 kbps in a 30-kHz AMPS voice channel, the modulation requires a spectral efficiency of 1.62 bps/Hz. This cannot be obtained with binary FSK modulation scheme. Moreover, spectral shaping on the digital channel must be used to limit adjacent channel interference. The spectral efficiency requirements can be met by using conventional pulse-shaped, four-phase digital modulation schemes such as QPSK and OQPSK. However, USDC voice and control channels use a symmetrical differential phase-shift keying technique known as $\pi/4$ Differential Quadrature Phase Shift Keying (DQPSK). This offers several advantages in a mobile radio environment such as improved bandwidth efficiency and cochannel interference rejection.

In a $\pi/4$ DQPSK modulator, data bits are split into two parallel channels that produce a specific phase shift in an analog carrier signal. There are four possible bit pairs, leading to four possible phase shifts using a quadrature I/Q modulator. Pulse shaping is used to minimise the bandwidth while limiting the intersymbol interference. In the transmitter, the PSK signal is filtered using a square-root raised cosine filter with a roll-off factor of 0.35. Using pulse shaping with $\pi/4$ DQPSK allows for the simultaneous transmission of three separate 48.6-kbps speech signals in a 30-kHz bandwidth. A 48.6-kbps data rate requires a symbol rate of 24.3 ksps (kilosymbols per second) with a symbol duration of 41.1523 μs. The use of pulse shaping with as much as 50 dB of adjacent-channel isolation enables to reduce the transmission bandwidth while limiting the intersymbol interference (ISI). Thus, the bandwidth efficiency using $\pi/4$ DQPSK is 3×48.6 kbps/30kHz = 4.86 bps/Hz.

For a system using DQPSK modulation at a symbol rate of 24.3 ksps, if the bit error rate due to intersymbol interference becomes intolerable for a σ/T value of 0.1 (where σ is the rms delay spread and T is the symbol duration) then the maximum value of rms delay spread that can be tolerated is 4.12 μs. If the rms delay spreads exceeds this value, it is necessary to use equalisation in order to reduce the BER.

There is one difference in the RF component of TDMA based USDC mobile subscriber phones. There is an addition of a new power class, that is, Class IV. It has three new power levels of –26 dBW, –30 dBW, and –34 dBW ERP. Since the transmitter operates only one-third of the time, battery life is improved considerably. This is a desirable feature in microcell environment.

Facts to Know!

For USDC, a Decision Feedback Equaliser consisting of four feed-forward taps having a 1/2 symbol spacing and one feedback tap can be used. The coefficients of the adaptive filter are updated using the recursive least squares algorithm. This type of fractional spacing makes the equaliser robust against sample timing jitter.

EXAMPLE 10.11 Power level of TDMA mobile phone

Express the minimum power output for a Class IV TDMA mobile phone in mW. How does it compare with Class I mobile phone?

Solution

Step 1. *To express minimum power level for a Class IV phone in dBm*

Minimum power level for Class IV mobile phone = –34 dBW (standard)

We know that the power level in dBm = Power level in dBW –30 dB

Therefore, power level in dBm = –34 dBW–30 dB = = –64 dBm

Step 2. *To express minimum power level for a Class IV phone in mW*

We know that the power level in mW = antilog (power level in dBm/10)

Therefore, power level in mW = antilog (–64/10)

Hence, power level in mW = 4×10^{-7} mW

Step 3. *To express ERP for a Class I phone in dBm*

The ERP for a Class I mobile phone = +6 dBW (standard)

We know that the power level in dBm = Power level in dBW–30 dB
Therefore, power level in dBm = +6 dBW– 30 dB = = –24 dBm

Step 4. *To express ERP for a Class I phone in mW*
We know that the power level in mW = antilog (power level in dBm/10)
Therefore, power level in mW = antilog (–24/10)
Hence, power level in mW = 4×10^{-3} mW

Step 5. *To compare power levels*
Power level for a Class IV phone = 4×10^{-7} mW
Power level for a Class I phone = 4×10^{-3} mW
The ratio of power level of Class I phone and Class IV phone is equal to

$$(4 \times 10^{-3}\ \text{mW})/(4 \times 10^{-7}\ \text{mW}) = 10{,}000$$

Expressing it in dB, 10 log 10,000 = 40 dB.
Hence, minimum power level for a Class I phone is greater than that of minimum power level of Class IV mobile phone by a factor of 40 dB or 10 000.

The USDC specifies 11 radiated power levels for four classifications of mobile subscriber units, including the eight power levels used by standard AMPS transmitters. The fourth classification is for dual-mode TDMA/AMPS cellular telephones. The USDC power classifications are listed in Table 10.7.

Table 10.7 *Power levels for USDC mobile phone*

Power Level	*Class I dBm (mW)*	*Class II dBm (mW)*	*Class III dBm (mW)*	*Class IV dBm*
0	36 (4000)	32 (1600)	28 (640)	28
1	32 (1600)	32 (1600)	28 (640)	28
2	28 (640)	28 (640)	28 (640)	28
3	24 (256)	24 (256)	24 (256)	24
4	20 (100)	20 (100)	20 (100)	20
5	16 (41)	16 (41)	16 (41)	16
6	12 (16)	12 (16)	12 (16)	12
7	8 (6.6)	8 (6.6)	8 (6.6)	8
8	---	---	---	4 dBm ±3 dB
9	---	---	---	0 dBm ±6 dB
10	---	---	---	−4 dBm ±9 dB
Defined for dual mode mobile phone only				

The highest power level is +36 dBm (4 W), and successive power levels differ by 4 dB, with the lowest power level for class I through class III being +8 dBm (6.6 mW). The lowest transmit power level for dual-mode mobile subscriber units is –4 dBm (0.4 mW) ± 9 dB. In a dual-mode system, the three lowest power levels can be assigned only to digital control channels and digital voice channels. Analog digital voice channels and FSK control channels transmitting in the standard AMPS format are confined to the eight power levels in the AMPS specification. The average transmitted power is 4.8 dB below specifications because transmitters in the TDMA mode are active only one-third of the time.

10.8.2 USDC Control Channels

The USDC IS-54 standard specifies two distinct types of control channels: analog control channels (ACCH), and digital control channels (DCCH). The ACCH specifies the same 42 primary control channels as AMPS. The USDC specifies additional 42 control channels called secondary control channels which are actually digital using FSK at a channel data rate of 10 kbps. Thus, USDC is capable of providing twice the capacity of control traffic within a given service area.

The USDC IS-136 provides three distinct types of control channels: Analog Control Channels (ACCH), 10-kbps binary FSK Digital Control Channel (DCCH), and a digital control channel with a signaling rate of 48.6 kbps on USDC-only control channels. The new digital control channel is meant to eventually replace the analog control channel. With the addition of a digital control channel, a mobile subscriber unit is able to operate in the digital domain. Digital control channels consist of pairs of slots on the same RF channels that are used for voice. The DCCH can be assigned to any RF channel. As with the voice channels, separate forward and reverse control channels are needed. Normally, there is one DCCH pair per cell, or per sector in a sectorised cellular system. Figure 10.15 depicts TDMA digital control channel frame format.

The total bit rate for a DCCH is one-third of the RF channel bit rate, that is, $44.6/3 = 14.9$ kbps, compared with 10 kbps for an ACCH. This additional capacity makes the digital control channels useful for special features such as call-status display and short text messages.

Figure 10.16 shows how one time slot is divided in IS-136 TDMA for a forward-control digital channel.

The forward channel is under the control of the cell-site, but many mobile subscribers share a single reverse control channel. In the forward channel, the 28-synchronising (SYNC) bits have the same function as for the voice channels, allowing the mobile subscriber receivers to lock on the beginning of the transmission. The 12-Shared Channel Feedback (SCF) bits perform several functions. They provide acknowledgement of messages from the mobile subscriber and inform the mobile subscriber of the status of the reverse control channel. By monitoring the status of the reverse channel as reported by the cell-site in the SCF field, the mobile subscribers can reduce the possibility of collisions during simultaneous transmissions of data.

The 12-Coded Superframe Phase (CSFP) bits identify the location of this time slot in a larger frame that extends over 16 TDMA frames or 32 blocks of control-channel data in a time span of 640 milliseconds. Each block is designated as containing broadcast, paging, messaging, or access response information. Each of these types of data as well as the voice are time-division multiplexed. Four of every six time slots on the RF channel are used for voice.

A hyperframe comprises of two superframes. The hyperframe structure allows data to be repeated. This means that a mobile receiver can check the signal strength on other channels while transmitting or receiving information data.

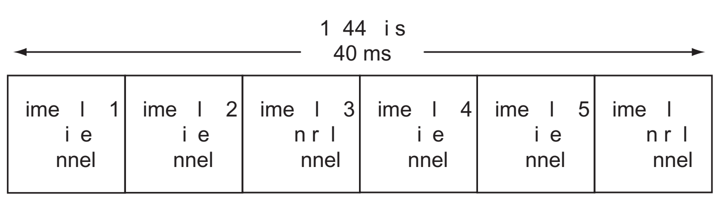

Fig.10.15 *TDMA digital control channel frame*

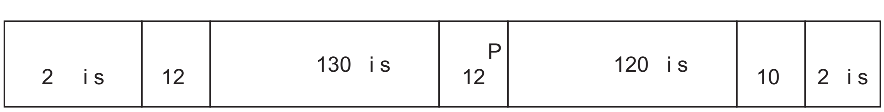

Fig.10.16 *IS-136 TDMA digital-control forward-channel time slot*

EXAMPLE 10.12 Transmission bit rate in digital cellular systems

A digital cellular communication system has a forward channel frequency band of 810 MHz to 826 MHz and a reverse channel frequency band of 940 MHz to 956 MHz. Assume that 10 per cent of the total bandwidth is used by control channels exclusively. It is required to support at least 1150 simultaneous calls using FDMA. The modulation scheme employed has a spectral efficiency of 1.68 bps/Hz. Assuming that the channel impairments necessitate the use of rate 1/2 FEC codes, find the upper bound on the transmission bit rate that a speech coder used in this system should provide.

Solution

Step 1. *To find bandwidth available on forward channel*

Spectrum allocated on forward channel = 810 MHz to 826 MHz (given)

Bandwidth available on forward channel = 826 MHz – 810 MHz = 16 MHz

Step 2. *To find bandwidth available on reverse channel*

Spectrum allocated on forward channel = 940 MHz to 956 MHz (given)

Bandwidth available on reverse channel = 956 MHz – 940 MHz = 16 MHz

Step 3. *To find bandwidth available for voice channels*

Bandwidth used by control channels = 10 % of total bandwidth (given)

Bandwidth used by control channels = 0.1×16 MHz = 1.6 MHz

Bandwidth available for voice channels = 16 MHz – 1.6 MHz = 14,400 kHz

Step 4. *To find maximum channel bandwidth*

Number of simultaneous calls using FDMA = 1150 (given)

Maximum channel bandwidth = 14,400/1150

Therefore, maximum channel bandwidth = 12.5 kHz or 12,500 Hz

Step 5. *To find maximum net transmission data rate*

Spectral efficiency = 1.68 bps/Hz (given)

Maximum channel data rate = 1.68×12500 bps = 21000 bps or 21 kbps

FEC coder rate = or 0.5 (given)

Therefore, maximum net data rate = 21×0.5 kbps = 10.5 kbps

Thus, there is a need to design a speech coder with a maximum data rate of 10.5 kbps.

USDC IS-136 Digital Logical Control Channels The new digital control channel includes several logical channels in the data section of the TDMA digital control channel with different functions. These logical channels can be broadly categorised as

– SMS point-to-point, Paging, and Access Response Channel (SPACH)
– Broadcast Control Channel (BCCH)
– Shared Channel Feedback (SCF) channel
– Random Access Channel (RACH)

A SPACH logical channel can carry messages related to a single mobile subscriber unit or to a small group of mobile subscriber units. Larger messages are split into smaller blocks for transmission. The Short Message Service Channel (SMSCH), Paging Channel (PCH), and Access Response Channel (ARCH) are used for control messages as well as short paging-type messages to be displayed on the mobile phone. Page messages are always transmitted and then repeated again. A mobile subscriber unit automatically moves to an access response channel immediately after successful completion of contention- or reservation-based access on a random access channel. It is not necessary for every mobile subscriber to monitor all these messages. The battery life of the mobile phone is extended. The mobile subscriber is instructed which block to monitor and can go into an idle or sleep mode while it waits for a call.

The Broadcast Control Channel (BCCH) contains information intended for all mobile subscribers. The Fast Broadcast Control Channel (F-BCCH) is used to transmit system parameters to all the mobile subscribers. These include the structure of the superframe itself, the system identification, registration and access parameters. Mobile subscriber units use F-BCCH information when initially accessing the system to determine the beginning and ending of each logical channel in the DCCH frame.

The F-BCCH logical channel also includes information pertaining to access parameters including information necessary for authentication and encryptions. The Extended Broadcast Control Channel (E-BCCH) has less critical information, such as lists of the channels used in neighbouring cells. This information can be transmitted over 1 to 8 time slots per superframe. Figure 10.17 shows how one time slot is divided for IS-136 TDMA reverse digital control channel.

The reverse digital control channel is quite different from the forward digital control channel. There is no broadcast information needed. Only one logical channel called the Random Access Channel (RACH) is required. This is used by the mobile subscribers to access the base station for registration, authentication, and call set-up procedures. As with the reverse voice channel, guard time (6-bit Guard field shown as G) is needed to avoid interference between mobile subscribers located at different distances from the cell-site, and time has to be allocated for ramping up the mobile transmitter power (6-bit Ramp field shown as R).

Table 10.8 summarises the logical control channels for the USDC IS-136 standard.

Fig.10.17 *IS-136 TDMA digital control reverse channel time slot*

Table 10.8 *Logical digital control channels in USDC IS-136*

Channel	*Sub-Channel*	*Transmission direction*	*Major functions*
SMS point-to-point, Paging, and Access Response Channel (SPACH)	Paging Channel (PCH)	Downlink (BS → MS)	Delivers pages and orders; transmits paging and user-alerting messages; messages such as call-history count updates and shared secret data updates used for the authentication and encryption process. Each PCH message can carry up to five mobile identifiers.
	Access Response Channel (ARCH)	Downlink (BS → MS)	Messages assigning a mobile unit to an analog voice or a digital voice channel or redirecting it to a different cell along with registration access (accept, reject, or release) messages.
	SMS Channel (SMSCH)	Downlink (BS → MS)	Used to deliver short point-to-point messages to a specific mobile station. Each message is limited to a maximum of 200 characters of text. Mobile-originated SMS is also supported.
Broadcast Control Channel (BCCH)	Fast broadcast Control Channel (F-BCCH)	Downlink (BS → MS)	Broadcasts digital control channel (DCCH) structure parameters, including information about the number of F-BCCH, E-BCCH, and S-BCCH time slots in the DCCH frame.

(*Continued*)

Table 10.8 (*Continued*)

Channel	*Sub-Channel*	*Transmission Direction*	*Major functions*
	Extended Broadcast Control Channel (E-BCCH)	Downlink (BS → MS)	Carries information about neighbouring analog and TDMA cells and optional messages, such as emergency information, time and date messaging, and the types of services supported by neighbouring cells.
	SMS Broadcast Control Channel (S-BCCH)	Downlink (BS → MS)	Used for send short messages to individual mobile units.
Shared Channel Feedback (SCF) channel	---	Downlink (BS → MS)	Used to support random access channel operation by providing information about which time slots the mobile unit can use for access attempts and also if a mobile unit's previous RACH transmission was successfully received.
Random Access Channel (RACH)	---	Uplink (MS → BS)	Access messages, such as origination, registration, page responses, audit confirmation, serial number, message confirmation, information on authentication, security parameter updates, and point-to-point SMS.

10.8.3 USDC IS-136 Digital Voice Channel

USDC IS-136 digital voice channel is assigned a 30-kHz bandwidth on both the forward and the reverse link for digitised voice transmissions. Each radio-frequency voice channel consists of one 40-ms TDMA frame comprised of six time slots containing 324 bits each. There are 25 such TDMA voice frames per second. Each voice channel can support as many as six half-rate mobile users and three full-rate mobile users simultaneously, as shown in Fig. 10.18 and Fig. 10.19 respectively.

Six voice signals, occupying one time slot each, can be accommodated with a half-rate speech TDMA signal. Each voice signal is assigned to two time slots to support full-speech rate user, thereby accommodating three voice signals in one frame containing six uniform time slots. For example, Mobile user 1 occupies time slots 1 and 4, Mobile user 2 occupies time slots 2 and 5, and Mobile user 3 occupies time slots 3 and 6. A mobile transmits during two of the six time slots and receives on different two time slots. The remaining two time slots are used by the mobile subscribers to check the signal strength in adjacent cells which assist in initiating a hand-off.

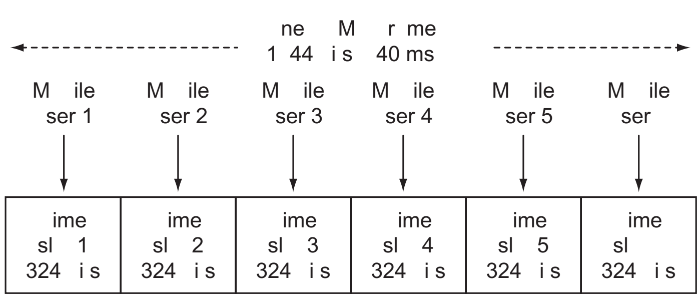

Fig.10.18 *The USDC TDMA frame format (half-rate users)*

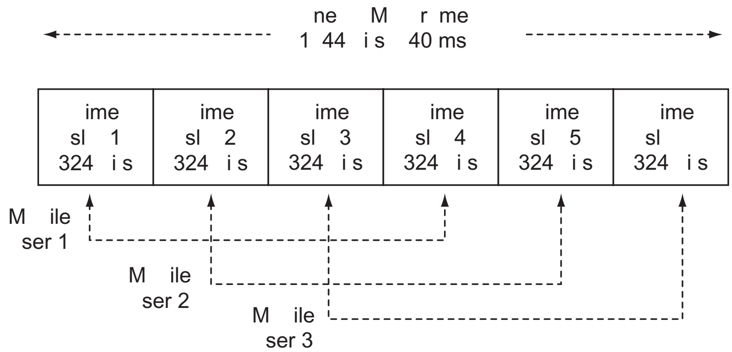

Fig.10.19 *The USDC TDMA frame format (full-rate users)*

In any time slot, 260 bits are used for the 13-kbps full-rate voice data. In addition, the bits corresponding to each 20 ms of speech are divided among two time slots. Interleaving the data bits helps to reduce the effect of burst errors. The other 64 bits are overhead; 28 of these are for synchronisation that contain a specific known bit sequence to establish frame alignment. Overhead reduces the number of data bits available per time slot to 260 bits instead of 324 bits.

Speech encoding is used to limit the bit rate to approximately 8 kbps for each speech channel for the full-rate system. The data rate available for each voice channel is 260 bits/20 ms = 13 kbps. The voice is actually encoded at 7.95 kbps and the remaining bits are used for error correction. The half-rate system will use 4 kbps for voice coding.

In fact, the IS-54 system has different synchronisation sequences for each of the six time slots making up the frame, thereby allowing each receiver to synchronise to its own preassigned time slots. An additional 12 bits in every time slot are for the SACCH, that is system control information. The Digital Verification Colour Code (DVCC) is the equivalent of the supervisory audio tone used in the AMPS system. There are 256 different 8-bit color codes, which are protected by a (12, 8, 3) Hamming code. Each cell-site has its own preassigned digital verification colour code, so any incoming interfering signals from distant cells can be ignored.

Each time slot contains 324 bits for both forward and reverse channels. The frames are synchronised for the forward and reverse channels, but the timing is offset so that a frame starts 90 bits (1.85 ms) earlier at the mobile. The allocation of these bits is different for the forward and reverse links, as shown in Fig. 10.20.

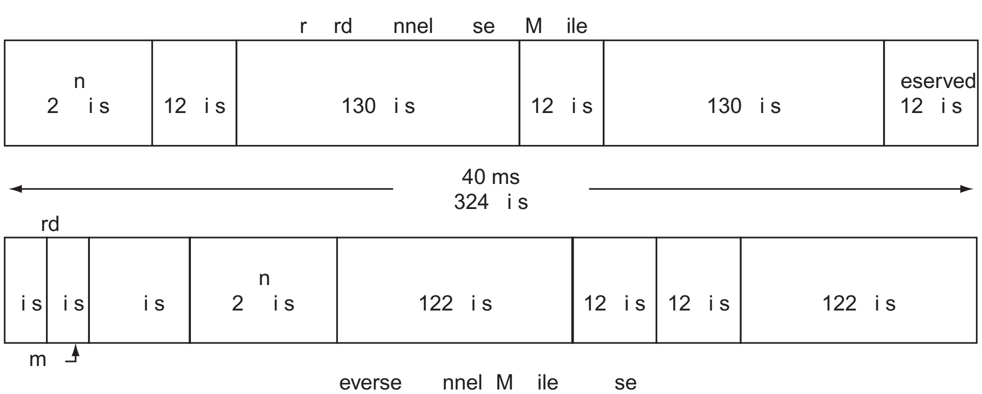

Fig.10.20 *TDMA voice time slots*

In a forward-channel time slot, the cell-site transmitter is on all the time, since it uses all six time slots to transmit to three mobiles on the same RF channel. In the reverse-channel time slot, the mobile waits for a guard time before transmitting. This is necessary because different amounts of propagation delay could cause mobile transmissions to overlap when received at the base station. It is mandatory that the mobile subscriber must be off when it is not expected to transmit in order to avoid interference. Then the mobile subscriber needs time to switch its transmitter on for each transmit time slot. Additional six-bit periods, called *ramp bits* (equivalent to 123 μs) are allocated for this purpose.

EXAMPLE 10.13 TDMA voice frame

In a full-rate TDMA system used in USDC IS-54 standard, calculate
(a) the duration and number of bits in a time slot of a voice frame
(b) the duration of a bit and guard time
(c) the maximum distance between the cell-site and mobile that can be accommodated with a guard time calculated in part (b)

Solution

(a) *To calculate the duration and number of bits in a time slot of a voice frame*

Step 1. *To calculate the duration of a time slot of a voice frame*

Duration of a TDMA voice frame = 40 ms (standard)

Number of time slots in a frame = 6 (standard)

Duration of a time slot of a voice frame = 40 ms/6 = 6.667 ms

Step 2. *To calculate the number of bits in a time slot of a voice frame*

Number of bits in a voice frame = 1944 bits (standard)

Number of bits in a time slot = 1944/6 = 324 bits

(b) *To calculate the duration of a bit and guard time*

Step 3. *To calculate the duration of a bit*

Duration of a TDMA voice frame = 40 ms (standard)

Number of bits in a voice frame = 1944 bits (standard)

The bit duration = 40 ms / 1944 bits

Hence, the bit duration = 20.5 μs

Step 4. *To calculate the guard time*

Number of bits in guard band = 6 (standard)

Guard time, $t_g = 6 \times 20.5\ \mu s = 123\ \mu s$

(c) *To calculate the maximum distance between cell-site and mobile*

The signal from cell-site to mobile is delayed by the propagation time between the cell-site and mobile. This causes the synchronisation of the mobile to be off by that much time, and it will be late in starting its transmission. In addition, the signal is further delayed by the propagation time from mobile to cell-site. Therefore, the given guard time, t_g of 123 μs (as calculated in Step 4) must include the round-trip propagation time.

Since the electromagnetic radio waves propagate at the speed of light ($c = 3 \times 10^5$ km/s), the maximum total round-trip distance is

$$d_{rt} = c\ t_g$$

Or, $d_{rt} = (3 \times 10^5 \text{ km/s}) \times (123 \times 10^{-6} \text{ s}) = 36.9$ km

The maximum distance between the cell-site and mobile is half of the round-trip distance. Therefore,

The maximum distance between cell-site and mobile, $d_{max} = d_{rt} / 2$

Or, $d_{max} = 36.9 \text{ km}/2 = 18.45$ km

Example 10.14 **Gross bit-rate of TDMA voice frame**

In a full-rate TDMA system used in USDC IS-54 standard, calculate the total gross bit and baud rate of RF signal, and the bandwidth efficiency in bps/Hz. Comment on the results obtained.

Solution

Step 1. *To calculate the total gross bit rate of RF signal*

Duration of a TDMA voice frame = 40 ms (standard)

Number of frames in one second = 1 s/40 ms = 25 frames

Number of bits in a TDMA voice frame = 1944 bits (standard)

Total gross bit rate for the RF signal = 1944 × 25 = 48.6 kbps

Step 2. *To calculate the baud rate of RF signal*

Since two bits per symbol are used in $\pi/4$ QPSK modulation, therefore,

Total gross baud rate for the RF signal = 48.6 /2 = 24.3 kbps

Step 3. *To calculate bandwidth efficiency of RF signal*

Total gross bit rate per RF channel = 48.6 kbps

Channel bandwidth = 30 KHz (standard)

Bandwidth efficiency = 48.6 / 30 = 1.6 bps/Hz

Comment on the results The bandwidth efficiency is quite conservative. This is because of the mobile radio environment which is subjected to noise, interference, and deep fading.

Each time slot in every USDC voice-channel frame contains four data channels—three for control channel and one for digitised voice and user data. The full-duplex digital voice channel carries digitised voice information and consists of a forward digital voice channel and a reverse digital voice channel and that carry digitised speech information or user data. The forward digital voice channel carries user-speech data from the base station to the mobile subscriber unit and the reverse digital voice channel carries speech data from the mobile subscriber unit to the base station.

Digital voice time slot contains three supervisory channels. These are the Coded Digital Verification Colour Code (CDVCC), the Slow Associated Control Channel (SACCH), and the Fast Associated Control Channel (FACCH) for synchronisng, equaliser training, and control information. The CDVCC is a 12-bit message transmitted in every time slot. The CDVCC consists of an eight-bit digital voice colour code number between 1 and 255 appended with four additional coding bits derived from a shortened Hamming code. The CDVCC provides cochannel identification. The base station transmits a CDVCC number on the forward voice channel, and each mobile subscriber unit retransmits the same code back to the base station on the reverse voice channel. If the two CDVCC values are not identical, the time slot is released for other mobile subscribers.

The SACCH is a signaling channel for transmission of control and supervision messages between the base station and the mobile subscriber unit during a call. These messages include information such as communicating power-level changes and hand-off requests. The SACCH is also used by the mobile subscriber unit to report signal-strength measurements of neighbouring base stations so that the base station can initiate a mobile-assisted hand-off. The SACCH uses 12 coded bits per TDMA burst and is transmitted in every time slot. This provides a signaling channel in parallel with the digitised speech information. Therefore, SACCH messages can be transmitted without interfering with the processing of digitised speech signals. The SACCH can take up to 22 frames for a single SACCH message to be transmitted.

There are provisions to steal voice data bits for additional control information as required. These stolen bits form the Fast Associated Control Channel (FACCH), which is used for urgent information such as hand-off commands. The FACCH is a second signaling channel for transmission of control and specialised supervision and voice messages between the base station and the mobile subscriber units. The FACCH channel does not have a dedicated time slot. It is a blank-and-burst type of transmission. It replaces digitised speech information with control and supervision messages within a subscriber's allocated time slot.

The net digitised voice transmission rate of 13-kbps cannot be reduced below 3250 bps in any given time slot. There are no fields within a standard time slot to identify it as digitised speech or an FACCH message. To determine if an FACCH message is being received, the mobile subscriber unit calculates the cyclic redundancy check. If it is correct, the message is assumed to be an FACCH message. The FACCH data are packaged and interleaved to fit in a time slot similar to the way digitised speech is handled. The FACCH supports transmission of DTMF tones, call release instruction, flash hook instructions, and mobile-assisted hand-off.

10.8.4 Speech Coding in USDC System

Figure 10.21 shows the block diagram for a USDC digital voice-channel speech encoder.

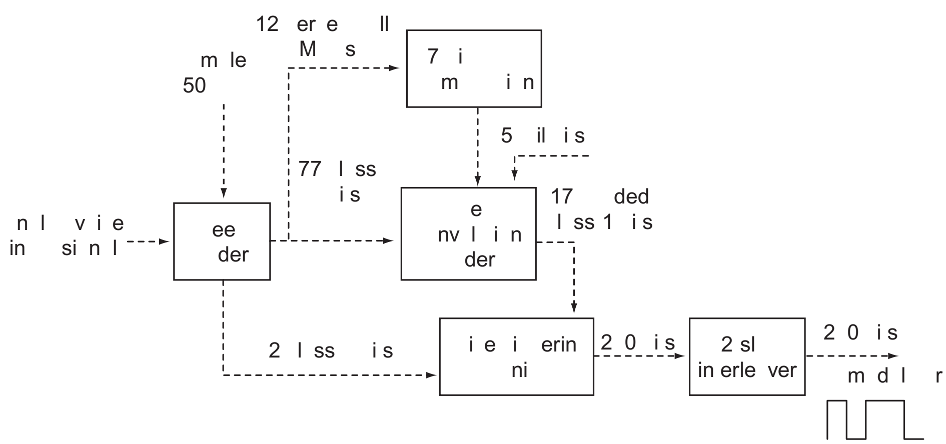

Fig. 10.21 *USDC digital voice-channel speech coder*

Channel error control for the digitised speech data uses three mechanisms for minimising channel errors:

(1) a rate one-half convolutional code is used to protect the more vulnerable bits of the speech coder data stream;
(2) transmitted data are interleaved for each speech coder frame over two time slots to reduce the effects of Rayleigh fading; and
(3) a cyclic redundancy check is performed on the most perceptually significant bits of the digitised speech data. With USDC, incoming analog voice signals are sampled first and then converted to a binary PCM in a special speech coder (vocoder) called a Vector Sum Excited Linear Predictive (VSELP) coder among the class of code excited linear predictive coders. Linear predictive coders are time-domain types of vocoders that attempt to extract the most significant characteristics from the time-varying speech waveform. These coders are based upon quantising the residual excitation signal. With linear predictive coders, it is possible to transmit good-quality voice at 4.8 kbps and acceptable voice quality at lower bit rates.

The VSELP coder has a bit rate of 7950 bps and produces a speech frame every 20 ms. Therefore, number of bits per frame = 7950 bits/second × 20 ms/frame = 159 bits-per-frame. In one second, fifty speech frames are produced by the coder for a particular subscriber, each containing 159 bits of speech, or output data rate = 50 frames/second × 159 bits/frame = 7950 bps. According to the significance, the bits included in each speech coder frame are divided into two classes in which they are perceived. There are 77 Class-I bits and 82 Class-II bits. The Class I bits being the most significant bits, are error protected using a rate 1/2 convolutional code of constraint length $L = 6$. The 12 most significant Class-I bits are block coded using a seven-bit CRC error-detection code prior to convolutional coding so to ensure that the most significant speech coder bits are decoded with a low probability of error. The less significant Class-II bits have no means of error protection.

After channel coding, the 159 bits in each speech coder frame are represented by 260 channel-coded bits. 50 such frames are transmitted each second, and the gross bit rate of the speech coder with added channel coding is 13.0 kbps (that is, 260 bits/frame × 50 frames/second). Hence, the transmission bit rate is increased from 7950 bps for each digital voice channel to 13 kbps. Before transmission, the encoded speech data is interleaved over two time slots with the speech data from adjacent speech frames. In other words, each time slot contains exactly half of the data from each of two sequential speech coder frames. This means only 130 of the necessary 260 bits are provided for two consecutive frames—the previous speech frame and the present or most recent speech frame. The encoded speech data for the two adjacent frames are placed into the interleaver in such a manner that intermixes the Class-II bits and the Class-I bits.

A FACCH data block contains 49 bits of data per each 20 ms frame in the channel coding. A 16 bit CRC code word is appended to it. The 65-bit coded word is then processed through a rate 1/4 convolutional coder of constraint length, six. This results into 260 bits of FACCH data per each 20 ms frame. A FACCH data block occupies the same amount of bandwidth as a single frame of coded speech. Speech data on the digital voice channel can be replaced with coded FACCH data. Interleaving of digital voice channel and FACCH data is handled identically in USDC. The interleaving approach for coded FACCH blocks is identical to that used for speech data. The SACCH data word of 6 bits during each 20 ms speech frame is processed through a rate 1/2 convolutional coder of constraint length five to produce 12 coded bits. The SACCH coded message word uses an incremental interleaver that spans over 12 consecutive time slots.

EXAMPLE 10.15 Comparison of the capacity of AMPS and IS-136 cellular systems

Compute the user capacity of 1G AMPS FDMA analog cellular system and 2G IS-136 TDMA digital cellular system. Comment on the results obtained.

Solution

Step 1. *To compute the capacity of the AMPS analog cellular system*

Let the allocated bandwidth, $B_t = 12.5$ MHz

The carrier channel bandwidth, $B_c = 30$ kHz

Frequency reuse factor, $K = 7$ (commonly used in AMPS system)

Total number of available channels, $N = B_t/B_c = 12.5 \text{ MHz} / 30 \text{ kHz} \approx 416$

Assuming uniform traffic distribution, number of channels allocated in each cell would be same and is equal to $416/7 \approx 60$ users per cell.

Alternately, user capacity per cell, M, is given by the expression

$$M = (B_t/B_c) \times (1/K)$$

Or, $M = (12.5 \text{ MHz}/30 \text{ kHz}) \times 1/7 \approx 60$ users per cell

Step 2. *To compute the capacity of the IS-136 digital cellular system*

Let the allocated bandwidth, $B_t = 12.5$ MHz (same as that of AMPS)

The carrier channel bandwidth, $B_c = 30$ kHz
Number of users per channel, $N_u = 3$
Total number of available channels, $N = B_t/B_c = 12.5$ MHz/30 kHz ≈ 416
Total number of users per channels, $N_U = N \times N_u = 416 \times 3 = 1248$
Frequency reuse factor, $K = 4$ (commonly used in IS-136 system)
Assuming uniform traffic distribution, number of users in each cell would be same and is equal to 1248/4 = 312 users per cell.
Alternately, user capacity per cell, M, is given by the expression
$M = (B_t / B_c) \times N_u \times (1/K)$
Or, $M = (12.5$ MHz/30 kHz$) \times 3 \times 1/4 \approx 312$ users per cell

Comment on the results The capacity of the IS-136 digital cellular system is many times more than the capacity of AMPS analog cellular system.

10.9 PERSONAL DIGITAL CELLULAR (PDC)

Previously known as Japanese Digital Cellular (JDC), the Personal Digital Cellular (PDC) is a Japanese standard for a 2G digital cellular radio system operating in the 800/900 MHz, and 1400 MHz frequency bands. PDC employs Frequency Division Duplexing (FDD) technique to support full-duplex wireless mobile communications. In 800/900 MHz band, the downlink transmission in the 940 MHz–956 MHz band and uplink transmission in the 810 MHz–826 MHz band, employing duplex spacing of 130 MHz. In 1400 MHz frequency band, the downlink transmission in the 1477 MHz–1501 MHz and uplink transmission in the 1429 MHz–1453 MHz, employing duplex spacing of 48 MHz.

PDC multiplexes three time slots onto each carrier, like IS-54/IS-136 USDC standards, but has 25-kHz channel spacing to facilitate migration to PDC from the analog JTACS systems. PDC uses $\pi/4$-DQPSK modulation, also like IS-54/IS-136, with interleaving. The signaling rate is 42 kbps. Speech coding is provided with a 6.7 kbps VSELP speech coder, and channel coding is provided using convolutional coding having code rate of 9/17 and constraint length of 5 with CRC. The channel data rate is 11.2 kbps. Optionally, six users per 20-ms frame are supported with half-rate speech and channel coding. Mobile assisted handoffs are used, like in GSM and IS-54/IS-136. With a typical frequency reuse factor of 4, PDC offers relative system capacity of 6.3 times more than that of AMPS.

Key Terms

- ADPCM
- Advance Mobile Phone Service (AMPS)
- Air interface
- Bands
- Bandwidth
- Cordless
- Digital Enhanced Cordless Telephony (DECT)
- Downlink
- Frequency Division Duplexing (FDD)
- Frequency Division Multiple Access (FDMA)
- Full-duplex transmission
- Integrated Services Digital Network (ISDN)
- Interim Standard – 136 (IS-136)
- Local Multipoint Distribution Service (LMDS)
- Narrowband
- Narrow-band transmissions
- Personal Digital Cellular (PDC)
- Total Access Communications System (TACS)
- Two-way paging
- Wireless Local Loop (WLL)

Summary

In this chapter different standards and specifications of various wireless mobile communication systems and networks developed across the world have been presented. A brief description of paging and messaging systems, cellular telephone systems, cordless telephone systems, and wireless local loop including their respective worldwide standards has been given for clear understanding of the present and future wireless communication systems. The design aspects of first-generation analog cellular systems AMPS and ETACS are covered in detail. This is followed by second-generation digital cellular systems such as USDC IS-54/136 standards and Personal Digital Cellular (PDC) are included so as to understand the operation and complexity of other digital cellular systems being covered in subsequent chapters.

Short-Answer Type Questions with Answers

A10.1 Distinguish between tone pager receiver and numeric pager receiver.
The traditional subscriber pager receiver is simply a fixed-tuned wireless radio receiver that uses a transmitted code to identify received messages meant for it. The tone or beep page receivers used to beep a tone whenever they received a message. That was simply an indication to the user of the pager to make a telephone call to the central office of the Pager system to find out further details of the page. However, a numeric pager receiver decodes a short text or numeric or alphanumeric message which is usually a telephone number for the pager subscriber to make a call, but may have numerical codes for standard predefined messages.

A10.2 Why are small RF bandwidths used in transmitting messages in the paging systems
Paging messages are relatively very short, infrequent and usually require only a few seconds to transmit. Small RF bandwidths are used to maximise the signal-to-noise ratio at each paging receiver. Thus, low transmission data rates of the order of 6400 bps or less are used for maximum coverage from each paging transmitters. Many pagers can share a common assigned frequency using TDMA technique.

A10.3 What is the necessity of having a large number of cell-sites in a cellular system
In case of low power, handheld cellular mobile phones, a large number of cell-sites are required to ensure that any mobile phone is in close range to a cell-site within its service area. If the cell-site, area is not within the radio coverage range of the handheld mobile phone then a high transmitter power would be required at the mobile phone which is limited by the battery life, and therefore not recommended.

A10.4 Differentiate between LMDS and MMDS applications of WLL technology.
The Local Multipoint Distribution Service (LMDS) is capable of providing telephony, video and high data-transmission rates of the order of several Mbps, within the short range from the base station. The Multichannel Multipoint Distribution Service (MMDS) can operate in considerably larger cells within a radius of 50 km in the line-of-sight region. MMDS can be used to support two-way services as well as an alternative for broadband services such as Internet access. MMDS offers much less bandwidth than LMDS. Therefore, LMDS is useful for larger companies requiring greater bandwidths whereas MMDS is likely to be used by residential subscribers and small businesses.

A10.5 How are control messages sent during on-going voice communication in AMPS
Since there is no access to control channel during on-going voice communication, any control messages that have to be sent in such situations must use the voice channel. This is done employing in-channel out-of-band supervisory signals such as the Supervisory Audio Tone (SAT), the Signaling Tone (ST), and blank-and-burst. During blank-and-burst signaling, the voice signal is muted for about 100 milliseconds and signaling data at 10 kbps rate using FSK modulation is sent over the voice channel.

A10.6 State the criterion for controlling the mobile transmitter power in AMPS system.
In order to minimise the interference and unnecessary radiations, the mobile transmitter power is adjusted by the cell-site in 4 dB steps, with the lowest power level being –22 dBW. It is mandatory that the mobile transmitter must transmit at within 3 dB of the correct power level within 2 milliseconds of switching on the mobile equipment and must reduce its output to –60 dBm ERP or less within 2 milliseconds of being switched off.

A10.7 What are the salient features of Narrowband AMPS (N-AMPS) system
N-AMPS provides access to three mobile subscribers simultaneously in a 30-kHz AMPS channel, thereby an increase in user capacity by threefold. One N-AMPS channel uses the carrier frequency for the existing AMPS channel and, with the other two channels, the carrier frequencies are offset by ± 10 kHz. N-AMPS uses the SAT, ST, and blank-and-burst data signaling in exactly the same manner as AMPS, except the signalling is implemented by using sub-audible data signals using a continuous 200 bps NRZ encoded FSK data. When the voice channel is busy, the voice channel signalling is carried out with 100 bps Manchester encoded FSK data. N-AMPS systems are capable of providing dual-mode operation in the sense that mobile subscriber units are capable of operating either with 30-kHz channels available in standard AMPS or with 10-kHz channels in N-AMPS.

A10.8 Summarise the key differences between the first-generation analog cellular and second-generation digital cellular systems.
In first-generation cellular systems, each cell supports a number of subscribers on different frequency channels using FDMA technique. Second-generation cellular systems also provide multiple channels per cell, but each channel is dynamically shared by a number of subscribers using TDMA or CDMA technique. The first-generation cellular systems are analog using FM, whereas second-generation cellular systems are digital in terms of signal processing, providing digitised voice channels using GMSK or QPSK, data encryption, and error-correction techniques.

A10.9 What is meant by dual-mode operation in analog AMPS and the digital SDC standards and how is it achieved
The dual-mode operation specifies that a mobile subscriber complying with the IS-54 cellular standard must be capable of operating in either the analog AMPS or the digital USDC mode for voice transmissions. Using IS-54, a cellular service provider could convert any or all of its existing analog channels to digital channels by providing digital control channels and both analog and digital voice channels. The key criterion for achieving dual-mode operation is that IS-54 digital channels cannot interfere with transmissions from existing analog AMPS base stations and mobile subscribers.

A10.10 List three distinct types of control channels specified in the SDC IS-136 standard.
The USDC IS-136 standard provides three distinct types of control channels: Analog Control Channels (ACCH), 10-kbps binary FSK Digital Control Channel (DCCH), and a digital control channel with a signaling rate of 48.6 kbps on USDC-only control channels.

A10.11 What are the main functions of SPACH digital logical control channel in SDC IS-136 standard
The SPACH control logical channel can carry messages related to a single mobile subscriber unit or to a small group of mobile subscriber units. Larger messages are split into smaller blocks for transmission. The SPACH control logical channel comprises of the Short Message Service Channel (SMSCH), Paging Channel (PCH), and Access Response Channel (ARCH) which are used for control messages as well as short paging-type messages to be displayed on the mobile phone.

A10.12 Distinguish between half-rate and full rate TDMA voice signals in SDC IS-136 digital voice channel.
In USDC IS-136 digital voice channel, six voice signals, occupying one time slot each, can be accommodated with half-rate TDMA voice signals. Each voice signal is assigned to two time slots to support full rate TDMA voice signals, thereby accommodating three voice signals in one TDMA frame containing six uniform time slots. For example, Mobile user 1 occupies time slots

1 and 4, Mobile user 2 occupies time slots 2 and 5, and Mobile user 3 occupies time slots 3 and 6. A mobile transmits during two of the six time slots and receives on a different two time slots. The remaining two time slots are used by mobile subscribers to check the signal strength in adjacent cells which assist in initiating a hand-off.

A10.13 What are three supervisory channels specified in digital voice time slot of a TDMA frame in SDC IS-136 standard

Digital voice time slot of a TDMA frame contains three supervisory channels: Coded Digital Verification Colour Code (CDVCC), the Slow Associated Control Channel (SACCH), and the Fast Associated Control Channel (FACCH) for the purpose of providing synchronisation, equaliser training, and control information.

A10.14 How is signaling data transmitted in a FACCH channel

The FACCH channel does not have a dedicated time slot. It is a blank-and-burst type of transmission. It replaces digitised speech information with control and supervision messages within a subscriber's allocated time slot.

Self-Test Quiz

S10.1 In the paging system, paging messages require bandwidth.

(a) very small (b) small
(c) large (d) very large

S10.2 The pager receivers usually have

(a) extremely small size (b) long battery life
(c) low cost (d) all of the above

S10.3 The paging systems usually employ modulation technique.

(a) FM (b) FSK
(c) BPSK (d) QPSK

S10.4 The channel bandwidth of FLEX paging system is 15 kHz. The spectral efficiency for data rate of 6400 bps is bps/Hz approximately.

(a) 2.34 (b) 1.17
(c) 0.43 (d) 0.215

S10.5 The channel bandwidth in USDC/IS-54 cellular standard is

(a) 10 kHz (b) 25 kHz
(c) 30 kHz (d) 200 kHz

S10.6 The analog AMPS system requires that the received signal strength be at least above the cochannel interference to provide acceptable signal quality.

(a) 9 dB (b) 12 dB
(c) 18 dB (d) 24 dB

S10.7 The major North American Cordless Systems Standard is

(a) PACS (b) CT2
(c) DECT (d) PHS

S10.8 The AMPS system uses about transmitters at cell-sites and about 4 watts or less transmitters at the mobile subscribers.

(a) 4 watts (b) 20 watts
(c) 40 watts (d) 100 watts

S10.9 In AMPS standard, the Tx frequency of the base station for any channel is the Rx frequency of the mobile subscriber unit.

(a) greater than (b) lower than
(c) equal to (d) exactly 45 MHz away from

S10.10 USDC system offers as much as times increase in the capacity of AMPS.

(a) two (b) three
(c) four (d) six

S10.11 USDC voice channels use 4-ary $\pi/4$ DQPSK modulation with actual channel data rate of

(a) 48.6 kbps (b) 24.3 kbps
(c) 13 kbps (d) 10 kbps

S10.12 The lowest transmit power level for dual-mode mobile subscriber units is specified as

(a) –4 dBm $\pm$ 9 dB (b) 0 dBm $\pm$6 dB
(c) 4 dBm $\pm$3 dB (d) none of the above

S10.13 In USDC IS-136 standard, ______ is used in uplink transmission only.
(a) Broadcast Control Channel (BCCH)
(b) Random Access Channel (RACH)
(c) SMS Channel (SMSCH)
(d) Access Response Channel (ARCH)

S10.14 In USDC IS-136 digital voice channel, there are ______ TDMA voice frames per second.
(a) 6 (b) 8
(c) 20 (d) 25

S10.15 The Digital Verification Colour Code (DVCC) used in USDC IS-136 is the equivalent of the ______ used in the AMPS system.
(a) Supervisory Audio Tone (SAT)
(b) Signaling Tone (ST)
(c) blank-and-burst signalling
(d) none of the above

S10.16 The VSELP coders used in USDC system has a bit rate of 7950 bps and produces a speech frame every ______.
(a) 5 ms (b) 10 ms
(c) 20 ms (d) 40 ms

S10.17 The USDC system employs ______ modulation technique.
(a) BPSK (b) QPSK
(c) π/4-DQPSK (d) GMSK

Answers to Self-Test Quiz

S10.1 (a); S10.2 (d); S10.3 (b); S10.4 (c); S10.5 (c); S10.6 (c); S10.7 (a); S10.8 (d); S10.9 (c); S10.10 (d); S10.11 (a); S10.12 (a); S10.13 (b); S10.14 (d); S10.15 (a); S10.16 (c); S10.17 (c)

Review Questions

Q10.1 Mention the frequency bands utilised for various wireless communication standards such as AMPS, NMT, TACS, GSM, DECT, and PCS.

Q10.2 List the significant improvements introduced in the first, second, and third-generation standards of cellular communication systems.

Q10.3 Compare the technical parameters of US, European and Japanese cellular standards.

Q10.4 List and describe the three classifications of AMPS cellular standards.

Q10.5 Draw the block diagram of a cellular telephone communication System and explain the major functions of each block.

Q10.6 Describe and outline the frequency allocation for the Advanced Mobile Telephone Service (AMPS).

Q10.7 What are the major functions and characteristics of a SAT tone?

Q10.8 Explain the TDMA scheme used with USDC and describe the format for its digital voice channel.

Q10.9 List the advantages of digital TDMA USDC over analog FDMA AMPS cellular systems.

Q10.10 Describe the differences between the radiated power classifications for AMPS and USDC.

Q10.11 Describe the purpose of IS-54 and what is meant by dual-mode operation.

Q10.12 Describe briefly IS-136 standard and explain its relationship to IS-54 standard.

Analytical Problems

P10.1 (a) In AMPS cellular system, compute the transmitter power level (in dBW, dBm and watts) of a handheld mobile phone corresponding to 3-bit mobile attenuation code (MAC) ‘101’, transmitted from the cell-site to adjust its power.

(b) Is this power level different for a vehicle-mounted mobile phone?
(c) Express the maximum and minimum power levels for each class of AMPS mobile phone in milliwatts.

P10.2 (a) Assume that the range of AMPS mobile phone is limited by its transmitter power. If a hand-held phone (Class III) can communicate over a distance of 5 km, what would be the maximum range for a permanently installed vehicle-mounted mobile phone (Class I) under the same circumstances. Assume free-space attenuation.
(b) Repeat Part (a), but assume that signal attenuation is proportional to the fourth power of distance. This is more typical of a mobile environment.

P10.3 Calculate the maximum distance between cell-site and mobile operating in a IS-54 digital cellular system if the mobile instructs its serving cell-site to advance its transmission by up to 3 bit periods. A cell-site and mobile are separated by 5 km. What is the propagation time for a signal traveling between them?

P10.4 Draw the TDMA voice time slots showing the distribution of bits in forward channel and reverse channel in IS 54 digital cellular system. Compute the TDMA frame efficiency.

P10.5 Suppose the Short Message Service Channel (SMSCH) in a IS-136 TDMA digital cellular system occupies five time slots in each superframe.
(a) Compute the total bit rate available for short messages.
(b) If the five mobile subscribers in the cell or a sector have messages addressed to them at the same time, what is the bit rate available for each subscriber?

P10.6 Draw the allocation of bits in a USDC half-rate time slot. Compute the following:
(a) The channel data rate for the USDC air interface
(b) Number of user bits in each USDC time slot
(c) The time duration for each USDC frame
(d) The frame efficiency for USDC

P10.7 For a full-rate USDC system, consider how the data is allocated between channels. Compute the following parameters:
(a) The gross RF data rate
(b) The individual gross RF data rate for the SACCH, the CDDVC, synchronisation, ramp-up, and guard time.

Is sum of individual gross data rate as calculated in Part (b) same as that of calculated in Part (a)? What is the end-user data rate provided in full-rate USDC?

P10.8 The US Digital Cellular TDMA system uses a 48.6 kbps data rate to support three users per frame. Each user occupies two of the six time slots per frame. What is the raw data rate provided for each subscriber?

P10.9 In the US Digital Cellular TDMA system, assume each reverse channel frame contains six time slots with 324 bits per time slot, and within each time slot, there are six guard bits, six bits reserved for ramp-up, 28 synchronisation bits, 12 control channel bits, 12 bits for supervisory control signals, and 260 data bits.
(a) Determine the frame efficiency for the US Digital Cellular standard.
(b) If half-rate speech coding is used, then six users may be supported within a frame. Determine the raw data rate and frame efficiency for users with half-rate speech coding.

P10.10 In AMPS cellular system, a mobile phone is using a digital modem at 9600 bps. Calculate the number of bits that will be lost during
(a) blank-and-burst signaling
(b) handoff, if it results in a 0.5 s loss of signal

P10.11 (a) Find the propagation-path loss in a typical urban environment in the handheld mobile phone frequency range of 850 MHz if the cell-site antenna has a height of 30 m and the distance of the mobile from its serving cell-site is 5 km. (Use the Hata Model.)
(b) Calculate the power delivered to the cell-site receiver under the same circumstances if the cell-site receiving antenna has a gain of 6 dBi and the receiver feedline has a loss of 2 dB.

(c) If the handheld mobile phone described in Part (a) is used inside a vehicle, which causes an additional loss of 20 dB, what would be the effect on the answers for parts (a) and (b)?

P10.12 (a) Neglecting the frequency spectrum used for control channels, what is the maximum number of two-way voice channels that can fit inside the frequencies allocated to the AMPS system?

(b) What is the number of channels in each cell? Note that $K = 7$ is originally used in the AMPS.

(c) Repeat (b) for IS-136 TDMA system in which $K = 4$ and the number of slots per TDMA channel is three.

P10.13 An analog FM AMPS system is deployed with half channel bandwidth of 15 kHz rather than the existing 30 kHz. Also, assume that in analog FM, the carrier-to-interference ratio (C/I) requirement is inversely proportional to the square of the bandwidth (4 time increase in C/I for dividing the channel bandwidth into two).

(a) What is the required C/I in dB for the 15 MHz uplink or downlink spectra if the required C/I for the 30 kHz systems is 18 dB?

(b) Determine the frequency reuse factor K needed for the implementation of this 15 kHz per user analog cellular system.

(c) If a service provider is allocated a 12.5-MHz band in each direction (uplink and downlink) and it would install 30 antenna sites to provide its service, what would be the maximum number of simultaneous subscribers that the system could support in all cells? Assume all available channels are used for carrying traffic.

(d) If the same antenna sites are used a 30 kHz per channel system with $K = 7$ (instead of the 15 kHz system), what would be the capacity of the new system?

P10.14 Considering the frequency allocation strategy of the IS-136 systems, determine

(a) the total number of traffic channels per 50 MHz of bandwidth used for two-way communications

(b) the total number of channels per MHz of bandwidth

(c) the number of channels per cell for frequency reuse factors of $K = 4$ and $K = 7$

P10.15 Determine the transmit and receive carrier frequencies for AMPS base station and mobile unit corresponding to

(a) AMPS channel 3

(b) AMPS channel 333

(c) AMPS channel 799

(d) AMPS channel 991

P10.16 If a total of 25 MHz of bandwidth is allocated to a particular FDD cellular telephone system which uses two 25-kHz simplex channels to provide full-duplex voice and control channels, compute the number of full-duplex channels available per cell if a system uses

(a) four-cell reuse cellular pattern

(b) seven-cell reuse cellular pattern

(c) 12-cell reuse cellular pattern

(d) If 1 MHz of the allocated spectrum is dedicated to control channels, determine an equitable distribution of control channels and voice channels in each cell for each of the three systems in parts (a), (b), and (c) above.

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GSM (Global System for Mobile communications or Groupe Speciale Mobile) communications, initiated by the ETSI (European Telecommunications Standardisation Institute), is the second-generation pan-European digital mobile cellular communication standard with international roaming. It is used in the 900-MHz cellular band as well as 1800-MHz PCS band in Europe and most parts of Asia, whereas in the 1900-MHz PCS band in North America. This chapter gives a detailed description of GSM which includes architectures and interfaces of GSM systems and subsystems, signaling protocol architecture, logical channels and frame structure, GSM security aspects, and various services offered by the system.

11

Global System for Mobile (GSM)

11.1 GSM NETWORK ARCHITECTURE

GSM uses two 25-MHz frequency bands, that is, the 890-MHz to 915-MHz band is used for mobile subscriber unit to base station transmissions (reverse-link transmissions), and the 935-MHz to 960-MHz frequency band is used for base station to mobile subscriber unit transmission (forward-link transmissions). GSM uses Frequency-Division Duplexing (FDD) and a combination of TDMA and FDMA techniques to provide simultaneous access to multiple mobile subscriber units.

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The GSM system has an allocation of 50 MHz in the 900-MHz frequency band. Using FDMA, this band is divided into 124 RF channels, each with a carrier channel bandwidth of 200 kHz. Using TDMA, each of these carrier channels is further divided into 8 time slots. Thus, with the combination of FDMA and TDMA, a maximum of 992 channels can be realised for transmit and receive.

The GSM network architecture consists of three major subsystems:

– Mobile Station (MS)
– Base Station Subsystem (BSS)
– Network and Switching Subsystem (NSS)

The wireless link interface is between the MS and the Base Transceiver Station (BTS), which is a part of BSS. Many BTSs

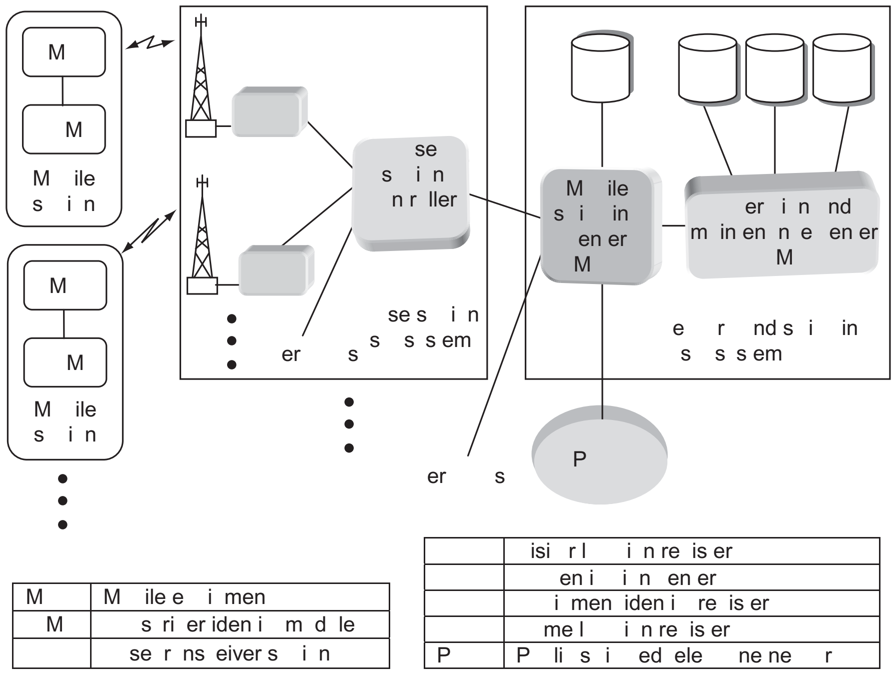

Fig. 11.1 *GSM network architecture*

are controlled by a Base Station Controller (BSC). BSC is connected to the Mobile Switching Center (MSC), which is a part of NSS. Figure 11.1 shows the key functional elements in the GSM network architecture.

11.1.1 Mobile Station (MS)

A mobile station communicates across the air interface with a base station transceiver in the same cell in which the mobile subscriber unit is located. The MS communicates the information with the user and modifies it to the transmission protocols of the air-interface to communicate with the BSS. The user's voice information is interfaced with the MS through a microphone and speaker for the speech, keypad, and display for short messaging, and the cable connection for other data terminals. The MS has two elements. The Mobile Equipment (ME) refers to the physical device, which comprises of the transceiver, digital signal processors, and the antenna.

The second element of the MS in the GSM is the Subscriber Identity Module (SIM) that is a smart card issued at the subscription time identifying the specifications of a user such as a unique number and the type of service. The SIM card is unique to the GSM system. It is about postage-stamp size with 32 k bytes of memory that can be plugged into any GSM mobile phone. From the user's point of view, one of the most remarkable features of GSM is the SIM card, which is a portable device in the form of a smart card or plug-in module memory device that stores information such as the subscriber's identification number, privacy keys, the cellular networks and regions where the subscriber is authorised to service, and other user-specific information.

The GSM subscriber units are totally generic until a SIM is inserted. Therefore, a subscriber need only to carry a SIM card to use a wide variety of mobile equipments simply by inserting the SIM in the device to be used. In fact, except for certain emergency communications, the subscriber units will not work without a SIM inserted. Thus, the mobile equipment does not roam, it is the SIM which roams. The calls in the GSM are directed to the SIM inserted in any mobile phone. Short messages are also stored in the SIM card.

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The Mobile Station is the technical name of the mobile or the cellphone. Although modern cellphones have become smaller and lighter, they are still called Mobile Stations as these were called so in early days due to their being bulky and sometimes installed in vehicles.

Facts to Know!

The 32 kbytes SIM card has inherent security features that make wireless transactions more secure than conducting transactions over the Internet.

The SIM card allows a mobile subscriber to use any GSM mobile phone anywhere in the world where GSM services are available. Alternatively, people visiting different GSM-enabled countries that are not keen on making calls at their home number can always carry their own mobile phone and purchase a SIM card in any other country. This way they avoid roaming charges and the expense of having a different contact number. Several users can also share a mobile phone with different SIM cards.

Because SIM cards carry the private information for a user, a security mechanism is implemented in the GSM that asks for a four-digit PIN number to make the information on the SIM card available to the user. The SIM card also offers some protection against fraudulent use. A GSM mobile phone is useless without a SIM.

For example, if the mobile subscriber removes the SIM card when leaving the mobile phone in a vehicle, the mobile phone cannot be used unless another person has a valid SIM. Unfortunately, the SIM cards can be stolen too. The SIM can be set up to require the user to enter a Personal Identification Number (PIN) whenever the mobile phone is switched on to provide some security in case the card is lost or stolen.

Once a mobile phone user has a valid SIM, buying a new GSM mobile phone is easy. No set-up or programming is required. Similarly, a user can have a permanently vehicle-installed mobile phone and a handheld mobile phone with the same phone number, provided that only one is used at a time. The phone number of a mobile subscriber is usually of 10–15 digits. The first three digits are the country code; the next two are the digits for the specific MSC, and the rest are the telephone number. The IMSI of the same user is totally different from the ISDN telephone number. The first three digits of the IMSI identify the country, and the next two digits, the service provider.

Besides the SIM card, the next most remarkable feature of GSM is the on-the-air privacy which is provided by the system. Unlike analog FM cellular phone systems which can be readily monitored, it is virtually impossible to eavesdrop on a GSM radio transmission. The privacy is made possible by encrypting the digital bit stream sent by a GSM transmitter, according to a specific secret cryptographic key which changes with time for each user that is known only to the service provider.

EXAMPLE 11.1 GSM air-interface standards

List the important GSM air-interface standard specifications.

Solution

The GSM standard specifications are given in Table 11.1.

Table 11.1 *GSM standard air-interface specifications*

S. No.	*Parameter*	*Specification*
1.	Frequency band	*Uplink* 890-MHz to 915-MHz *Downlink* 935-MHz to 960-MHz
2.	Spectral allocation	50-MHz (25 MHz each for uplink and downlink)
3.	Forward and reverse channel frequency spacing	45-MHz
4.	Tx/Rx time slot spacing	3 time slots
5.	RF channel bandwidth	200-kHz (ARFCN channel spacing)

(*Continued*)

Table 11.1 (*Continued*)

S. No.	*Parameter*	*Specification*
6.	ARFCN number	0 to 124 and 975 to 1023
7.	Multiple-access technique	TDMA/FDMA
8.	Duplexing technique	FDD
9.	Modulation scheme	GMSK ($B \times T_b = 0.3$)
10.	Number of time-slots per RF channel bandwidth	8 (users per frame full rate)
11.	Number of voice channels	1000
12.	Modulation data rate	270.833 kbps
13.	Spectrum efficiency	1.35 bps/Hz
14.	Frame period	4.615 ms
15.	Time slot period	577 μs
16.	Bit period	3.692 μs
17.	Interleaving (max. delay)	40 ms
18.	Speech coding	RELP-LTP @ 13.4 kbps
19.	Channel coding	CRC with r = ; L = 5 convolutional coding
20.	Type of equalisers	Adaptive
21.	Handheld mobile Tx power	1 W max; 125 mW avg.

11.1.2 Base Station Subsystem (BSS)

A base station subsystem consists of a base station controller and one or more base transceiver stations. Each Base Transceiver Station (BTS) defines a single cell. A cell can have a radius of between 100 m and 35 km, depending on the environment. A Base Station Controller (BSC) may be collocated with a BTS. It may control multiple BTS units and hence multiple cells. The BSC reserves radio frequencies, manages the hand-off of a mobile unit from one cell to another within the BSS, and controls paging. The BSS manages the radio interface between the mobile stations and all other subsystems of GSM such as MSC.

The BSS translates between the wireless-interface and fixed wired infrastructure protocols. The needs for the wireless and wired media are different because the wireless medium is unreliable, bandwidth limited, and needs to support mobility. As a result, protocols used in the wireless medium and wired medium are different. The BSS provides for the translation among these protocols.

There are two main architectural elements in the BSS—the Base Transceiver Subsystem (BTS), and the Base Station Controller (BSC). The BSS consists of many BSCs which connect to a single MSC, and each BSC typically controls up to several hundred BTSs. Some of the BTSs may be co-located at the BSC, and others may be remotely distributed and physically connected to the BSC by microwave links or dedicated leased lines. The BTS is the counterpart of the MS for physical communication over the air-interface. The BTS components include a transmitter, a receiver, and signaling equipment to operate over the air-interface, and it is physically located in the centre of the cells where the BSS antenna is installed. One BSS may have from one up to several hundred BTSs under its control. The hand-offs of calls between two BTSs under the control of the same BSC are handled by the BSC, and not the MSC. This greatly reduces the switching burden of the MSC.

The interface that connects a BTS to a BSC is called the A-bis interface. The A-bis interface carries traffic and maintenance data. The main function of the BSC is to look over a certain number of BTSs to ensure proper operation. The BSC is a small switch inside the BSS in charge of frequency administration, maintains appropriate power levels of the signal and hand-off among the BTSs inside a BSS. The hardware of the BSC in a single BTS site is located at the antenna and in the multi-BTS systems, in a switching centre where other hardware elements

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In a metropolitan city, a large number of BTSs are potentially deployed. A BTS is usually placed in the centre of a cell and its transmitting power defines the size of a cell.

of NSS are located. The interface between a BSC and an MSC is called the *A interface*, which is standardised within GSM.

The user's speech signal is converted into 13 kbps-digitised voice with a speech coder and communicated over the air-interface to provide a bandwidth efficient air-interface. The backbone wired network uses a 64 kbps PCM digitised voice in the PSTN hierarchy. Conversion from analog speech signal to 13 kbps digitised voice signal takes place at the mobile station, and the change from 13 kbps to 64 kbps coding takes place at the BSS. The call is established through the exchange of a number of packets.

11.1.3 Network and Switching Subsystem (NSS)

The NSS is responsible for the network operation. It provides the link between the cellular network and the Public Switched Telecommunications Networks (PSTN or ISDN or Data Networks). The NSS controls hand-offs between cells in different BSSs, authenticates users and validates their accounts, and includes functions for enabling worldwide roaming of mobile subscribers. The NSS could be interpreted as a wireless specific switch that communicates with other switches in the PSTN and at the same time supports functionalities that are needed for a cellular mobile environment. The NSS interconnects to the PSTN through ISDN protocols. The NSS provides communications with other wired and wireless networks, as well as support for registration and maintenance of the connection with the MSs via BSCs in the radio subsystem.

The network and the switching subsystem together include the main switching functions of GSM as well as the databases needed for subscriber data and mobility management. In particular, the switching subsystem consists of

– Mobile Switch Centre (MSC)
– Home Location Register (HLR)
– Visitor Location Register (VLR)
– Authentication Centre (AuC)
– Equipment Identity Register (EIR)
– Interworking Function (IWF)

The NSS is the most elaborate element of the GSM network, and it has one hardware, Mobile Switching Centre (MSC), and four software database elements: Home Location Register (HLR), Visitor Location Register (VLR), Equipment Identification Register (EIR), and Authentication Centre (AuC). An MSC is the hardware part of the wireless switch that can communicate with PSTN switches using the signaling system-7 (SS-7) protocol, as well as other MSCs in the coverage area of a service provider. If the MSC has an interface to the PSTN then it is called a Gateway MSC (GMSC). The MSC also provides the network the specific information on the status of the mobile terminals. The MSC basically performs the switching functions of the system by controlling calls to and from other telephone and data systems. It also does functions such as network interfacing and common channel signaling.

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NSS acts like a normal switching node for mobile subscribers of the same GSM network or for the PSTN fixed telephone or for ISDN, and provides all the functionality such as registration, authentication, location updating, handovers and call routing to handle a mobile subscriber.

Because the GSM represents an independent network, it must dispose of entities which provide connection to other users. Therefore, the main component of the switching subsystem is the Mobile Switching Centre, MSC. The main role of the MSC is to manage the communications between the GSM users and other telecommunications network users. The basic switching function is performed by the MSC, whose main function is to coordinate setting up calls to and from GSM users. The MSC has interfaces with the BSS on one side (through which MSC VLR is in contact with GSM users) and the external networks on the other side (ISDN/PSTN). An MSC is generally connected to several BSSs, which provide radio coverage to the MSC area. The MSC is also connected to other GSM Public Land Mobile Network (PLMN) entities such as other MSCs and HLR through a fixed network.

The MSC is the telephone switching office for mobile-originated or terminated traffic. The MSC controls the call set-up and routing procedures in a manner similar to the functions of a land network end office. The MSC provides call set-up, routing, and handover between BSCs in its own area and to/from other MSCs; an interface to the fixed PSTN; and other functions such as billing. It also performs such functions as toll ticketing, network interfacing, common channel signaling, and others.

The HLR is database software that handles the management of the mobile subscriber account. It stores the subscriber's address, service type, current location, forwarding address, authentication/ciphering keys, and billing information. In addition to the ISDN telephone number for the terminal, the SIM card is identified with an International Mobile Subscriber Identity (IMSI) number that is totally different from the ISDN telephone number. The IMSI is used totally for internal networking applications. Each HLR is identified by the HLR number which is sent to all the required VLRs. The HLR is the reference database that permanently stores data related to subscribers, including a subscriber's service profile, location information, and activity status. When an individual user buys a subscription from one of the GSM service providers, he is registered in the HLR of that service provider.

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When a cellphone is powered off, the administrative information such as current location of the mobile, all the service provisioning information and authentication data is stored in the HLR.

Various identification numbers and addresses as well as authentication parameters, services subscribed, and special routing information are stored in the HLR. Current subscriber status, including a subscriber's temporary roaming number and associated VLR if the mobile is roaming, are maintained. Location registration is performed by HLR. The HLR provides data needed to route calls to all MS-SIMs home based in its MSC area, even when they are roaming out of area or in other GSM networks. The HLR provides the current location data needed to support searching for and paging the MS-SIM for incoming calls, wherever the MS-SIM may be. The HLR is responsible for storage and provision of SIM authentication and encryption parameters needed by the MSC where the MS-SIM is operating. It obtains these parameters from the AuC. The HLR maintains records of which supplementary services each user has subscribed to and provides permission control in granting access to these services.

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Although an HLR may be implemented as a distributed database, yet there is logically one HLR per GSM network.

Based on described functions, different types of data are stored in the HLR. Some data are permanent; that is, they are modified only for administrative reasons, while others are temporary and modified automatically by other network entities depending on the movements and actions performed by the subscriber. Some data are mandatory, other data are optional. Both the HLR and the VLR can be implemented in the same equipment in an MSC (collocated). A PLMN may contain one or several HLRs.

The VLR is a temporary database software similar to the HLR identifying the mobile subscribers visiting inside the coverage area of an MSC. The VLR assigns a Temporary Mobile Subscriber Identity (TMSI) that is used to avoid using IMSI on the air. The location of the mobile subscriber is determined by the VLR into which the mobile subscriber is entered. The visitor location register maintains information about mobile subscribers that are currently physically in the region covered by the switching centre. It records whether or not the subscriber is active and other parameters associated with the subscriber. For a call coming to the mobile subscriber, the system uses the mobile phone number associated to identify the home switching centre of the mobile subscriber. The home switching centre can find in its HLR the switching centre in which the mobile subscriber is presently located. For a call coming from the mobile subscriber, the VLR is used to initiate the call. Even if the mobile subscriber is in the area covered by its home switching centre, it is also represented in the switching centre's VLR.

A VLR is linked to one or more MSCs. The function of the VLR is to memorise temporarily information about the mobiles which are currently located in the geographical area controlled by the linked MSC. The VLR is a database that contains temporary information about subscribers that is needed by the MSC in order to service visiting subscribers. The VLR supports a mobile paging-and-tracking subsystem in the local area where the mobile is presently roaming. The VLR is always integrated with the MSC. A VLR may be in charge of one or several MSC LAs (Location Areas).

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The VLR can be considered as a temporary copy of some of the important information (selected administrative information necessary for call control and provisioning of the subscribed services) stored in the HLR. HLR is the persistent storage whereas VLR is similar to a cache.

When a mobile subscriber roams from one LA to another, their current location is automatically updated in their VLR. When a mobile station roams into a new MSC area, if the old and new LAs are under the control of two different VLRs, the VLR connected to that MSC will request data about the mobile station from the HLR. The entry on the old VLR is deleted and an entry is created in the new VLR by copying the basic data from the HLR. Later, if the mobile station makes a call, the VLR will have the information needed for call set-up without having to interrogate the HLR each time. The subscriber's current VLR address, stored at the HLR, is also updated. This provides the information necessary to complete calls to roaming mobiles. These two databases, HLR and VLR, are used to keep track of the current location of an MS in GSM. Maintenance of two databases at home and at the visiting location allows a mechanism to support dialing and call routing in a roaming situation where the MS is visiting the coverage area of a different MSC.

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The HLR is always fixed and remains in the home network, whereas the VLR logically moves with the subscriber. For example, if a subscriber of a GSM network in Mumbai is roaming in Kolkata, the HLR data of the subscriber will remain in Mumbai with the home network; however, the VLR data will be copied to the roaming network in Kolkata. This enables the subscribed services without needing to refer to the HLR each time a communication is established. Though the visiting network in Kolkata will provide the user services, the billing will be done by the home network in Mumbai.

GSM transmission is encrypted. The AuC database holds different algorithms that are used for authentication and encryption of the mobile subscribers that verify the mobile user's identity and ensure the confidentiality of each call. The AuC protects network cellular operators from different types of frauds and spoofing found in today's cellular world. AuC holds the authentication and encryption keys for all the subscribers in both the home and visitor location registers. A stream cipher, A5, is used to encrypt the transmission from subscriber to base transceiver. However, the conversation is in the clear in the landline network. Another cipher, A3, is used for authentication. Different classes of SIM cards have their own algorithms, and the AuC collects all of these algorithms to allow the NSS to operate with different mobile terminals from different geographic areas.

The EIR is another database that keeps the information about the identity of mobile equipment such as the International Mobile Equipment Identity (IMEI) that reveals the details about the manufacturer, country of production, and device type. This information is used to prevent calls from being misused, to prevent unauthorised or defective MSs, to report stolen mobile phones or check if the mobile phone is operating according to the specification of its type.

Each mobile equipment is identified by IMEI which is memorised by the manufacturer and cannot be removed. By the registration mechanism the MS always sends the IMEI to the network, so that the EIR can memorise and assign them to three different lists:

White List This list contains the IMEI of the phones who are allowed to enter in the network.

Black List This list on the contrary contains the IMEI of the phones who are not allowed to enter in the network, for example because they are stolen. Those phones are not able to enter in all the GSM networks which dispose of an EIR.

Grey List This list contains the IMEI of the phones momentarily not allowed to enter in the network, for example because the software version is too old or because they are in repair.

By the registration mechanism, the MSC checks if the MS is contained in the black or grey list; if so, the mobile cannot enter the network. One EIR per GSM network is enough. In the future there will be an interconnection between all the EIRs to avoid the situation where a mobile stolen in one country can be used in a GSM network from a different country. Both AuC and EIR can be implemented as individual stand-alone nodes or as a combined AuC/EIR node. The implementation of the EIR is left optional to the service provider.

IWF-Interworking Function It is a subsystem in the PLMN that allows for non-speech communication between the GSM and the other networks. The tasks of an IWF are particularly to adapt transmission parameters and protocol conversion. The physical manifestation of an IWF may be through a modem which is activated by the MSC dependent on the bearer service and the destination network.

The SS supports operation and maintenance of the system and allows engineers to monitor, diagnose, and troubleshoot every aspect of the GSM network. The OSS supports one or several Operation Maintenance Centres (OMC) that are used to monitor and maintain the performance of each MS, BS, BSC, and MSC within a GSM system. The OSS has three main functions, which are to maintain all telecommunications hardware and network operations with a particular service area, manage all mobile equipment in the system, and manage all charging and billing procedures. Within each GSM system, an OMC is dedicated to each of these tasks and has provisions for adjusting all base-station parameters and billing procedures, as well as for providing system operators with the ability to determine the performance and integrity of each unit of mobile subscriber equipment in the system.

11.2 GSM SIGNALING PROTOCOL ARCHITECTURE

Figure 11.2 shows the signaling protocol architecture for communication between the main hardware elements of the GSM network architecture and the associated interfaces.

The GSM standard specifies the interfaces among all the elements of the architecture. The air-interface 'U_m', which specifies communication between the MS and BTS, is the wireless related interface. Messages between the BTS and BSC flow through the A-bis interface. The support on this interface is for voice traffic at 64 kbps and data/signaling traffic at 16 kbps. Both types of traffic are carried over LAPD (which is a data link protocol used in ISDN).

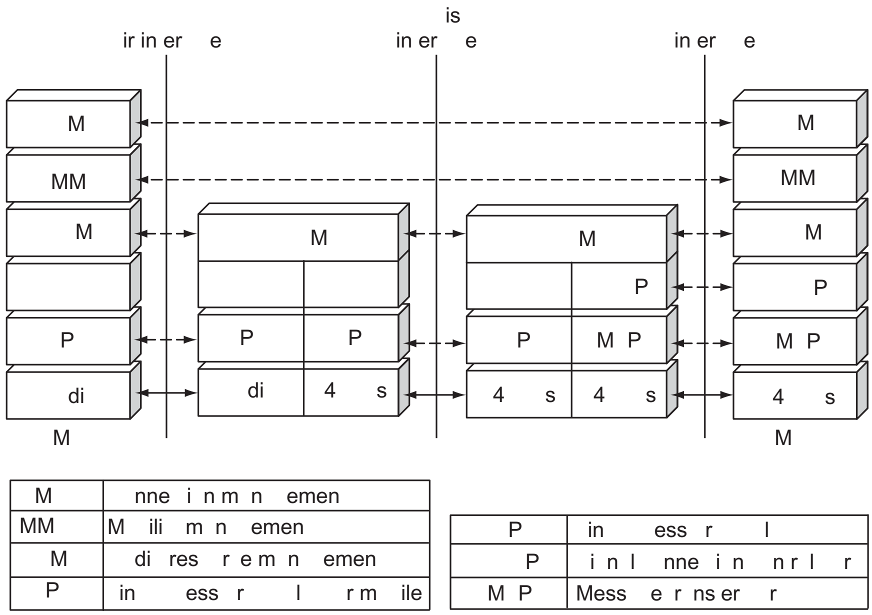

Fig. 11.2 | *The GSM signaling protocol architecture*

The interface between a BSC and a MSC is called the A' interface, which is standardised within GSM. The A' interface uses an SS7 protocol called the Signaling Connection Control Part (SCCP) which supports communication between the MSC and the BSS, as well as network messages between the individual mobile subscribers and the MSC. The A' interface allows a service provider to use base stations and switching equipment made by different manufacturers.

A number of control messages are exchanged between the key entities of GSM network architecture that deal with radio resources, mobility management, and connection management. The protocol stack is divided into three layers:

Layer 1 Physical Layer
Layer 2 Data Link Layer (DLL)
Layer 3 Networking or Messaging Layer

11.2.1 Layer I: Physical Layer

The physical layer defined in the GSM specifications is for the U_m air-interface. The radio link carries higher-level data inside the TDMA format between the mobile station and the base transceiver station. This layer specifies how the information from different voice and data services are formatted into packets and sent through the radio channel. It specifies the radio modem details, the packaging of a variety of services into the bits of a packet, traffic structure and control packets. This layer specifies modulation and coding techniques, power control methodology, and time synchronisation approaches which enable establishment and maintenance of the channels. The physical layer of the A and A-bis interfaces follow the ISDN standard with 64 kbps digital data per voice user.

11.2.2 Layer II: Data Link Layer

The control and signaling data transfer may be through the same physical channels or through separate physical channels. Signaling and control data are conveyed through Layer II and Layer III messages. At the link layer, a data link control protocol known as LAPD$_m$ is used where *m* refers to the modified version of LAPD adapted to the mobile environment. In essence, LAPD is designed to convert a potentially unreliable physical link into a reliable data link. It does this by using a cyclic redundancy check to perform error detection and Automatic Repeat Request (ARQ) to retransmit damaged frames. The LAPD protocol is used for the A-bis and A interfaces connecting the BTS to BSC and BSC to MSC, respectively.

The overall purpose of DLL is to check the flow of packets for Layer III and allow multiple Service Access Points (SAP) with one physical layer. The remaining links use the normal LAPD protocol. The DLL checks the address and sequence number for Layer III and manages acknowledgments for transmission of the packets. In addition, the DLL allows two SAPs for signaling and Short Messages (SMS). The SMS traffic channel in the GSM is not communicated through voice channels. In GSM, the SMS is transmitted through a fake signaling packet that carries user information over signaling channels. The DLL in GSM provides this mechanism for multiplexing the SMS data into signaling streams.

Signaling packets delivered to the physical layer are each 184 bits, same as that of the length of the DLL packets in the LAPD protocol used in the ISDN networks. The length of the LAPD$_m$ packets, shown in Fig. 11.3, is the same as LAPD, but the format is slightly adjusted to fit the mobile environment.

1 4 is ixed				
ddress ield is	n r l ield is	en indi r is	n rm i n is v ri le	ill in is v ri le

Fig. 11.3 *Frame format of the Layer II in LAPD$_m$*

Since GSM has the time synchronisation and strong coding at the physical layer, the synchronisation bits and CRC codes in LAPD are eliminated in the LAPD$_m$. The address field is optional, and it identifies the SAP, protocol revision type, and nature of the message. The control field is also optional, and it holds the type of the frame (command or response) and the transmitted and received sequence numbers.

The length indicator identifies the length of the information field. Fill-in bits are all 1s bits to extend the length to the desired 184 bits. In peer-to-peer Layer II communications, such as DLL acknowledgments, there is no Layer III payload and fill-in bits cover this field. The information field carries the Layer III payload data.

The peer-to-peer Layer II messages are *unnumbered acknowledgment, receiver ready, receiver not ready, disconnect*, and *reject*. These messages do not have Layer III information bits and are referred to as Layer II messages. The information bits in Layer II packets specify Layer III operations implemented on the logical signaling channels. These information bits are different for different operations.

11.2.3 Layer III: Networking or Signaling Layer

The networking or signaling layer implements the protocols needed to support the mechanisms required to establish, maintain, and terminate a mobile communication session. It is also responsible for control functions for supplementary and SMS services. The traffic channels are carried by normal bursts in different formats associated with different speech or data services. The signaling information uses other bursts and more complicated DLL packaging. A signaling procedure such as the registration process is composed of a sequence of communication events or messages between hardware elements of the systems that are implemented on the logical channels encapsulated in the DLL frames.

Layer III defines the details of implementation of messages on the logical channels encapsulated in DLL frames. Among all messages communicated between two elements of the network only a few, such as DLL acknowledgment, do not carry Layer III information. Information bits of the Layer II packets specify the operation of a Layer III message. As shown in Fig. 11.4, these bits are further divided into several fields.

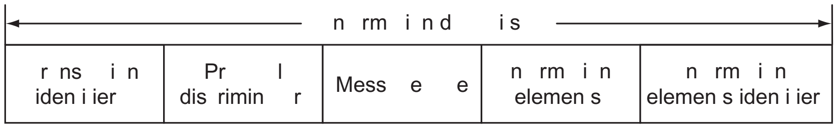

Fig. 11.4 *Typical Layer III message format*

The Transaction Identifier (TI) field is used to identify a procedure or protocol that consists of a sequence of messages. This field allows multiple procedures to operate in parallel. The Protocol Discriminator (PD) identifies the category of the operation (management, supplementary services, call control, and test procedure). The Message Type (MT) identifies the type of message for a given PD. Information Elements (IE) is an optional field for the time that an instruction carries some information that is specified by an IE Identifier (IEI). The number of Layer III messages is much larger than the number of Layer II messages.

To further simplify the description of the Layer III messages, GSM standard divides the messages into three sublayers that provide specific functions:

– Radio Resource Management (RRM)
– Mobility Management (MM)
– Communication Management (CM)

The RRM sublayer of Layer III manages the frequency of operation and the quality of the radio link. Radio resource management establishes and releases connections between MSs and an MSC and maintains them despite subscriber movements. The RRM functions are mainly performed by the MS and the BSC. The main responsibilities of the RRM are to assign the radio channel and hop to new channels in implementation of the slow frequency-hopping option, to manage hand-off procedure and measurement reports from MS for hand-off decision, to implement power control procedure, and to adapt to timing advance for synchronisation.

The major functions of Mobility Management (MM) sublayer are location update, registration procedures, authentication procedure, TMSI handling, and attachment and detachment procedures for the IMSI. This sublayer handles mobility issues that are not directly related to the radio, and include management of security functions. Mobility management functions are handled by the MS/SIM, the MSC/VLR, and the HLR/AuC.

The Communication Management (CM) sublayer is used to establish, maintain, and release the circuit-switched connection between the calling and called subscribers of GSM network. Specific procedures for the CM sublayer include mobile-originated and mobile-terminated call establishment, change of transmission mode during the call, control of dialing using dual-tones, and call reestablishment. In addition to call management, it includes supplementary services management and SMS management.

The Mobile Application Part (MAP) handles most of the signaling between different entities in the fixed part of the network, such as between the HLR and VLR. It runs on top of two intermediate protocols—Signal Connection Control Part (SCCP) and Message Transfer Part (MTP). SCCP and MTP protocols are part of Signaling System Number 7, which is a set of protocols designed to provide control signaling within digital circuit-switching networks.

Facts to Know!

SS7 is also used for many other Intelligent Network Services (like virtual calling card service and local number portability) within the GSM.

11.2.4 SS7 Signaling

Common Channel Signaling System No. 7 (SS7 or CC7) is a global standard that defines the procedures and protocol by which network elements in PSTN exchange information over a digital signaling network to effect wireless (cellular) and wireline call set-up, routing and control. The SS7 signaling protocols are mainly used for basic call set-up, call management, wireless services such as PCS, wireless roaming, mobile subscriber authentication, local number portability, toll-free and toll wireline services, enhanced call features such as call forwarding, calling party name/number display, and three-way calling, efficient and secure worldwide telecommunications. The SS7 protocol provides both error correction and retransmission capabilities to allow continued service in the event of signaling point or link failures.

SS7 messages are exchanged between network elements over 64 kbps bi-directional channels called signaling links. Signaling occurs out-of-band on dedicated channels rather than in-band on voice channels. Compared to in-band signaling, out-of-band signaling provides faster call set-up times, more efficient use of voice circuits, support for Intelligent Network (IN) services which require signaling to network elements without voice trunks (for example, database systems), and improved control over fraudulent network usage. There are three kinds of signaling points in the SS7 network:

– Service Switching Point (SSP)
– Signal Transfer Point (STP)
– Service Control Point (SCP)

SSPs are switches that originate, terminate, or tandem calls. An SSP sends signaling messages to other SSPs to set up, manage, and release voice circuits required to complete a call. An SSP may also send a query message to a centralised database, an SCP, to determine how to route a call. An SCP sends a response to the originating SSP containing the routing number(s) associated with the dialed number. An alternate routing number may be used by the SSP if the primary number is busy or the call is unanswered within a specified time. Actual call features vary from network to network and from service to service.

Network traffic between signaling points may be routed via a packet switch called STP. The STP routes each incoming message to an outgoing signaling link based on routing information contained in the SS7 message. Because it acts as a network hub, STP provides improved utilisation of the SS7 network by eliminating the need for direct links between signaling points. The STP may perform global title translation, a procedure by which the destination signaling point is determined from digits present in the signaling message. The STP can also act as a firewall to screen SS7 messages exchanged with other networks. Because the SS7 network is critical to call processing, SCPs and STPs are usually deployed in pair configurations in separate physical locations to ensure network-wide service in the event of an isolated failure.

Facts to Know!

SCPs provide the access mechanism required for a service. These are used for a variety of applications such as calling card verification, toll-free calls, and premium tariff calls.

11.2.5 Addressing and Routing

Within the GSM network, two types of routing can be described:

– SS7 addressing and message signaling routing
– Call control/number routing

The SS7 MTP layer 3 provides the routing function. This layer is used to route within a local network using the Signaling Point Code addressing. To interconnect all the local networks on the national SS7 networks, the SCCP Global Title Translation (GTT) functionality is used. Global Title Translation is one of the strong routing capabilities of SS7 SCCP layer. This SCCP functionality allows a centralised network to hold and maintain all the addresses and routing tables, centralising the routing function. For MSC to send a message to a particular HLR, the MSC does not need to know each Mobile's HLR point code. Only the adjacent STP point code and the dialed digits (MSISDN) needs to be provided to the STP in order to route the message to the HLR. The STP will perform the translation of the dialed digits to physical point code (HLR or MSC).

The STP pair after checking the SCCP header information will determine if the message requires GTT translation. It will then extract the IMSI of the subscriber from the calling number address field in the SCCP header

> **Facts to Know!**
>
> STP is similar to a switch or a node in the SS7 signaling network performing the basic routing functions. For fault tolerance, STPs are always installed in pairs with cross-connections.

and from a database table determine the HLR point code where the validation/authentication should be sent. This will eliminate book-keeping on every MSC and centralise the routing/translation on the SS7 STP network.

A landline calling party dials the GSM mobile directory number (MS ISDN number). The PSTN after performing the digits translation routes the call to the home PLMN GMSC. The GMSC contains either the routing tables to relate the MSISDN number with the corresponding HLR, or if the GMSC is connected to the SS7 network with the GTT functionality, the SS7 network will identify the HLR. Once the GMSC interrogates the HLR with the MSISDN number, the HLR determines the IMSI from the MSISDN number. The HLR stores the subscriber's information based on IMSI, not MSISDN. The HLR locates the visiting MSC/VLR point code and if the MSRN is available, it will return the information to GMSC. If the HLR does not have the MSRN for the subscriber it will request one from the visiting MSC/VLR. The latter can be done via GTT if an SS7 backbone with GTT (IMSI to point code) functionality is available/supported. The GMSC once it receives the MSRN and the MSC/VLR point code, will route the call to the VMSC/VLR. The MSC/VLR will then page the mobile subscriber.

The call-originating information including the dialed digits will be sent to the MSC/VLR. The MSC/VLR with the subscriber's profile information performs digits translation (if supported) and routes the call either to the PSTN or to other MSCs. If the MSC cannot perform the digits translation it would route the call to GMSC for translation and routing.

11.2.6 Location Update

A list of relevant functions of a mobile station includes provision of location updates. The location-updating procedures, and subsequent call routing, use the MSC and two location registers: HLR and VLR. When a mobile station is switched on in a new location area, or it moves to a new location area or different operator's PLMN, it must register with the network to indicate its current location. In the normal case, a location update message is sent to the new MSC/VLR, which records the location area information, and then sends the location information to the subscriber's HLR. The information sent to the HLR is normally the SS7 address of the new VLR, although it may be a routing number.

> **Facts to Know!**
>
> The reason a routing number is not normally assigned, even though it would reduce signaling, is that there is only a limited number of routing numbers available in the new MSC/VLR and they are allocated on demand for incoming calls.

If the subscriber is entitled to service, the HLR sends a subset of the subscriber information, needed for call control, to the new MSC/VLR, and sends a message to the old MSC/VLR to cancel the old registration. For reliability reasons, the GSM also has a periodic location updating procedure. If an HLR or MSC/VLR fails, to have each mobile register simultaneously to make the database up-to-date would cause overloading. Therefore, the database is updated as location updating events occur. The enabling of periodic updating, and the time period between periodic updates, is controlled by the operator, and is a trade-off between signaling traffic and speed of recovery. If a mobile does not register after the updating time period, it is de-registered.

A procedure related to location updating is the IMSI attach and detach. A detach permits the network to know that the mobile station is unreachable, and avoids having to needlessly allocate channels and send paging messages. An attach is similar to a location update, and informs the system that the mobile is reachable again. The activation of IMSI attach/detach is up to the operator on an individual cell basis. Location update is a typical example for the connection-oriented transactions in GSM. The local operation code UpdatLocation is required directly after the location update for the new VLR to update the location information in the HLR. Because this is a confirmed service, it requires all four variants: request, indication, response and confirmation.

11.3 IDENTIFIERS USED IN GSM SYSTEM

Several identity numbers are associated with a GSM system, which are briefly described below.

11.3.1 IMSI

The International Mobile Subscriber Identity (IMSI) number is usually 15 digits or less. When an MS attempts a call, it needs to contact a BS. The BS can offer its service only if it identifies the MS as a valid subscriber. For this, the MS needs to store certain values uniquely defined for the MS, like the country of subscription, network type, subscriber ID, and so on. These values are called the International Mobile Subscriber Identity (IMSI). The structure of an IMSI is shown in Fig. 11.5. The first three digits specify the country code, the next two specify the network provider code, and the rest are the mobile subscriber identification code (the customer ID number).

Another use of IMSI is to find information about the subscriber's home Public Land Mobile Network (PLMN). All such information is placed on the SIM card.

11.3.2 Subscriber Identity Module (SIM)

Every time the MS has to communicate with a BS, it must correctly identify itself. An MS does this by storing the mobile phone number, personal identification number for the mobile station, authentication parameters, and so on, in the SIM card. Smart SIM cards also have a flash memory that can be used to store small messages sent to the unit.

The main advantage of SIM is that it supports roaming with or without a cellphone, also called *SIM roaming*. All a user needs to do is to carry the SIM card alone, and insert it into any GSM mobile phone to make it work as per customised MS. In other words, the SIM card is the heart of a GSM mobile phone, and the MS hardware equipment is unusable without it.

11.3.3 Mobile System ISDN (MSISDN)

MSISDN is the number that identifies a particular MS's subscriber, with the format shown in Fig. 11.6.

The GSM actually does not identify a particular mobile phone, but a particular HLR. It is the responsibility of the HLR to contact the mobile phone.

11.3.4 Location Area Identity (LAI)

As shown in Fig. 11.7, the GSM service area is usually divided into a hierarchical structure that facilitates the system to access any MS quickly.

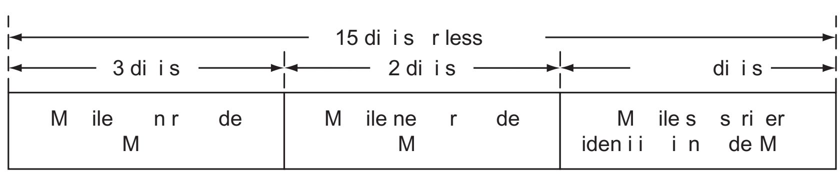

Fig. 11.5 *Format of IMSI*

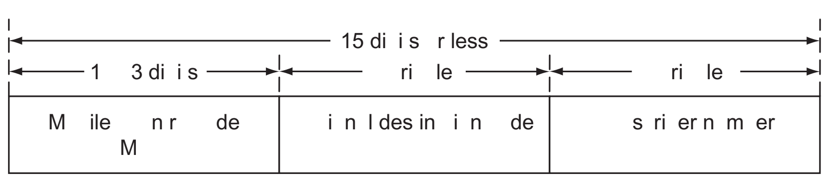

Fig. 11.6 *Format of MSISDN*

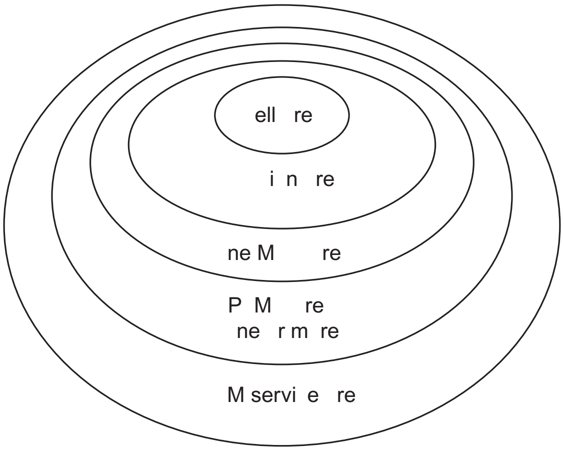

Fig. 11.7 GSM system hierarchy

Each PLMN area is divided into many MSCs. Each MSC typically contains a VLR to inform the system if a particular cellphone is roaming, and if it is roaming, the VLR of the MSC, in which the cellphone is, reflects the fact. Each MSC is divided into many Location Areas (LAs). An LA is a cell or a group of cells and is useful when the MS is roaming in a different cell but the same LA. Since any LA has to be identified as a part of the hierarchical structure, the identifier should contain the country code, the mobile network code, and the LA code.

11.3.5 IMSEI

Each manufactured GSM mobile phone equipment is assigned a 15-bit long International MS Equipment Identity (IMSEI) number to contain manufacturing information, as shown in Fig. 11.8. Conceptually, when the mobile phone equipment passes the interoperability tests, it is assigned a type approval code. Since a single mobile unit may not be manufactured at the same place, a field in IMSEI, called the final assembly code, identifies the final assembly place of the mobile unit. To identify uniquely a unit manufactured, a Serial Number (SN) is assigned. A spare digit is available to allow further assignment depending on requirements.

11.3.6 MS Roaming Number (MSRN)

When an MS roams into another MSC, that unit has to be identified based on the numbering scheme format used in that MSC. Hence, the MS is given a temporary roaming number called the MS Roaming Number (MSRN), with the format shown in Fig. 11.9. This MSRN is stored by the HLR, and any calls coming to that MS are rerouted to the cell where the MS is currently located.

11.3.7 TMSI

As all transmission is sent through the air interface, there is a constant threat to the security of information sent. A temporary identity Temporary Mobile Subscriber Identity (TMSI) is usually sent in place of IMSEI.

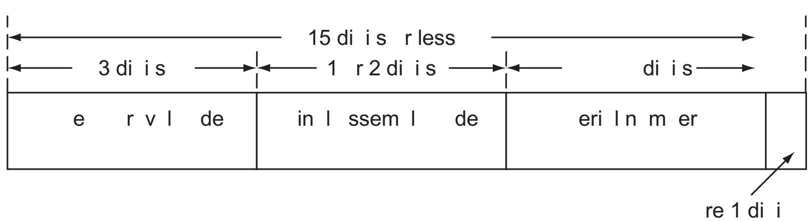

Fig. 11.8 Format of IMSEI

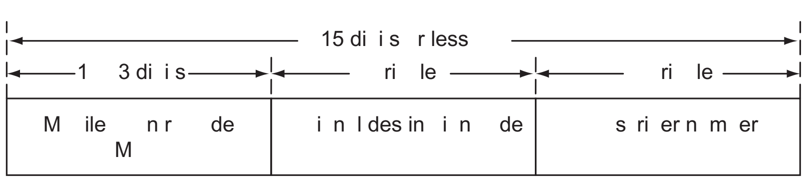

Fig. 11.9 Format of MSRN

11.4 GSM CHANNELS

GSM-900 has been allocated an operational frequency from 890 MHz to 960 MHz. GSM uses the frequency band 890 MHz–915 MHz for uplink (reverse) transmission, and for downlink (forward) transmission, it uses the frequency band 935 MHz–960 MHz. The available 25-MHz spectrum for each direction is divided into 124 Frequency Division Multiplexing (FDM) channels, each occupying 200 kHz with 100 kHz guard band at two edges of the spectrum. This is shown in Fig. 11.10.

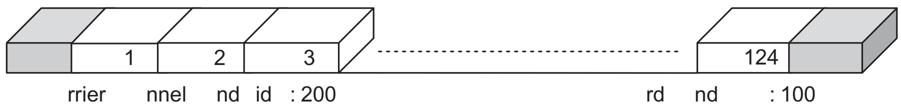

Fig. 11.10 *Frequency channels in GSM-900*

GSM uses FDD and a combination of TDMA and FDMA schemes to provide multiple access to mobile subscribers. The available forward and reverse frequency bands are divided into 200-kHz wide channels called ARFCNs (Absolute Radio Frequency Channel Numbers). The ARFCN denotes a forward and reverse channel pair that is separated in frequency by 45 MHz and each channel is time shared between as many as eight subscribers using TDMA.

The total number of available channels within a 25-MHz bandwidth is 125 (assuming no guard band). Since each radio channel consists of eight time slots, there are thus a total of 1000 traffic channels within a GSM. In practical implementations, a guard band of 100 kHz is provided at the upper and lower end of the GSM spectrum, and only 124 channels are available for use.

Each carrier supports eight time slots for the TDMA operation. Each of the eight subscribers uses the same ARFCN and occupies a unique Time Slot (TS) per frame. A guard frame of 8.25 bits is used in between any two frames transmitted either by the BS or the MS. The data rate of each carrier is 270.833 kbps that is provided with a modulation scheme known as GMSK (Gaussian Minimum Shift Keying) that is a variant of FSK having a frequency deviation of ± 67.71 kHz. The channel data rate of GSM is 270.833 kbps, which is exactly four times the RF frequency shift. This minimises the bandwidth occupied by the modulation spectrum and hence improves channel capacity. The MSK modulated signal is passed through a Gaussian filter to smooth the rapid frequency transitions that would otherwise spread energy into adjacent channels. The bandwidth-time product ($B \times T_b$) of GMSK is standardised at 0.3, a normalised bandwidth expansion factor where 0.3 describes the 3 dB bandwidth of the Gaussian pulse shaping filter with relation to the bit rate, which provides the best compromise between increased bandwidth occupancy and resistance to cochannel interference. Ninety-nine per cent of the RF power of GMSK signals so specified is confined to a bandwidth of 250 kHz, which means that, for all practical purposes, the sidelobes of the GMSK signal are insignificant, and outside this frequency band.

EXAMPLE 11.2 GMSK signal bandwidth in GSM

Show that the 3-dB bandwidth for a Gaussian LPF used to produce $B \times T_b = 0.3$ GMSK modulation in GSM standard is 81.3 kHz. The channel data rate is 270.833 kbps.

Solution

Channel data rate, $R_b = 270.833$ kbps (given)

Baseband symbol duration, $T_b = 1/R_b$

Baseband symbol duration, $T_b = 1/270.833$ kbps $= 3.69$ μs

Product of 3-dB bandwidth and baseband symbol duration, $B \times T_b = 0.3$ (given)

Therefore, 3-dB bandwidth, $B = 0.3/T_b$

Hence, 3-dB bandwidth, $B = 0.3/3.69\ \mu s = 81.3$ kHz
Hence, the 3-dB bandwidth for a Gaussian LPF used to produce $B \times T_b = 0.3$ GMSK modulation in GSM standard is 81.3 kHz.

With the channel data rate of 270.833 kbps, the duration of each bit is 3.69 μs, and the effective channel transmission rate per user is 33.854 kbps (270.833 kpbs/8 users). With overhead, user data is actually transmitted at a maximum rate of 24.7 kbps. The user transmission packet burst is fixed at 577 μs, which accommodates information bits and a time gap between the packets for duration equivalent to 156.25 channel bits times the bit duration of 3.69 μs.

EXAMPLE 11.3 Theoretical S/N in GSM

The channel data rate is 270.833 kbps in GSM standard that is 40% (say) of theoretical maximum data rate that can be supported in a 200-kHz channel bandwidth. Calculate the corresponding theoretical S/N required.

Solution

Step 1. *To calculate theoretical maximum data rate, C*
Channel data rate, $R_b = 270.833$ kbps (given)
Channel data rate, R_b is 40% of theoretical maximum data rate, C (given)
Or, Channel data rate, $R_b = 0.4 \times C$
Theoretical maximum data rate, $C = R_b/0.4$
Hence, theoretical maximum data rate, $C = 270.833$ kbps/0.4 = 677 kbps

Step 2. *To calculate the corresponding theoretical S/N required*
The channel bandwidth, $B = 200$ kHz (given)
The maximum possible theoretical data rate, C is given by Shannon's channel capacity formula

$$C = B \times \log_2 (1 + S/N)$$

Or, $\log_2 (1 + S/N) = C/B$
Or, $\log_2 (1 + S/N) = 677$ kbps/200 kHz
Or, $\log_2 (1 + S/N) = 3.385$
Or, $S/N = 2^{3.385} - 1 = 9.447$
Or, S/N (dB) $= 10 \log_{10} (9.447) = 9.75$ dB
Hence, the corresponding theoretical S required is 9.75 dB

EXAMPLE 11.4 Bandwidth efficiency in GSM

The GSM system uses the GMSK modulation scheme. Show that the bandwidth efficiency of the standard GSM system is 1.35 bps/Hz.

Solution

The channel bandwidth = 200 kHz (standard)
The channel data rate = 270.833 kbps (standard)
Bandwidth efficiency = Channel data rate/Channel bandwidth
Therefore, bandwidth efficiency = 270.833 kbps/200 kHz
Hence, bandwidth efficiency = 270.833 kbps/200 kHz = 1.35 bps/Hz

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GSM employs a moderately complicated, 13-kbps regular pulse-excited speech codec (coder/decoder) with a long-term predictor. To provide error protection for the speech-encoded bits, concatenated convolutional codes and multilayer interleaving are employed. An overall speech delay of 57.5 ms occurs in the system.

11.4.1 GSM Logical Channels

The combination of a time slot number and an ARFCN constitutes a physical channel for both the forward and reverse links. Each physical channel in a GSM system can be mapped into different logical channels at different times. That is, each specific time slot or frame may be dedicated to either handling traffic data (user data such as speech, facsimile, or teletext data), signaling data, or control channel data (from the MSC, base station, or mobile subscriber).

Communication between the mobile subscriber and the base station is involved with both voice as well as signaling and control. Voice and signaling packets are inserted in a hierarchy and mobile subscribers use counters to identify the location of specific packet bursts in the overall structure of the frames. The entire communication system can be thought of as a distributed real-time computer that uses a number of instructions to transfer information packets from one location to another. Initial signaling is needed for registration and call establishment, followed by maintaining the synchronisation among the mobile subscribers, mobility management, and need to transfer the data traffic.

There is a need of a set of instructions and ports to instruct different elements of the cellular network to perform their specified duties. In cellular communication systems, these ports are referred to as logical channels. Logical channels use a physical TDMA slot or a portion of a physical slot to specify an operation in the network in GSM. GSM uses a variety of multiplexing techniques to create a collection of logical channels.

The GSM specification defines a wide variety of logical channels that can be used to link the physical layer with the data link layer of the GSM network. These logical channels efficiently transmit user data while simultaneously providing control of the network on each ARFCN. GSM provides explicit assignments of time slots and frames for specific logical channels. The logical channels used by a GSM system are shown in Table 11.2.

Table 11.2 *Logical channels in GSM*

Channel type	*Channel group*	*Channel*	*Direction*
Control Channel (CCH)	Broadcast Channel (BCH)	Broadcast Control Channel (BCCH)	Downlink (BS → MS)
		Frequency Correction Channel (FCCH)	Downlink (BS → MS)
		Synchronisation Channel (SCH)	Downlink (BS → MS)
	Common control Channel (CCCH)	Paging Channel (PCH)	Downlink (BS → MS)
		Random Access Channel (RACH)	Uplink (MS → BS)
		Access Grant Channel (AGCH)	Downlink (BS → MS)
	Dedicated control Channel (DCCH)	Standalone Dedicated Control Channel (SDCCH)	Uplink and Downlink (BS ↔ MS)
		Slow Associated Control Channel (SACCH)	Uplink and Downlink (BS ↔ MS)
		Fast Associated Control Channel (FACCH)	Uplink and Downlink (BS ↔ MS)
Traffic Channel (TCH)	Traffic Channel (TCH)	Full-rate Traffic Channel (TCH/F)	Uplink and Downlink (BS ↔ MS)
		Half-rate Traffic Channel (TCH/H)	Uplink and Downlink (BS ↔ MS)

The logical channels in the GSM network are divided into two principal categories: Control Channels (CCHs) and Traffic Channels (TCHs). Control channels carry signaling and synchronising commands between the base station and the mobile station. Certain types of control channels are defined for just the forward or reverse link. Traffic channels carry digitally encoded user speech or user data and have identical functions and formats on both the forward and reverse link. GSM system uses a variety of logical control channels to ensure uninterrupted communication between MSs and the BS.

11.4.2 GSM Control Channels

There are three classes of control channels defined in GSM: Broadcast Channels (BCH), Common Control Channels (CCCH), and Dedicated Control Channels (DCCH). Each control channel consists of several logical channels that are distributed in time to provide the necessary GSM control functions.

Facts to Know!

The BCH and CCCH forward control channels in GSM are implemented only on certain ARFCN channels and are allocated time slots in a very specific manner.

Specifically, the BCH and CCCH forward control channels are allocated only TS 0 and are broadcast only during certain frames within a repetitive fifty-one frame sequence (called the control channel multiframe) on those ARFCNs which are designated as broadcast channels. TS1 through TS7 carry regular TCH traffic, so that ARFCNs that are designated as control channels are still able to carry full-rate users on seven of the eight time slots.

The GSM specification defines thirty-four ARFCNs as standard broadcast channels. For each broadcast channel, the frame number 51 does not contain any BCH/CCCH forward channel data and is considered to be an idle frame. However, the reverse channel CCCH is able to receive subscriber transmissions during TS 0 of any frame (even the idle frame). On the other hand, DCCH data may be sent during any time slot and any frame, and entire frames are specifically dedicated to certain DCCH transmissions.

The BCH channels are broadcast from the BTS to MSs in the coverage area of the BTS, and thus are one-way channels. The broadcast channel operates on the forward link of a specific ARFCN within each cell, and transmits data only in the first time slot (TS 0) of certain GSM frames. The BCH provides synchronisation for all mobiles within the cell and is occasionally monitored by mobiles in neighbouring cells so that received power and MAHO decisions may be made by out-of-cell users. Although BCH data is transmitted in TS 0, the other seven time slots in a GSM frame for that same ARFCN are available for TCH data, DCCH data, or are filled with dummy bursts. Furthermore, all eight time slots on all other ARFCNs within the cell are available for TCH or DCCH data. There are three separate broadcast channels that are given access to TS 0 during various frames of the 51-frame sequence.

(a) The Broadcast Control Channel (BCCH) is used by BTS to broadcast system parameters such as the frequency of operation in the cell, operator identifiers, cell ID, and available services to all the MSs. Once the carrier, bit, and frame synchronisation between the BTS and MS are established, the BCCH informs the MS about the environment parameters associated with the BTS covering that area such as current control channel structure, channel availability, and congestion. The BCCH also broadcasts a list of channels that are currently in use within the cell. Frames 2 through frame 5 in a control multiframe (4 out of every 51 frames) contain BCCH data. The BCCH is physically implemented over the Normal Burst (NB). The BCCH is also a continuously keyed channel, and it is used for signal strength measurements for hand-off.

(b) The Frequency Correction Control Channel (FCCH) is used by the BTS to broadcast frequency references and frequency correction burst of 148 bits length. An MS in the coverage area of a BTS uses the broadcast FCCH signal to synchronise its carrier frequency and bit timing. The FCCH is a special data burst that occupies TS 0 for the very first GSM frame (Frame 0) and is repeated every ten frames within a control channel multiframe. The physical Frequency Correction Burst (FCB) is used to implement the logical FCCH.

(c) The Synchronisation Channel (SCH) is used by the BTS to broadcast frame synchronisation signals containing the synchronisation training sequences burst of 64 bits length to all MSs. Using SCH, MSs will synchronise their counters to specify the location of arriving packets in the TDMA hierarchy. SCH is broadcast in TS 0 of the frame immediately following the FCCH frame and is used to identify the serving base station while allowing each mobile to frame-synchronise with the base station. The frame number, which ranges from 0 to 2,715,647, is sent with the Base Station Identity Code (BSIC) during the SCH burst. The BSIC is uniquely assigned to each BTS in a GSM system. Since a mobile may be as far as 30 km away from a serving base station, it is often necessary to adjust the timing of a particular mobile user such that the received signal at the base station is synchronised with the base station clock. The BSC issues coarse timing advancement commands to the mobile stations over the SCH, as well. The SCH is transmitted once every ten frames within the control channel multiframe. The physical Synchronisation Burst (SB) is used to implement SCH.

The Common Control Channels (CCCH) are also one-way channels used for establishing links between the MS and the BS, as well as for any ongoing call management. CCCHs are the most commonly used control channels and are used to page specific subscribers, assign signaling channels to specific users, and receive mobile requests for service. On the broadcast channel ARFCN, the common control channels occupy TS 0 of every GSM frame that is not otherwise used by the BCH or the Idle frame. There are three CCCH logical channels:

(a) The Paging Channel (PCH) is a forward link channel and is used by the BTS to page or notify a specific individual MS for an incoming call in the cell. The PCH transmits the IMSI of the target subscriber, along with a request for acknowledgment from the mobile unit on the RACH. Alternatively, the PCH may be used to provide cell broadcast ASCII text messages to all subscribers, as part of the SMS feature of GSM. The PCH is implemented on a Normal Burst (NB).

(b) The Random Access Channel (RACH) is a reverse link channel and is used by the MS either to access the BTS requesting the dedicated channel for call establishment or to acknowledge a page from the PCH. The RACH is used with implementation of a slotted-ALOHA protocol, which is used by MSs to contend for one of the available slots in the GSM traffic frames. The RACH is implemented on the short Random Access Burst (RAB). All mobiles must request access or respond to a PCH alert within TS 0 of a GSM frame. At the BTS, every frame (even the idle frame) will accept RACH transmissions from mobiles during TS 0. In establishing service, the GSM base station must respond to the RACH transmission by allocating a channel and assigning a Standalone Dedicated Control Channel (SDCCH) for signaling during a call. This connection is confirmed by the base station over the AGCH.

(c) The Access Grant Channel (AGCH) is used by the base station to provide forward link communication to the mobile for implementation of the acknowledgement from the BTS to the MS after a successful attempt by MS using RACH in a previous CCCH frame. AGCH is also used by the BS to send information about timing and synchronisation to the MS. The AGCH carries data that instructs the mobile to operate in a particular physical channel (time slot and ARFCN) with a particular dedicated control channel. The AGCH is the final CCCH message sent by the base station before a subscriber is moved off the control channel. This channel is implemented on a Normal Burst (NB) and indicates the TCH for access to the GSM network.

The Dedicated Control Channels (DCCH) are two-way channels having the same format and function on both the forward and reverse links, supporting signaling and control for individual mobile subscribers, and are used along with voice channels to serve for any control information transmission during actual voice communication. DCCHs may exist in any time slot and on any ARFCN except TS 0 of the BCH ARFCN. There are three DCCH logical channels:

(a) The Stand-alone Dedicated Control Channel (SDCCH) is a two-way channel allocated with SACCH to each mobile terminal to transfer network control and signaling information for call establishment and mobility management, just before a TCH assignment is issued by the base station. The SDCCH ensures that the mobile station and the base station remain connected while the base station and MSC verify the subscriber unit and allocate resources for the mobile. The SDCCH is used to send authentication and alert messages (but not speech) as the mobile synchronises itself with the frame structure and waits for a TCH. SDCCHs may be assigned their own physical channel or may occupy TS 0 of the BCH if there is low demand for BCH or CCCH traffic. The physical channel for SDCCH occupies four slots in every 51 control-multiframes with an approximated gross data rate of 2 kbps per mobile terminal.

(b) The Slow Associated Control Channel (SACCH) is a two-way channel always associated with a TCH or a SDCCH and maps onto the same physical channel. The SACCH is used to exchange the necessary parameters between the BTS and the MS during the actual transmission to maintain the communication link. Each ARFCN systematically carries SACCH data for all of its current users. The gross data rate of the SACCH channel is half of that of the SDCCH. On the forward link, the SACCH is used to send slow but regularly changing control information to the mobile subscriber, such as transmit power level instructions and specific timing advance instructions for each user on the ARFCN. The reverse SACCH carries information about the received signal strength and quality of the TCH, as well as BCH measurement results from neighbouring cells. The SACCH is transmitted during the thirteenth frame (and the twenty-sixth frame when half-rate traffic is used) of every speech/dedicated control channel multiframe, and within this frame, the eight time slots are dedicated to providing SACCH data to each of the eight full-rate (or sixteen half-rate) users on the ARFCN.

(c) The Fast Associated Control Channel (FACCH) is a two-way channel used to support fast transitions such as a hand-off request in the channel when SACCH is not adequate. The FACCH is physically multiplexed with the TCH or SDCCH to provide additional support to the SACCH. In fact, FACCH is not a dedicated control channel but carries the same information as SDCCH. However, FACCH is a part of the traffic channel, while SDCCH is a part of the control channel. To facilitate FACCH to steal certain bursts from the TCH forward channel burst, there are 2 flag bits, called the *stealing bits* in the message. If the stealing bits are set, the time slot is known to contain FACCH data, not a TCH, for that frame.

Control information in GSM is mainly on two logical channels—the Broadcast Channel (BCCH) and the Paging Channel (PCH). As with TDMA, it is unnecessary to use a whole RF channel for this. Instead, one of the eight time slots on one RF channel in each cell or sector is designated as a control channel. The broadcast information is transmitted first, followed by paging information. Figure 11.11 shows the structure of a GSM logical control channel.

The corresponding reverse channel to the forward channels BCCH and PCH is called the Random-Access Channel (RACH). It is used by the mobile subscribers to communicate with the cell-site. Mobile subscribers transmit on RACH channel whenever they have information. Transmissions are shorter than the duration of the time slot to prevent interference caused by the propagation delay between the mobile and cell-site.

The mobile subscriber equipment has only one receiver. It cannot count on receiving the broadcast channel during a call because both channels may use the same time slot. It is necessary to send control information

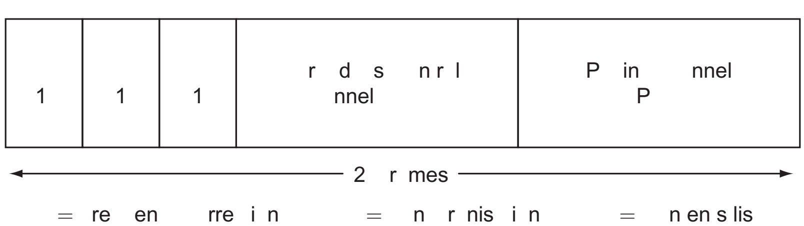

Fig. 11.11 *GSM control channel structure*

on the voice channels. There are two control channels associated with the voice channel. The slow associated control channel uses one of every 26 bursts on the voice channel. It is used to inform the serving cell-site about the measurements of the signal strength of adjacent cells made by the mobile subscriber. The fast associated control channel steals bits from the voice signal and is used for urgent messages from the cell-site such as hand-off instructions.

11.4.3 GSM Traffic Channels

Voice channels are called Traffic Channels (TCH) in GSM. Traffic channels are two-way channels carrying the voice and data traffic between the MS and BTS. In the GSM standard, TCH data may not be sent in TS 0 within a TDMA frame on certain ARFCNs that serve as the broadcast station for each cell (since TS 0 is reserved for control channel bursts in every frame). Traffic channels carry digitally encoded user speech or user data and have identical functions and formats on both the forward and reverse link. One RF channel is shared by eight voice transmissions using TDMA. In terms of spectral efficiency, GSM works out to 25 kHz per voice channel, compared to about 30 kHz for AMPS and about 10 kHz for TDMA-based IS-54 or IS-136 systems. This is an approximate comparison as it ignores differences in control-channel overhead. As in TDMA-based systems, the mobile transmitter operates only during its allotted time slot. Assuming other parameters similar, a GSM mobile phone has longer battery life than a phone using either AMPS or IS-54/IS-136 because GSM mobile transmits in one-eighth of the time, compared with one-third of the time in TDMA-based IS-54/IS-136 system.

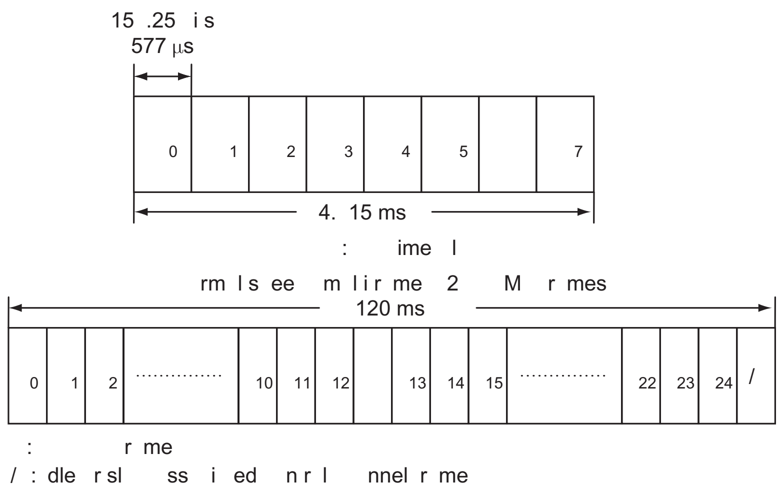

Fig. 11.12 *The TCH frame and multiframe structure*

Figure 11.12 illustrates how the TCH data is transmitted in consecutive frames.

Furthermore, frames of TCH data are broken up every thirteenth frame by either Slow Associated Control Channel Data (SACCH) or idle frames. Each group of twenty-six consecutive TDMA frames is called a multiframe (or speech multiframe, to distinguish it from the control channel multiframe described later). For every twenty-six frames, the thirteenth and twenty-sixth frames consist of Slow Associated Control Channel (SACCH) data, or the idle frame, respectively. The twenty-sixth frame contains idle bits for the case when full-rate TCHs are used, and contains SACCH data when half-rate TCHs are used. TCH logical channels are implemented over the normal burst. There are two types of TCH channels:

The full-rate traffic channel (TCH/F) uses a 13 kbps speech-coding scheme and 9,600 bps, 4,800 bps, and 2,400 bps data. After including signaling overhead, each full-rate traffic channel has a gross bit rate of 22.8 kbps for the network. When transmitted as full-rate, user data is contained within one time slot per frame. The following full-rate speech and data channels are supported:

Full-Rate Traffic Data Channel (TCH/F) The full-rate speech channel carries user speech that is digitised at a raw data rate of 13 kbps. With GSM channel coding added to the digitised speech, the full-rate speech channel carries 22.8 kbps.

Full-Rate Data Channel at 9600 bps (TCH/F9.6) The full-rate traffic data channel carries raw user data that is sent at 9600 bps. With additional forward-error-correction coding applied by the GSM standard, the 9600 bps data is sent at 22.8 kbps.

Full-Rate Data Channel at 4800 bps (TCH/F4.8) The full-rate traffic data channel carries raw user data that is sent at 4800 bps. With additional forward-error-correction coding applied by the GSM standard, the 4800 bps is sent at 22.8 kbps.

Full-Rate Data Channel at 2400 bps (TCH/F2.4) The full-rate traffic data channel carries raw user data that is sent at 2400 bps. With additional forward error correction coding applied by the GSM standard, the 2400 bps is sent at 22.8 kbps.

The half-rate traffic channel (TCH/H) uses 16 time slots per frame that has a gross bit rate of 11.4 kbps (half of gross bit rate of full-rate traffic channel). The half-rate TCH supports 4800 bps and 2400 bps data rate only. When transmitted as half-rate, user data is mapped onto the same time slot, but is sent in alternate frames. That is, two half-rate channel users would share the same time slot, but would alternately transmit during every other frame. The following half-rate speech and data channels are supported:

Half-Rate Traffic Data Channel (TCH/H) The half-rate speech channel has been designed to carry digitised speech which is sampled at a rate half that of the full-rate channel. GSM anticipates the availability of speech coders that can digitise speech at about 6.5 kbps. With GSM channel coding added to the digitised speech, the half-rate speech channel will carry 11.4 kbps.

Half-Rate Traffic Data Channel at 4800 bps (TCH/H4.8) The half-rate traffic data channel carries raw user data that is sent at 4800 bps. With additional forward-error-correction coding applied by the GSM standard, the 4800 bps data is sent at 11.4 kbps.

Half-Rate Traffic Data Channel at 2400 bps (TCH/H2.4) The half-rate traffic data channel carries raw user data that is sent at 2400 bps. With additional forward-error-correction coding applied by the GSM standard, the 2400 bps data is sent at 11.4 kbps.

The cell-site instructs the mobile subscriber to advance or retard the timing of its transmissions to compensate for the changes in propagation delay as it moves about in the cell. In this way, the transmission delay problem is avoided on the traffic channels.

11.5 FRAME STRUCTURE FOR GSM

Transmission in any TDMA-based wireless communication system is in the form of a repetitive sequence of frames. Each TDMA frame is divided into a number of uniform time slots. Each time slot position across the sequence of frames forms a separate logical channel. It is very critical to determine the length and composition of the logical channel time slot that will provide effective speech and data transmission with efficient use of the available frequency spectrum. In designing an appropriate frame structure in TDMA, the following requirements are generally considered:

Frequency Band of Operation The most common spectrum allocated to cellular mobile communication applications is around 900 MHz.

Number of Logical Channels or Number of Time Slots in TDMA Frame In order to justify the additional costs of multiplexing, let the minimum number of time slots per TDMA frame be 8 so as to serve eight simultaneous users.

Channel Bandwidth The current channel bandwidth being used for analog FM cellular systems in Europe is 25 kHz. To serve 8 mobile subscribers using TDMA technique, the channel bandwidth should not exceed 200 kHz.

Maximum Cell Radius (R) To provide radio service to high traffic in rural areas, let the maximum cell radius be 35 km.

Maximum Vehicle Speed (V_m) To accommodate mobile subscriber units traveling on expressways or high-speed trains, the maximum vehicle speed be 250 km/h.

Maximum Delay Spread (Δ_m) Delay spread is the difference in propagation delay among different multipath signals arriving at the same Rx antenna. Typical delay spread in mountainous regions is about 10 seconds.

Maximum Coding Delay To avoid unnecessary delays within the fixed wireless network, which may involve satellite links, maximum coding delay be approximately 20 milliseconds.

Figure 11.13 suggests the general steps to be considered in designing the time slot in a TDMA frame.

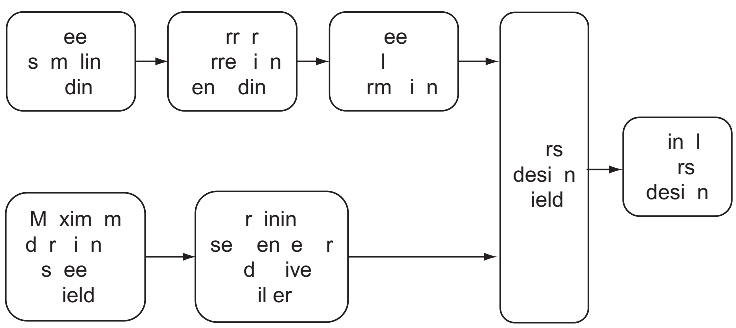

Fig. 11.13 *Steps in design of TDMA time slot*

In the design of time slot of a TDMA frame, an appropriate data rate of the speech coder should be decided first. It is desirable that the speech coder must provide satisfactory speech quality at minimum data rate. The PCM speech coder has a data rate of 64 kbps, which is undesirably high for use with wireless systems. A data rate of 12 kbps is reasonable for reproducing good-quality speech. Since the coding delay is restricted to 20 milliseconds, the encoded speech can be formed into blocks of 20 ms duration. This converts the speech samples of 12 kbps $\times$ 20 ms $=$ 240 bits.

Error correction can then be applied to the 240-bit blocks. Using a convolutional error-correcting technique with a code rate of , the number of bits in a block of 20-ms speech at 12 kbps rate increases to (2 $\times$ 240 bits $=$) 480-bits. With a constraint length of 5, 4 bits per block of 240 bits are added to account for the length of the shift register. This brings the speech block length to (480 $+$ 2 $\times$ 8 $=$) 488 bits. With these parameters, the minimum bit rate for an eight-channel TDMA system can be computed as follows:

Number of bits in one channel $=$ 488 bits

Number of channels or time sots $=$ 8

Total number of bits in 8 time slots $=$ 488 bits $\times$ 8 $=$ 3904 bits

Duration of one speech block $=$ 20 ms

Overall minimum channel bit rate $=$3904 bits/20 ms $=$ 195.2 kbps

To take care of other design considerations, the gross channel bit rate will be slightly higher, and let it be greater than 200 kbps in the available channel bandwidth of 200 kHz. In a mobile radio environment, such data rates can be achieved with the use of adaptive equalisation. Adaptive equalisation will require the inclusion of a new training sequence in each time slot when the mobile subscriber moves a sufficient distance to potentially cause changes in the characteristics of transmission path. Assume that the phase angle of the carrier signal changes by λ_c /20 of the maximum vehicle speed. Thus, at 900 MHz (λ_c = 0.333m), we have

Maximum transmission duration (one-way) = $(\lambda_c/20)/V_m$

Maximum transmission duration (one-way) = (0.333m/20)/250 km/h

Maximum transmission duration (one-way) = 0.24 ms

Or, Maximum transmission duration (two-way) = 2 × 0.24 = 0.48 ms

Hence, Duration for data transmission in a time slot, $_d$ = 0.48 ms

Assuming the number of taps on the adaptive equaliser to be equal to 6 times the number of bits transmitted in the maximum dispersal time (Δ_m = 0.01 ms), the amount of time needed for the training sequence in the time slot, $_t$ = 0.06 ms.

To account for the differing amounts of delay between different mobile units and the base station, a guard interval is needed at the end of each time slot. Because eight mobile subscriber units share the same TDMA frame, it is necessary to adjust the timing of the transmissions of the mobile subscriber units so that the transmission from one mobile subscriber does not interfere with adjacent time slots. The guard time can be computed as follows.

Let the average duration of the voice call = 120 seconds

Maximum vehicle speed of the mobile = 250 km/h

Therefore, the radial distance a mobile moving toward or away from the base station located at the centre of the cell = (250 km/h) × (120 s) = 8333 m

The change in propagation delay = 8333 m/(3 × 10^8 m/s) ≈ 0.03 ms

Or, Required duration for guard interval, $_g$ = 0.03 ms

Figure 11.14 shows the tentative design of a time slot, depicting time duration of two blocks of data before and after the training sequence and guard time.

2 0.24 s	0.0 ms	2 0.24 s	0.03 ms

Fig. 11.14 *TDMA time slot—approximate field durations*

So, Maximum time duration of a time slot, $T_s = T_d + T_{ts} + T_g$

Maximum time duration of a time slot, T_s = 0.48 ms + 0.06 ms + 0.03 ms

Maximum time duration of a time slot, T_s = 0.57 ms

Number of time sots in a TDMA frame = 8

Duration of a TDMA frame = 8 × 0.57 ms = 4.6 ms

This is quite close to actual design of a TDMA time slot and frame structure used in GSM. Fundamentally, each 200-kHz frequency band is divided into 8 logical channels defined by the repetitive occurrence of time slots. The GSM system uses the TDMA scheme shown in Fig. 11.15 with a 4.615 ms-long frame, divided into eight time slots each of 557 μs. Each frame is 156.25 bits long, of which 8.25 bits are guard bits.

At the lowest level is the time slot or a burst period, which has a duration of approximately 577 μs. With a bit rate of 270.833 kbps, each time slot has a length of 156.25 bits. The time slot includes the following fields:

Tail Bits, T (3 Bits each at the Beginning and End of a Time Slot Excluding Guard Bits) It allows synchronisation of transmissions from mobile units located at different distances from the base station.

Encrypted Data (114 Bits) Data is encrypted in blocks by conventional encryption of 114 plaintext bits into 114 ciphertext bits; the encrypted bits are then placed in two 57-bit data fields in the time slot.

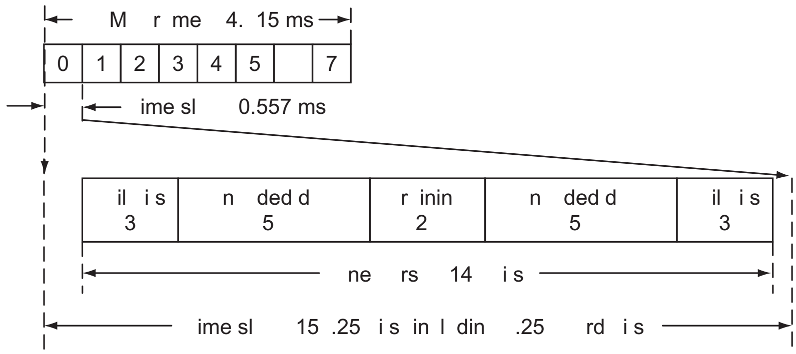

Fig. 11.15 *GSM basic frame structure*

Stealing Bit, S (1 Bit each at the End of Two 57-Bit Data Fields in the Time Slot) It is used to indicate whether this block contains data or is stolen for urgent control signaling purpose during the call.

Training Data (26 Bits) It is used to adapt the parameters of the receiver to the current path-propagation characteristics and to select the strongest signal in case of multipath propagation. The training sequence is a known bit pattern that differs for different adjacent cells. It enables the mobile subscriber units and base stations to determine that the received signal is from the desired base station and not from a strong interfering base station. In addition, the training sequence is used for multipath equalisation, which is used to extract the desired signal from unwanted reflections.

Guard Bits, G (8.25 Bits) It is used to avoid overlapping with other bursts due to different path delays.

The 148 bits of a data burst are used to transmit the information. Delimited by tail bits (consisting of 0s), the frame contains 26 training bits sandwiched between two bursts of data bits. These training bits allow the receiver to synchronise itself.

Moving up the frame format hierarchy, 8-time slots TDMA frames are typically organised into a 26-frame multiframe. One of the frames in the multiframe is used for control/signaling and another is currently unused, leaving 24 frames for data traffic. Thus, each traffic channel receives one slot per frame and 24 frames per 120-ms multiframe.

The gross channel data rate can be calculated as follows:

$$\text{Number of data bits per time slot} = 114 \text{ bits}$$

$$\text{Number of time slots per multiframe} = 24$$

So, $$\text{Number of bits per multiframe} = 24 \times 114 \text{ bits} = 2736 \text{ bits}$$

$$\text{Time duration of one multiframe} = 120 \text{ ms}$$

So, $$\text{Gross data rate} = 2736 \text{ bits}/120 \text{ ms} = 22.8 \text{ kbps}$$

GSM uses a complex hierarchy of TDMA frames to define logical channels, as shown in Fig. 11.16.

The GSM specification also allows half-rate traffic channels, with two traffic channels each occupying one time slot in 12 of the 26 frames. With the use of half-rate speech coders, this effectively doubles the capacity of the system. There is also a 51-frame multiframe used for control traffic. Thus, many frames are combined to constitute multiframe, superframe, and hyperframes.

EXAMPLE 11.5 Frame duration in GSM

GSM uses a frame structure where each frame consists of 8 time slots, and each time slot contains 156.25 bits and data is transmitted over a channel at 270.833 kbps. Find time duration of a bit; time duration of a time slot; time duration of a TDMA frame; and how long must a user wait when occupying a single time slot between two successive transmissions.

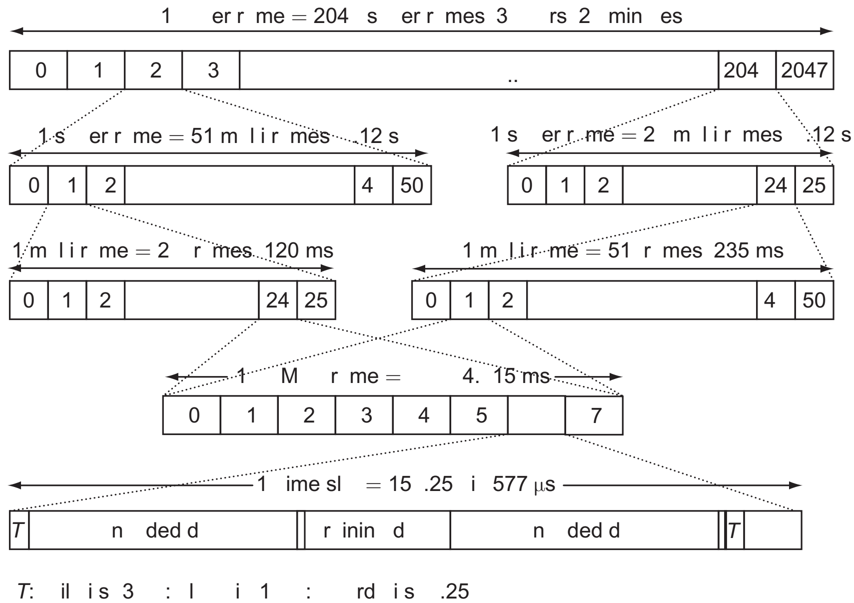

Fig. 11.16 *GSM hyperframe format*

Solution

Step 1. *To find time duration of a bit,* T_b

Channel data rate = 270.833 kbps (given)

Time duration of a bit, T_b = 1/Data rate

Hence, time duration of a bit, T_b = 1/270.833 kbps = 3.69 s

Step 2. *To find time duration of a time slot,* T_{slot}

No. of bits per time slot = 156.25 bits (given)

Time duration of a time slot, $T_{slot} = 156.25 \text{ bits} \times T_b$

Time duration of a time slot, $T_{slot} = 156.25 \text{ bits} \times 3.69\ \mu s = 577$ s

Step 3. *To find time duration of a TDMA frame,* T_f

No. of time slots per TDMA frame = 8 (given)

Time duration of a frame, T_f = No. of time slots × T_{slot}

Time duration of a frame, $T_f = 8 \times 577\ \mu s = 4.616$ ms

Step 4. *To find time duration for a user occupying a single time slot between two successive transmissions*

A user occupying a single time slot between two successive transmissions has to wait for the time duration of a frame. Hence, a user has to wait for 4.616 ms between two successive transmissions.

11.5.1 Physical Data Bursts in GSM

Each user transmits a burst of data during the time slot assigned to it. GSM supports five types of packet data bursts used for control and traffic signaling.

The normal burst, shown in Fig. 11.17, is used for TCH and DCCH transmissions on both the forward and reverse link.

The normal burst consists of three bits each at the beginning and at the end of the data burst, 8.25 bits of guard period, two sets of 58 bits encrypted bits (a total of 116 bits), and a 26-bits training sequence.

r is 3	n r ed d 5	r inin is 2	n r ed d 5	is 3	rd eri d .25 is

Fig. 11.17 *The normal data burst in GSM*

The start bits are 000 providing a gap time for the digital radio circuitry to cover the uncertainty period to ramp on and off for the radiated power and to initiate the convolutional decoding of the data. The 26-bit training sequence is used to train the adaptive equaliser at the receiver. The training of the equaliser is in the middle of the burst because the channel behaviour is constantly changing during the transmission of the data burst. The 116 encrypted data bits include 114 bits of data and two flag bits at the end of each part of the data that indicates whether data is user traffic or signaling and control information during the call.

The Frequency-Correction (FCCH) burst is used in TS 0 of specific frames to broadcast the frequency synchronisation control messages by the BTS on the forward link. MSs use it to synchronise with the master clock in the system. The frame format of the FCCH data burst is shown in Fig. 11.18.

r is 3	ixed is ll 0s 142	is 3	rd eri d .25 is

Fig. 11.18 *The FCCH data burst in GSM*

The FCCH burst has three bits at the start and the end of the data field. The rest of the data burst contains all 0s that allows transmission of the unmodulated carrier frequency. Guard period equivalent to 8.25-bits duration is used between two bursts.

The synchronisation (SCH) burst, as shown in Fig. 11.19, is very similar to the normal burst except that the training sequence is longer and the coded data are used for the specific task of identifying the network. The SCH burst is used in TS 0 of specific frames to broadcast the frequency and time synchronisation control messages on the forward link.

r is 3	n r ed d 3	r inin is 4	n r ed d 3	is 3	rd eri d .25 is

Fig. 11.19 *The SCH data burst in GSM*

The BTS broadcasts the SCH burst, and the MSs use it for initial training of the equaliser, initial learning of the network identity and to synchronise the time slots.

The random access (RACH) burst, as shown in Fig. 11.20, is used by the MS to access the BS as it registers to the network.

r is	n r nis i n 41	n r ed d 3	is 3	x ended rd eri d .25

Fig. 11.20 *The RACH data burst in GSM*

The overall structure of the RACH data burst is similar to the normal data burst except that a longer start bits and synchronisation sequence is used to initiate the equaliser. A much longer guard period of 68.25 bits allows approximate estimation of the distance of the MS from its serving BTS. The distance can be computed from the arrival time of the RACH burst. A guard period of 68.25 bits translates to 252 μs. The signal transmitted from a MS should travel more than 75.5 km (at the signal speed of 300,000 km/sec) before arriving at the BTS to exceed this guard period.

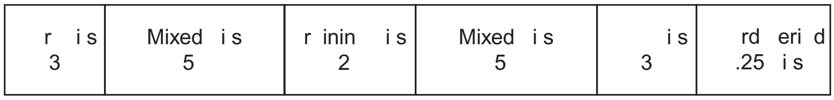

Fig. 11.21 The dummy burst in GSM

The dummy burst, as shown in Fig. 11.21, is used as filler information for unused time slots on the forward link.

EXAMPLE 11.6 GSM frame hierarchy

Explain the concept of GSM superframe, multiframe, TDMA frame and time slot in a GSM channel. Give suitable illustration for GSM frame hierarchy.

Solution

When a number of different time slots carry various types of control signals and user data traffic, a hierarchy is needed to identify the location of certain type of bursts among the large stream of bursts that are directed toward different mobile subscribers. Each mobile subscriber needs a number of counters to track the related frames at different levels of the frame hierarchy. Figure 11.22 illustrates the frame structure.

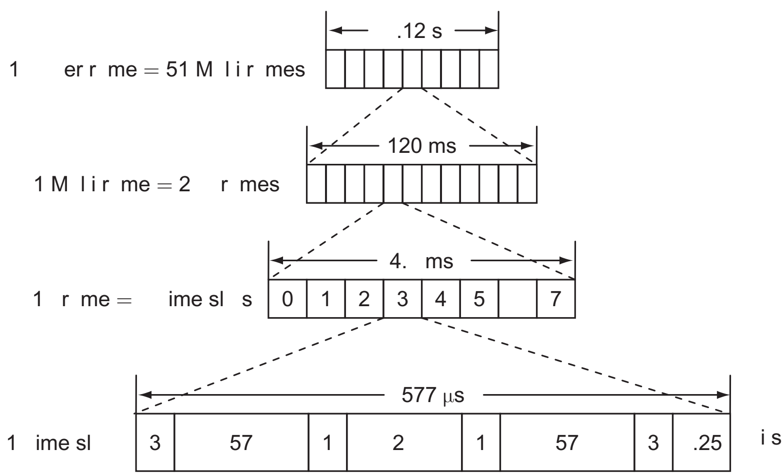

Fig. 11.22 GSM superframe structure

The Concept of Time Slot in GSM Channel A time slot consists of 156.25 bits that are transmitted at a rate of 270.833 kbps. In one time slot, 114 bits are encrypted data bits transmitted as two times 57 data bits each. The training sequence in the middle of the time slot consists of 26 bits. It allows the adaptive equaliser in the receivers of the base station and mobile unit to analyse the characteristics of the wireless channel before decoding the user data. On either side of the training sequence, there are control bits called stealing flags. The bit value of these two flags distinguishes the time slot to contain either the voice or control information during the call.

The Concept of TDMA Frames in GSM Channel During a frame, one time slot is used to transmit only and another time slot is used to receive only. The remaining six time slots of a frame can be used to measure received signal level from its serving base station as well as that from up to five adjacent base stations.

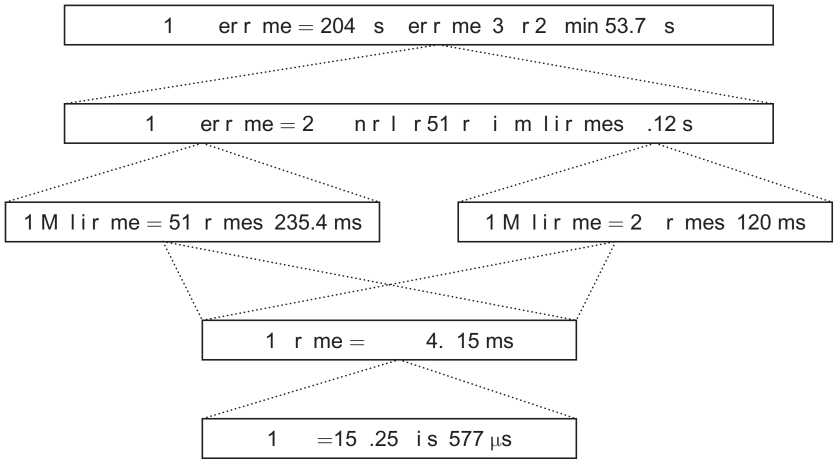

Fig. 11.23 *GSM frame hierarchy*

The GSM radio-interface standard provides a variety of control channels and traffic channels defined in a hierarchy built upon the basic eight-slot TDMA transmission format. The frame hierarchy, depicted in Fig. 11.23, shows the TDMA hierarchy of the GSM network from a normal data burst of 577 μs interval to a hyperframe of length of around three-and-half hours.

The basic building block of the GSM frame hierarchy is a 4.615-ms TDMA frame. Each frame comprises of eight data bursts or time slots. The time-slot interval is equivalent to the transmission time for about 156.25 bits, comprising of 114 bits user data and the remaining overhead bits. A TDMA frame contains 8×156.25 bits = 1250 bits. The frame rate is 270.833 kbps/1250 bits/frame = 216.66 frames per second.

The Concept of Multiframes in GSM Channel Each of the normal speech frames are grouped into larger structures called multiframes, superframes and hyperframes. The 13th or 26th frames are used for control data only. Each 120-ms multiframe is composed of 26 frames—24 frames carry user information, and two frames carry system control information related to individual users. The gross data rate per user is 24×114 bits / 120 ms = 22.8 kbps. The speech coder has a net data rate of 13 kbps, and the addition of error-correction coding results into gross transmission data rate up to 22.8 kbps per user. The eight-slot TDMA frames may be also organised into control multiframes. Control multiframes are used to establish several types of signaling and control channels used for system access, call set-up, synchronisation, and other system control functions. The control multiframes span 51 TDMA frames.

The Concept of Superframes in GSM Channel Either traffic or control multiframes are grouped into superframes, which are in turn grouped into hyperframes. One traffic multiframe contains 26 frames, and one traffic superframe contains 51 traffic multiframes, or 1326 frames. A hyperframe contains 2048 superframes, or 2,715,648 frames. A complete hyperframe takes about every 3 hours, 28 minutes, and 54 seconds. The encryption algorithms rely on the particular frame number, and sufficient security can only be obtained by using a large number of frames as provided by the hyperframe. Counters at the mobile subscribers need to track the frame numbers at hyperframe, superframe, and multiframe levels to communicate with the network.

11.6 GSM SPEECH CODING

The speech signal is compressed using an algorithm known as Regular Pulse Excited-Linear Predictive Coder (RPE-LPE). In essence, data from previous data samples are used to predict the current data sample. Each data sample is then encoded which is a sum of bits representing the coefficients of the linear combination of previous samples and an encoded form of the difference between the predicted and actual sample. This produces 260 bits every 20 ms, for a raw data rate of 260 bits/20 ms = 13 kbps. The GSM speech coder is based on the Residually Excited Linear Predictive Coder (RELP), enhanced by a Long-Term Predictor (LTP).

The GSM speech coder incorporates a voice activity detector in the speech coder because the speech signal is present for less than 40% of the time on an average. Based upon their significance in contributing to speech quality, the speech coder output bits are ordered into selected groups for error protection. This can be divided into three classes:

- *Class Ia* 50 bits, most sensitive to bit errors
- *Class Ib* 132 bits, moderately sensitive to bit errors
- *Class II* 78 bits, least sensitive to bit errors

The Class Ia 50 bits are protected by a 3-bit Cyclic Redundancy Check (CRC) error detection code, in which 3 parity check (CRC) bits are added. If an error is detected, the entire sample is discarded and replaced by a modified version of the preceding block. This facilitates the detection of non-correctable errors at the receiver.

These 53 bits (50 Class Ia bits + 3 parity bits) plus the next 132 Class Ib bits are reordered and appended by a 4-bit tail sequence of 0s, thus providing a data block of 189 bits. This block is then error-protected by a convolutional (1, 2, 5) error-correcting code having a rate of 1/2 with constraint length $L = 5$, resulting in a sequence of $189 \times 2 = 378$ bits.

The least important Class II 78 bits are unprotected and are concatenated to the existing sequence of 378 protected bits to produce a block of 456 bits in a 20 ms frame. Thus, this error-protection coding scheme increases the gross data rate of the GSM speech signal, with channel coding, to 456 bits/20 ms = 22.8 kbps, which is the GSM traffic channel data rate.

To summarise the packet formation of voice traffic, each 20 ms of the coded speech at 13 kbps forms a 260-bit packet. In this encoding scheme, there are three classes of speech-coded bits. The first 50 most significant Class Ia bits receive a 3-bit CRC code protection first, and then they are added to the Class Ib 132 bits with slightly lower significance and a 4-bit tail that is all zeros. The resulting $132 + 53 + 4 = 189$ bits are then encoded with a convolutional encoder that results into 378 bits. The convolutional code provides for error-correction capabilities. Figure 11.24 shows how the 456-bit packets are formed from the speech signal.

The 378 coded bits are added to the 78 least important Class II bits to form a 456-bits packet every 20 ms. In this way, the Class Ia 50 bits receives both CRC error detection and the rate convolutional

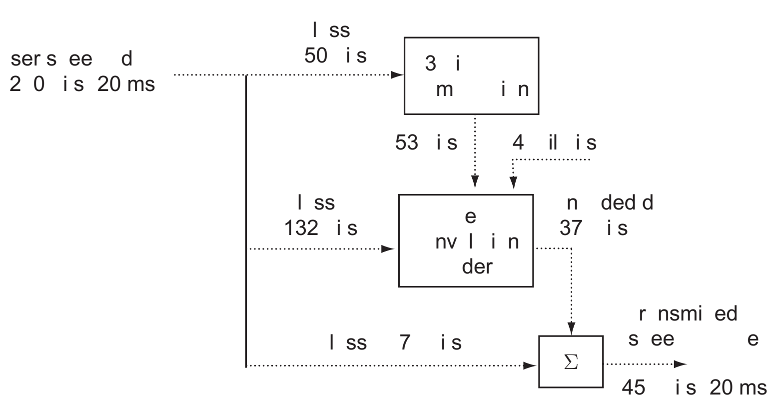

Fig. 11.24 *Coded speech packets in GSM*

error-correcting coding protection. The Class Ib 132 bits receive only the convolutional encoding protection, and the Class II 78 bits receives no protection. The 456-bit packets are used to form normal transmission bursts.

EXAMPLE 11.7 Gross channel data rate

GSM uses the RPE-LTP speech coder in which the encoding is done on blocks of samples of 20-ms duration (260 bits of coder output). The most significant first 50 bits (Class Ia) are appended with 3 CRC bits, the next 132 bits (Class Ib) are appended by 4 tail bits and concatenated with the first error-protected bits. This block is then convolutionally encoded with a rate 1/2 FEC coder, and then concatenated with last 78 bits (Class II). Show that the achievable gross channel data rate is 22.8 kbps.

Solution

Step 1. *To compute number of encoded Class Ia bits*

Number of uncoded Class Ia bits = 50 bits (given)

Number of CRC bits = 3 bits (given)

Number of encoded Class Ia bits = 50 + 3 = 53 bits

Step 2. *To compute number of encoded Class Ib bits.*

Number of uncoded Class Ib bits = 132 bits (given)

Number of tail bits = 4 bits (given)

Number of encoded Class Ib bits = 132 + 4 = 136 bits

Step 3. *To compute number of concatenation of encoded Class Ia + Class Ib bits*

Number of encoded Class Ia bits = 50 + 3 = 53 bits (as computed in Step 1)

Number of encoded Class Ib bits = 132 + 4 = 136 bits (as computed in Step 2)

Therefore, number of concatenated bits = 53 bits + 136 bits = 189 bits

Step 4. *To compute number of convolutionally encoded bits*

FEC coder rate of convolutional encoder = 1/2 (given)

Number of convolutionally encoded bits = 2 × 189 bits = 378 bits

Step 5. *To compute number of encoded bits to be transmitted*

Number of Class II bits = 78 bits (given)

Number of encoded bits to be transmitted = 378 + 78 = 456 bits

Step 6. *To determine achievable gross channel data rate*

Duration of transmission = 20 ms (given)

Therefore, gross channel bit rate = 456 bits/20 ms

Hence, gross channel bit rate = 22.8 kbps

11.6.1 Interleaving User Traffic Data

The purpose of interleaving user traffic data is to improve the signal quality by distributing the effects of fading among several mobile subscribers receiving data simultaneously from base stations. The user traffic data in frames of 456 bits are interleaved into the transmitted normal bursts in blocks of 57 bits plus one flag bit. The 456 bits are produced every 20 ms. The interleaving process specifies the method that maps the 20 ms of the traffic into the 456 bits as shown in Fig. 11.25.

To add protection against burst errors during wireless transmission, each 456-bit encoded data within each 20 ms traffic frame is divided into eight 57-bit sub-blocks forming a single speech frame, which are transmitted in eight consecutive time slots. Because each time slot can carry two 57-bit user data bits, each burst carries data from two different speech samples. The speech data are encrypted 114 bits at a time, assembled into time slots, and finally modulated for transmission. If a burst is lost due to interference or fading, channel coding ensures that enough bits will still be received correctly to allow the error correction to work satisfactorily.

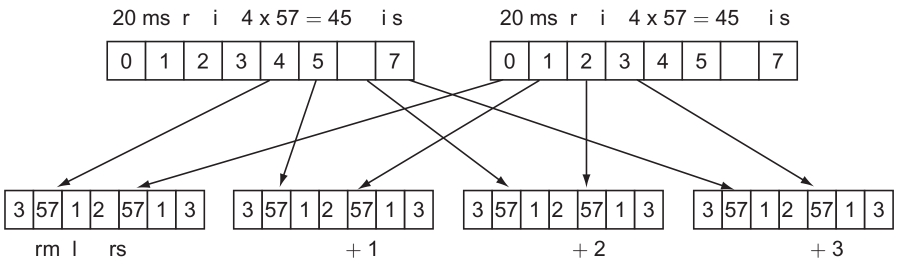

Fig. 11.25 *Interleaving traffic frames onto TDMA GSM frame*

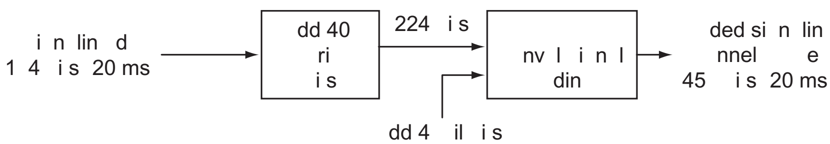

Fig. 11.26 *Coded signaling packet in GSM*

11.6.2 Packets of Signaling Channel

A number of signaling or control channels are used to determine how the traffic data are in the network. The formation of coded signaling packets is depicted in Fig. 11.26. Signaling channels employing the normal burst use 184 signaling bits in 20-ms duration to convey the signaling message. These bits are first block coded with 40 additional parity check bits and 4 tail bits to form a 228-bits block. The 228-bits block is then coded with a convolutional encoder having a rate. The output of a 456-bits packet occupying a 20-ms slot is the required packet of signaling channel which is transmitted.

11.6.3 Packets of Data Traffic

Figure 11.27 shows the formation of the 456-bit encoded data packet in 20 ms for user's data packet at transmission data rate of 9,600 bps.

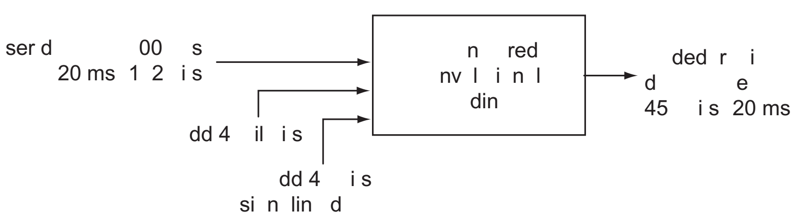

Fig. 11.27 *Coded data packets in GSM*

The 192 bits (9600 bps $\times$ 20 ms) of user's data information is accompanied by 48 bits of signaling data and 4 tail bits to form a ($192 + 48 + 4 =$) 244 bits packet. It is encoded using a rate punctured convolutional encoder. Punctured coding can eliminate the need for duplicating the number of transmitted bits by puncturing a certain number of bits. The resulting coded data packet of 456 bits are transmitted in normal bursts.

11.6.4 Channel Coding for Traffic/Control Data

Traffic data is processed in blocks of 240 bits every 20 ms for the actual supported data rates of 9.6 kbps, 4.8 kbps, and 2.4 kbps. Each block is augmented by four tail bits. A convolutional code (1, 2, 5) is used to produce a block of $244 \times 2 = 488$ bits. Then 32 bits of this block are puncturing, resulting into a data block of 456 bits. Each burst carries information from 5 or 6 consecutive data blocks. A bit-interleaving scheme is then used to spread the data over multiple bursts. The 488 bits are spread over 22 bursts in the following fashion:

- The 1st and 22nd bursts carry 6 bits each.
- The 2nd and 21st bursts carry 12 bits each.
- The 3rd and 20th bursts carry 18 bits each.
- The 4th through 19th bursts carry 24 bits each.

Control channel messages are defined to be 184 bits long, and are encoded using a shortened binary cyclic code. This is followed by a half-rate convolutional coder. The 184 message bits are concatenated with 40 parity bits plus four tail bits. The 228-bit data block are then applied to the convolutional coder (2, 1, 5). The 456 encoded bits are interleaved onto eight consecutive frames.

11.7 AUTHENTICATION AND SECURITY IN GSM

Security in cellular systems is implemented to prevent unauthorised use of the mobile subscriber number over the air. The voice conversations need to be encrypted to the extent possible. This is achieved using secrecy algorithms in GSM. Authentication in GSM is done with the help of a pre-defined protocol that is used to compare the IMSI of the MS reliably. The authentication process is shown in Fig. 11.28.

The SIM cards have a microprocessor chip that can perform the computations required for security purposes. A unique secret key is stored on the SIM card. This key is used in two algorithms A3 and A8 that are used for authentication and confidentiality respectively. The size of the secret key is 128 bits, and the response to the challenge is 32 bits long which is not very secure.

For authentication purpose, the secret key is used in a challenge response protocol using the A3 algorithm between the MS and BSS. The A3 algorithm is used to authenticate each mobile subscriber by verifying the users' password within the SIM with the cryptographic key at the MSC. The A5 algorithm provides the scrambling for the 114 coded data bits sent in each time slot. Burst formatting adds binary data to the ciphered blocks, in order to help synchronisation and equalisation of the received signal. The secret key is used to generate a privacy key that is used to encrypt voice or data messages using the A8 algorithm. Security is further enhanced by changing the encryption algorithm from call to call. The control channel signals are encrypted using A5 encryption algorithm.

The secret algorithms A3 and A8 are shared between different systems. The random number used in the challenge, the response to the challenge, and the data encryption key is exchanged between the VLR and the HLR. The VLR verifies if the response generated by the MS is the same.

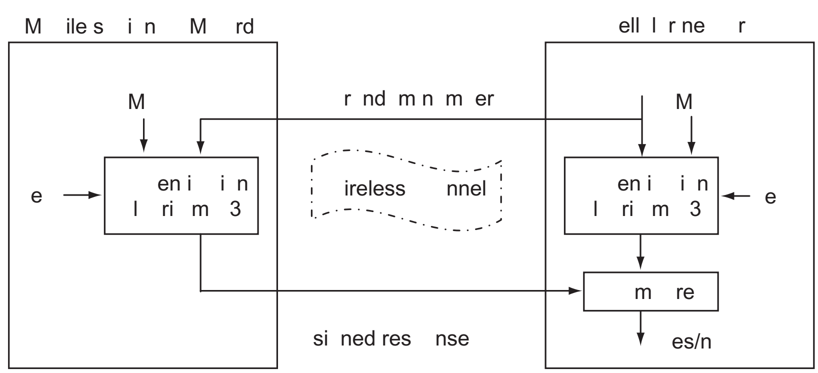

Fig. 11.28 *Authentication process in GSM*

When the MS requests for any service, the MSC sends it a random number. It also uses an authenticate algorithm to encrypt with the IMSI and the key stored in its memory. In the MS, the received random number is encrypted using IMSI and the same key is transmitted to the system, which compares it with the original value sent by the fixed network. If they match then the MS is authentic.

11.8 GSM CALL PROCEDURES

There are three mechanisms of call establishment which are embedded in all voice-oriented cellular communication networks that allow a mobile subscriber to establish and maintain a connection with the network. These mechanisms are registration of the mobile subscriber, call establishment, and hand-off procedures. Registration takes place as soon as the mobile subscriber unit is switched on. Call establishment occurs when the mobile subscriber initiates or receives a call. Hand-off mechanism enables the MS to change its connection link from one part of the network to another part.

11.8.1 Registration

For an MS to operate in an MSC, it must be registered by accessing the BTS. The MSC assigns a TMSI to the MS and updates the information in VLR and HLR. Whenever the MS is switched on, there is a need to possibly establish a new registration with the cellular network. Sometimes a mobile subscriber is required to connect to the cellular network at different locations through a BS that may not be owned by its home service provider. Technically speaking, the MS passively synchronises to the frequency, bit, and frame timings of the nearest BS to get ready for information exchange.

The MS receives the cell identity, and system parameters determine its location in the cellular network. If the present location is not the same as before, the MS initiates a registration procedure. During a registration procedure, the system provides the MS with a control channel for preliminary signaling. The MS provides its own identity, and finally the system authenticates the MS.

The MS registration process is described in Table 11.3 when a mobile subscriber is switched on in a new MSC area.

Firstly, a radio communication link is established between the MS and BTS to process the registration. It is followed by an authentication process by the system, and assignment of TMSI. Records are updated in the VLR and HLR. After successful registration, the temporary radio communication link is released.

Table 11.3 *MS registration procedure*

Step No.	*Action taken*	*Signal path*
1.	MS requests a channel	MS → BTS → BSC
2.	Channel activation response	BSC → BTS
3.	Acknowledgement for activation	BTS → BSC
4.	Assignment of channel	BSC → BTS → MS
5.	Location update request	MS → BTS → BSC → MSC
6.	Requests for authentication	MSC → BSC → BTS → MS
7.	MS authenticates	MS → BTS → BSC → MSC
8.	Verification of authentication	MSC ↔ VLR
9.	Assigns TMSI	MSC → BSC → BTS → MS
10.	TMSI acknowledgement	MS → BTS → BSC → MSC
11.	VLR/HLR updation	MSC ↔ VLR ↔ HLR
12.	Channel release	BSC → BTS → MS

In the mobile communication environment, there are two separate call-establishment procedures for mobile-to-network and network-to-mobile calls. Mobile-to-mobile calls are a combination of these two. The detailed procedure for both types of call establishment in the GSM network is described next.

11.8.2 Mobile-to-Network Call

The mobile subscriber monitors the BCH and gets synchronised to the nearest base station. By receiving the BCCH, FCCH, and SCH messages, the mobile subscriber is locked on to the system. To initiate a call, the mobile subscriber first dials the called subscriber number and presses the send button on the GSM mobile phone. The mobile subscriber transmits a burst of RACH data on the same ARFCN as the base station. The base station then responds with an AGCH message on the CCCH which assigns the mobile subscriber to a new channel for SDCCH connection. The mobile subscriber, receives its ARFCN and TS assignment from the AGCH on TS 0 of the BCH, and then immediately shifts to the new ARFCN and TS.

The step-by-step procedure for mobile-originated call establishment is described in Table 11.4.

Table 11.4 *Mobile-to-network call procedure*

Step No.	*Action taken*	*Signal Path*
1.	MS requests a channel	MS → BTS → BSC
2.	Assignment of channel	BSC → BTS → MS
3.	Call-establishment request	MS → BTS → BSC → MSC
4.	Requests for authentication	MSC → BSC → BTS → MS
5.	MS authenticates	MS → BTS → BSC → MSC
6.	Ciphering process	MSC → BSC → BTS → MS
7.	Ciphering successful	MS → BTS → BSC → MSC
8.	Called subscriber number	MS → BTS → BSC → MSC
9.	Call routing confirmation	MSC → BSC → BTS → MS
10.	Traffic channel request	MS → BTS → BSC
11.	Traffic channel assigned	BSC → BTS → MS
12.	Free/busy signal	BTS → MS
13.	Acceptance of call	MSC → BSC → BTS → MS
14.	Connection established	MS → BTS → BSC → MSC
15.	Conversation	MS ↔ BTS ↔ BSC ↔ MSC

The mobile subscriber first waits for the SACCH frame on the SDCCH, to inform the mobile subscriber of any required timing advance and transmitter power command. The mobile subscriber is now able to transmit normal burst messages as required for speech traffic. The SDCCH sends messages such as authentication and user validation between the mobile subscriber and the base station. The PSTN connects the called landline subscriber to the MSC. The MSC switches the voice connection to the serving base station. The mobile subscriber receives a command by the base station via the SDCCH to retune to a new ARFCN and TS for the TCH assignment. Speech data is transferred on both the forward and reverse channels, and the SDCCH is released.

11.8.3 Network-to-Mobile Call

To make a call from a landline telephone subscriber, the call-request information is processed through the gateway MSC to the destination MSC after getting the information from the home HLR of the called mobile

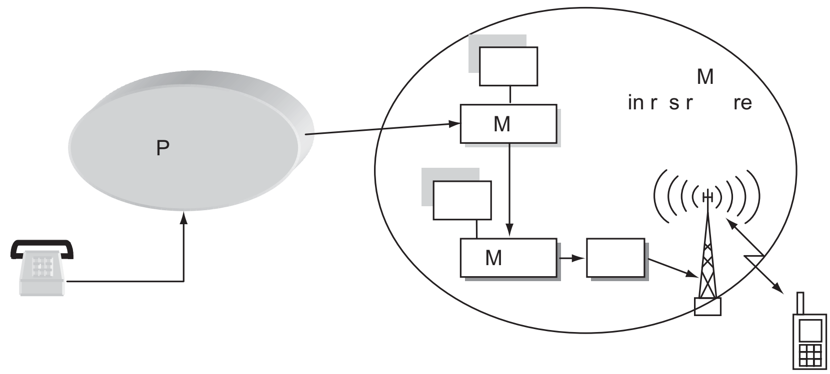

Fig. 11.29 *Network-to-mobile call scenario in a visiting network*

subscriber. Then the called mobile subscriber is contacted through the BSS. The base station broadcasts a PCH message on the BCH. The mobile subscriber locks on to the same ARFCN, decodes its page and responds with an RACH message. The base station then uses the AGCH on the CCCH to assign the mobile subscriber unit to a new channel for connection to the SDCCH and SACCH while the network and the serving base station are connected. Once the subscriber establishes timing advance and authentication on the SDCCH, the base station assigns the TCH. As shown in Fig. 11.29, the PSTN directs the call initiated by the calling landline telephone subscriber to the MSC identified by the dialed phone number of the called mobile subscriber.

The MSC requests routing information from the HLR. Since the mobile subscriber is roaming in the area of a different MSC, the gateway MSC contacts the destination MSC. The VLR initiates a paging procedure in all BSSs under the control of the MSC. After getting response from the MS, the VLR sends the necessary system parameters to the MSC to establish the link to the MS.

EXAMPLE 11.8 GSM Layer III sublayer categories used for call establishment

Illustrate the step-by-step procedure for mobile-initiated call-establishment procedure, giving the type of logical channel used along with sublayer categories of GSM Layer III.

Solution

Table 11.5 shows the mobile-initiated call-establishment procedure. The second column identifies the action taken or the message type. The third column identifies the logical channel that is used to carry the message. The fourth column identifies the sublayer of the Layer III in which GSM standard describes the message. Note that Layer III does not handle the traffic message, and therefore there is no sublayer associated for that part of the procedure.

Table 11.5 *Layer III sublayer categories for mobile-initiated call*

Step No.	*Message type*	*Logical channel used*	*Layer III sublayer category*
1.	Request for a channel	RACH	RRM
2.	Immediate assignment	AGCH	RRM
3.	Request for Call establishment	SDCCH	CM
4.	Request for authentication	SDCCH	MM
5.	Authentication response	SDCCH	MM
6.	Ciphering command	SDCCH	RRM
7.	Ciphering response	SDCCH	RRM

(*Continued*)

Table 11.5 (*Continued*)

Step No.	*Message type*	*Logical channel used*	*Layer III sublayer category*
8.	Send called number	SDCCH	CM
9.	Routing response	SDCCH	CM
10.	Traffic channel assigned	SDCCH	RRM
11.	Traffic channel established	FACCH	RRM
12.	Free/busy signal	FACCH	CM
13.	Acceptance of call	FACCH	CM
14.	Connection established	FACCH	CM
15.	Conversation	TCH	----

11.9 GSM HAND-OFF PROCEDURES

Hand-off is usually initiated because of signal strength deterioration at the edge of a cell boundary. Hand-off in GSM is divided into four major categories:

(a) Intracell-Cum-Intra-BTS Hand-off This type of hand-off is necessary when high interference occurs during the call. The channel for the connection is changed within the cell by moving to another frequency of the same cell or to another time slot of the same frequency. The hand-off process is initiated by the base station.

(b) Intercell-Cum-Intra-BSC Hand-off In this type of hand-off, the change is in the radio channel between two cells that are served by the same BSC. Initially, the hand-off request is initiated by the serving BSS to the MSC. The MSC transmits the hand-off request to the destination BSS. After receiving acknowledgement, the MSC gives the hand-off command. The mobile subscriber transmits a hand-off complete message to the new BSS, which relays it to the MSC for releasing the earlier occupied channel.

(c) Inter-BSC-Cum-Intra-MSC Hand-off A call connection is changed between two cells that are served by different BSCs but operate in the same MSC. When the measured value of the received signal strength at the mobile subscriber is lower than the threshold value, it informs the serving BSC which initiates the hand-off command to the MSC of that area. The MSC relays the hand-off request to the BSC, which sends a channel activation request to its BTS. It is then possible for the hand-off call to be handled in the new MSC.

The BTS provides the MS with a list of available channels in neighbouring cells via the BCCH. The mobile subscriber monitors the received signal strength from the BCCHs of these neighbouring cells and reports these measured data to the MSC using the SACCH. This is called mobile-assisted hand-off. The BTS also monitors the received signal strength from the mobile subscriber to make a hand-off decision. The MSC negotiates a new channel with the new BSS and indicates to the mobile subscriber that a hand-off should be made using a hand-off command. Upon completion of the hand-off, the mobile subscriber confirms with a hand-off complete message to MSC.

Figure 11.30 shows the hand-off procedure between two BSSs that are controlled by one MSC.

(d) Inter-MSC Hand-off A connection is changed between two cells that are in different MSCs. This situation occurs in case of roaming. This hand-off occurs wherein the home MSC is notified of the hand-off condition through the PSTN, and the home MSC sends the necessary data to the new MSC through the PSTN again.

11.10 GSM SERVICES AND FEATURES

GSM is an integrated voice-data service that provides a number of value-added services. A telecommunication service supported by the GSM is defined as a group of communication capabilities that the service provider

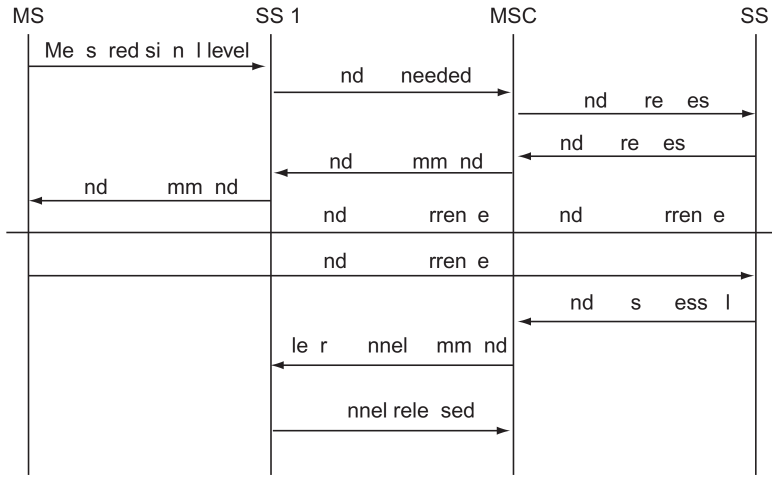

Fig. 11.30 *Hand-off involving a single MSC and two BSSs*

offers to the mobile subscribers. As per ISDN guidelines, GSM services are classified mainly into three categories: telephone services or teleservices, data or bearer services, and supplementary ISDN services.

Teleservices provides the mobile subscribers with necessary capabilities which enable them to communicate with other subscribers. Bearer services give the mobile subscribers the capacity required to transmit appropriate signals between network access points. Supplement services supplement basic teleservices and are offered together or in association with other services. Table 11.6 lists the GSM services.

Telephone services provide full-duplex voice communication applications between the calling and called subscribers according to a standard protocol.

Data or bearer services provide capabilities to transmit information among user-network-interfaces. Traditional bearer services include a variety of asynchronous and synchronous data access to PSTN/ISDN and packet switched public data networks either in a transparent mode (where GSM provides standard channel coding for the user data) or nontransparent mode (where GSM offers special coding efficiencies based on the particular data interface).

Supplementary ISDN services are not standalone services but they are digital signaling services that supplement telephone services or data services. Short Messaging Service (SMS) allows GSM subscribers and base stations to transmit alphanumeric pages of limited length (160 7-bit ASCII characters). SMS also provides cell broadcast, which allows GSM base stations to repetitively transmit messages with as many as fifteen 93-character strings in a concatenated fashion. SMS may be used for advertisement, safety, and advisory applications including broadcast of highway traffic conditions or weather information to all GSM subscribers within reception range.

Facts to Know!

SMS is a proactive bearer and is an always ON network. Through SMS, the popular value-added services such as news/stock quotes service, email through SMS, health-care services, alert services can be provided.

11.10.1 Short Message Service (SMS)

The Short Message Service (SMS) is the ability to send or receive a text message to or from mobile subscriber phones. GSM system supports SMS messages using unused available bandwidth and has several unique features. These features include automatic confirmation of message delivery. SMS can be transmitted and received simultaneously with voice, data, or fax calls.

Table 11.6 *GSM services*

Service category	*Service type with comments*
Telephone services	– Telephony voice at 13 kbps (full rate) – Emergency calls at specific emergency number – Short Messaging Service (SMS): Point to point and cell broadcast types – Message handling and storage services – Videotext, Teletex, FAX transmission – Half-rate speech coder, and Enhanced full rate: Optional implementation
Data or bearer services	– Rate adapted subrate information circuit-switched asynchronous duplex data @ 300 bps –9,600 bps (transparent/nontransparent) – Rate adapted subrate information circuit-switched synchronous data @ 2400 bps –9,600 bps transparent – Access to X.25 public data networks packet service for synchronous duplex packet data @ 2400 bps –9,600 bps – Access to PAD (Packet Assembler/Disassembler) functions for asynchronous data @ 300 bps –9,600 bps – Speech and data swapping during a call – Support of ARQ technique for improved error rates – Modem selection of 3.1 kHz audio service when interworking with ISDN
Supplementary ISDN Services	– Call forwarding or call diversion: All calls, when subscriber is not available – *Call barring* Outgoing calls with specifications – Caller or Calling Number Identification Presentation (CNIP) or restriction of displaying the caller's ID (CNIR) – Connected number ID Presentation (CNOP) or restriction of displaying the called ID (CNOR) – Malicious Call Identification (MCI) – *Call orwarding* Unconditional or when mobile busy or no reply or mobile not reachable – Call transfer to another mobile phone – Mobile access hunting allows multiple phones to be called in sequence – *Call Waiting* Incoming call during current conversation – *Call Holding* Put current call on hold to answer another – Completion of call to busy subscriber – *Multiparty communication* Up to five ongoing calls can be included in conference-type conversation – Closed user group – *Advice of charge* Online charge information, free phone service, reverse charging – *Operator determined call barring* Barring all originating calls or all incoming calls or outgoing international calls or incoming calls when roaming or restriction of certain features from individual subscribers by the operator – User-to-user signalling for sending user data to another GSM or ISDN phone

This is possible because voice/data/fax calls utilise dedicated voice channels for the duration of the call, while short messages occupy the control channels.

SMS is basically a store and forward service. If the recipient mobile subscriber is not available, the message can be stored with the system. Each mobile cellular network that supports SMS has one or more messaging centres to handle and manage the short messages. This means that SMS message is not sent directly from a sender to the receiver but always processed via an SMS centre at MSC.

A single SMS can be up to 160 characters of text in length. The characters may comprise of a combination of words, numbers, or alphanumeric characters. Nontext-based SMSs (binary format) are also supported. There are ways of sending multiple SMSs also. For example, SMSs concatenation and SMS compression within single short message features are available.

11.10.2 GSM Service Quality Requirements

The GSM standards specify various requirements on the quality of service delivered to the mobile users. Some of these service requirements are

- The time from switching to service ready of 4 seconds in the home service area
- The time from switching to service ready of 10 seconds in the visiting service area
- A connect time of 4 seconds to the called network
- A release time of 2 seconds to the called network
- The time to alert a called mobile subscriber of an incoming call of 4 seconds in the first attempt and 15 seconds in the subsequent attempts
- A maximum time gap due to hand-off of 150 ms in intercell handover
- A maximum time gap due to hand-off of 100 ms in intracell handover
- A maximum one-way speech delay of 90 ms
- An intelligibility of speech of 90%
- Probability of call release failure rate of less than 0.02%
- Probability of misconnection, incorrect charging, no tone or similar failure of less than 0.01%
- Probability of losing HLR/VLR messages of less than 0.000001%
- Mean accumulated downtime for one cell-site of less than 30 minutes per year

On the average, a fault that causes more than 50% of the established calls to be disconnected prematurely will occur less than once a year. The service availability of an MSC is expressed in terms of the frequency or duration of loss of service. The loss of service to particular switching circuits, groups of switching circuits, subsystems, or the complete MSC is determined by the nature of faults in the MSC. The average cumulative duration of service denial due to faults affecting more than 50% of the switching circuits will not exceed three minutes during the first year of operation and two minutes during each subsequent year.

11.10.3 Power Classes in GSM

Power management in the system helps the service provider to control the interference among the mobile subscribers and minimise the power consumption at the mobile unit. Therefore, power management has direct impact on quality of service and the battery life. There are three major classes of mobile phones: vehicle mounted, portable, and handheld mobile phones. Vehicle-mounted mobile phones use the car battery, portable mobile phones use larger rechargeable batteries, and handheld mobile phones use low-capacity smaller rechargeable batteries. The antenna for the vehicle-mounted mobile phones is mounted outside the car, which is away from the user's body. The antenna for the portable mobile phones is mounted along with the transceiver.

Facts to Know!

The antenna in handheld mobile phones is next to the ear and brain of the user which raises health concerns for high-radiated powers.

GSM cells have radii ranging from 300 m to 35 km. The size of the cell determines the required transmitted power for the BTS and the MS. To allow manufacturers and service providers to accommodate the diversified requirements for different BTS and MS, a number of radiated power classes are identified by the GSM standard. There are five power classes for the mobile phones from +29 dBm (0.8 W) up to +44 dBm (20 W) with a 4-dB separation between consecutive mobile classes. There are eight classes for the BTS radiated power ranging from +34 dBm (2.5 W) up to +55 dBm (320 W) in 3-dB steps.

EXAMPLE 11.9 GSM transmitter powers

Tabulate the transmitter powers EIRP (watts) for all classes of GSM base stations and mobile stations.

Solution

Table 11.7 shows EIRP levels for all eight classes of GSM base stations and five classes of GSM mobile phone equipments.

Table 11.7 *GSM transmitter power classes*

Power class	*Base station power (Watts)*	*Mobile phone power (Watts)*
I	320	20
II	160	8
III	80	5
IV	40	2
V	20	0.8
VI	10	---
VII	5	---
VIII	2.5	---

The mobile subscriber Tx power is always controlled to its minimum required value to minimise the interference and maximise the battery life. Power adjustments in both directions are made using closed-loop power control. The mobile subscriber is allowed to reduce its peak output power down to 20 mW in 2 dB steps. The BSS calculates the optimum Tx power level for individual mobile subscribers by monitoring certain metrics of performance such as the received signal power level, received signal-to-noise ratio, or received bit error rate, and sends this information through control signaling packets to the mobile subscriber.

11.10.4 Frequency Hopping in GSM

In voice-oriented cellular networks, corrupted data packets are either discarded or retained with incorrect value. In both these cases, distortion occurs in the voice signal at the receiver. If the frequency channel coincides with a deep frequency selective fading or when the cochannel interference is excessive, the distortion in the received voice signal will persist until the mobile subscriber moves to some other location such that the frequency selective fading pattern is changed or the cochannel interference is reduced.

One method to reduce the effect of the frequency selective fade or excessive cochannel interference is to provide for a slow frequency-hopping pattern. This option is exercised in the GSM system that supports an optional frequency-hopping pattern of 217.6 hops per second. When multipath fading is a problem, the GSM system allows for frequency hopping. All GSM mobile phones are capable of frequency hopping, but only those cells that are located in areas of severe fading are designated as hopping cells. The system can hop only among the frequencies that are assigned to the cell, so there will be only a few hopping possibilities. Thus GSM is not really a true spread-spectrum system, but rather a TDMA/FDM system with some spread-spectrum capability added on.

EXAMPLE 11.10 Maximum frequency hopping in GSM

In Europe, GSM uses the frequency band of 890 to 915 MHz for uplink transmission, and the frequency band of 935 to 960 MHz for downlink transmission. Determine the maximum frequency hop from one frame to the next for uplink transmission and downlink transmission. Express it as a percentage of the mean carrier frequency.

Solution

Step 1. *To find RF bandwidth for uplink transmission*

Frequency band for uplink transmission = 890 MHz – 915 MHz (given)

RF Bandwidth for uplink transmission = 915 – 890 = 25 MHz

Step 2. *To find RF bandwidth for downlink transmission*

Frequency band for downlink transmission = 935 MHz – 960 MHz (given)

RF Bandwidth for downlink transmission = 960 – 930 = 25 MHz

Step 3. *To find maximum frequency hop*

Therefore, in either case, the maximum frequency hop or change from one frame to the next could be 25 MHz.

Step 4. *To express maximum frequency hop in %age.*

For uplink transmission,

Mean carrier frequency = 890 + (915 – 890)/2 = 902.5 MHz

Maximum frequency hopping = 25/902.5

Hence, maximum frequency hopping = 0.0277 or 2.77%

For downlink transmission,

Mean carrier frequency = 935 + (960 – 935)/2 = 947.5 MHz

Maximum frequency hopping = 25/947.5

Hence, maximum frequency hopping = 0.0264 or 2.64%

The maximum RF spectrum bandwidth for uplink or downlink transmission is 25 MHz. Expressed as a percentage of the mean carrier frequency of 900 MHz, the maximum frequency hopping for the downlink is approximately $25/900 \times 100 = 2.8\%$.

With this percentage of maximum frequency hopping, it turns out that the time spent by a rapidly moving mobile user in a deep fade is reduced to about 4.6 ms. This incidentally matches with the frame duration. In the case of slowly moving mobile subscribers such as pedestrians, the frequency-hopping algorithm produces substantial gains against fades. Each successive frame in a given channel is carried on a different carrier frequency. Thus, the transmission frequency is changed once every 4.615 ms.

Key Terms

- Air interface
- Authentication
- Authentication Centre (AuC)
- Authorisation
- Base Station (BS)
- Base Station Subsystem (BSS)
- BCCH
- Downlink
- Encryption
- Frame
- Frequency Division Multiple Access
- Frequency hopping
- Global System For Mobile (GSM)
- Hand-off
- Mobile Station (MS)
- Network and Switching Subsystem (NSS)
- Packet
- Privacy
- Registration
- Short Message Service (SMS)
- Subscriber Identity Module (SIM)
- Superframe
- Time Division Multiple Access (TDMA)
- Time slots
- Uplink

Summary

This chapter has presented GSM as the most successful second-generation digital cellular network, which is an ETSI standard for 2G pan-European digital cellular in the frequency bands of 890–915 MHz and 935–960 MHz. GSM uses a combination of FDMA and TDMA, with a carrier channel bandwidth of 200 KHz and 8 time slots per carrier. The MS contains a unique SIM smart card that identifies the specifications of a user such as address and type of services subscribed. Another aspect of security in GSM is that the secret key information is not shared between systems. The system provides voice communication as well as limited data (SMS) transfer capability. GSM is a relatively mature technology, now several years in existence with a huge customer base across the world. Although GSM is primarily designed for voice transmission using a circuit-switched network, yet it evolved toward a more data-oriented transfer via GPRS using packet-switched system. Another competing 2G cellular technology, IS-95 CDMA standards, having different air interface and high potential to offer flexible and higher system capacity, is the main focus in the next chapter.

Short-Answer Type Questions with Answers

A11.1 How does a SIM card provide security against fraudulent use of GSM phone
A GSM mobile phone is useless without a valid SIM card. SIM cards carry the private information for a user. The SIM can be set up to require the user to enter a four-digit Personal Identification Number (PIN) whenever the mobile phone is switched on. This feature provides some security in case of its fraudulent use from a lost or stolen mobile phone. If the mobile subscriber removes the SIM card when leaving the mobile phone in a vehicle, the mobile phone cannot be used unless another person has a valid SIM.

A11.2 Why are the needs for the wireless and wired media different
The wireless medium is unreliable, bandwidth limited, and needs to support mobility. As a result, protocols used in the wireless and wired media are different. The BSS provides for the necessary translation among these protocols.

A11.3 What is the necessity of maintaining two databases HLR and VLR at MSC
Two databases, HLR and VLR, are used to keep track of current location of an MS in GSM. Maintenance of two databases at home and at the visiting location allows a mechanism to support dialing and call routing in a roaming situation where the MS is visiting the coverage area of a different MSC.

A11.4 List some of the main responsibilities of the Radio Resource Management (RRM) sublayer of the network layer.
The main responsibilities of the RRM include the assignment of the radio channel and hopping to new channels in implementation of the slow frequency-hopping option, managing hand-off procedure and measurement reports from MS for hand-off decision, to implement power control procedure, and to adapt to timing advance for synchronisation purpose.

A11.5 What is the significance of Temporary Mobile Subscriber Identity (TMSI)
As all transmissions between MS and BSS are sent through the air interface, there is a constant threat to the security of the information exchanged. A temporary identity TMSI is usually sent in place of International Mobile Subscriber Equipment Identity (IMSEI).

A11.6 What are the main functions of Frequency Correction Control Channel (FCCH)
The FCCH is used by the BTS to broadcast frequency references and frequency correction burst of 148 bits length. An MS in the coverage area of a BTS uses the broadcast FCCH signal to synchronise its carrier frequency and bit timing. The FCCH is a special data burst that occupies TS 0 for the very first GSM frame (frame 0) and is repeated every ten frames within a control channel multiframe. The

physical Frequency Correction Burst (FCB) is used to implement the logical FCCH.

A11.7 Which channels are defined as CCCH logical control channels
There are three CCCH logical channels: The Paging Channel (PCH), the Random Access Channel (RACH), and the Access Grant Channel (AGCH). The PCH is a forward link channel and is used by the BTS to page or notify a specific individual MS for an incoming call in the cell. The RACH is a reverse link channel and is used by the MS either to access the BTS requesting the dedicated channel for call establishment or to acknowledge a page from the PCH. The AGCH is used by the base station to provide forward-link communication to the mobile for implementation of the acknowledgement from the BTS to the MS after a successful attempt by MS using RACH in a previous CCCH frame.

A11.8 Distinguish between the full-rate and half-rate traffic channel.
The full-rate traffic channel (TCH/F) uses a 13 kbps speech-coding scheme and 9,600 bps, 4,800 bps, and 2,400 bps data. After including signaling overhead, each full-rate traffic channel has a gross bit rate of 22.8 kbps. When transmitted as full-rate, user data is contained within one time slot per frame. The Half-rate Traffic Channel (TCH/h) has a gross bit rate of 11.4 kbps. It supports 4800 bps and 2400 bps data rate only. Two half-rate traffic channel users would share the same time slot, but would alternately transmit during every other frame.

A11.9 How are changes in propagation delay compensated due to movement of the mobile subscriber with respect to the fixed cell-site
The cell-site instructs the mobile subscriber to advance or retard the timing of its transmissions to compensate for the changes in propagation delay as it moves about in the cell. In this way, the transmission delay problem is avoided on the traffic channels.

A11.10 What are the specific requirements which need to be considered in designing an appropriate frame structure in TDMA based communication systems
Frequency band of operation, channel bandwidth, number of time slots in a TDMA frame, maximum cell radius and vehicle speed, maximum allowable delay spread and coding delay are the specific requirements which need to be considered in designing an appropriate frame structure in TDMA-based communication systems.

A11.11 Mention various types of packet data bursts used for control and traffic signaling.
GSM supports five types of packet data bursts used for control and traffic signaling: the normal data burst used for TCH and DCCH transmissions on both the forward and reverse link; the frequency-correction data burst to broadcast the frequency synchronisation control messages; the synchronisation data burst used for the specific task of identifying the network; the random access data burst used by the MS to access the BS for registration with the network; and the dummy burst used as filler information for unused time slots on the forward link.

A11.12 Why is a voice activity detector incorporated in the GSM speech coder
The GSM speech coder incorporates a voice activity detector in the speech coder which is based on the Residually Excited Linear Predictive Coder (RELP), enhanced by a Long-Term Predictor (LTP). Generally, the speech signal is present for less than 40% of the time on an average. Based upon their significance in contributing to speech quality, the speech-coder output bits are ordered into selected groups for error protection.

A11.13 What are different categories of hand-off procedures in GSM
Hand-off in GSM is divided into four major categories: intracell-cum-intra-BTS hand-off, intercell-cum-intra-BSC hand-off, inter-BSC-cum-intra-MSC hand-off, and inter-MSC hand-off.

A11.14 Classify GSM services into different categories.
GSM services are mainly classified into three categories as per ISDN guidelines: telephone services or teleservices, data or bearer services, and supplementary ISDN services. Teleservices provide the mobile subscribers ability to communicate with other

subscribers. Bearer services give the mobile subscribers the capacity required to transmit appropriate signals between network access points. Supplement services supplement basic teleservices and are offered together or in association with other services.

A11.15 Distinguish between vehicle-mounted portable and handheld mobile phones.
There are three major classes of mobile phones: vehicle-mounted, portable, and handheld mobile phones. Vehicle-mounted mobile phones use the car battery, portables mobile phones use larger rechargeable batteries, and handheld mobile phones use low-capacity smaller rechargeable batteries. The antenna for the vehicle-mounted mobile phones is mounted outside the car, which is away from the user's body. The antenna for the portable mobile phones is mounted along with the transceiver. The antenna in the handheld mobile phones is next to the ear and brain of the user, which raises health concerns for high-radiated powers.

Self-Test Quiz

S11.1 The SIM card used in GSM mobile phone has ________ memory.
(a) 4 kb (b) 8 kB
(c) 16 kB (d) 64 kB

S11.2 The number of time-slots per RF channel bandwidth in GSM standard is
(a) 3 (b) 6 (c) 8 (d) 16

S11.3 The frame period of one TDMA frame in GSM standard is
(a) 3.692 μs (b) 577 μs
(c) 4.615 ms (d) 40 ms

S11.4 The ________ translates between the wireless interface and fixed wired infrastructure protocols.
(a) MS (b) BSS
(c) MSC (d) OMC

S11.5 The standard interface that connects a BTS to a BSC is called the ________ interface.
(a) U_m (b) A-bis
(c) A (d) D

S11.6 The ________ is the database at MSC that keeps the information about the identity of mobile phone equipment.
(a) HLR (b) VLR
(c) AuC (d) EIR

S11.7 The ________ standard interface allows a service provider to use base stations and switching equipment made by different manufacturers.
(a) U_m (b) A-bis
(c) A (d) D

S11.8 The ________ layer in GSM signaling protocol architecture specifies the modulation and coding techniques used in the system.
(a) physical (b) data Link
(c) networking (d) messaging

S11.9 The gross data rate of each carrier channel in GSM is
(a) 270.833 kbps
(b) 33.854 kbps
(c) 24.7 kbps
(d) 13.4 kbps

S11.10 The ________ is not a dedicated control channel but carries the same information as SDCCH.
(a) AGCH (b) SDCCH
(c) SACCH (d) FACCH

S11.11 The PCM speech coder has a data rate of ________, which is considerably high for use with wireless systems.
(a) 512 kbps (b) 256 kbps
(c) 128 kbps (d) 64 kbps

S11.12 One time slot of a TDMA frame in GSM standard contains ________ bits encrypted data.
(a) 156.25 (b) 114
(c) 57 (d) 26

S11.13 A user occupying a single time slot has to wait for time duration of ________ between two successive transmissions.
(a) 577 μs (b) 4.615 ms
(c) 120 ms (d) 6.12 s

S11.14 The purpose of ________ user traffic data is to improve the signal quality by distributing the effects of fading.
(a) speech coding (b) channel coding
(c) bit interleaving (d) equalisation

S11.15 The ________ modulation technique provides high spectrum efficiency and a constant amplitude signal.
(a) FSK (b) QPSK
(c) GMSK (d) OFDM

S11.16 There are ________ classes for the BTS radiated power ranging from +34 dBm up to +55 dBm in 3-dB steps.
(a) three (b) five
(c) eight (d) ten

S11.17 The mobile subscriber is allowed to reduce its peak output power down to 20 mW in ________ steps.
(a) 1 dB (b) 2 dB
(c) 3 dB (d) 4 dB

S11.18 When ________ is a problem, the GSM system allows for frequency hopping.
(a) cochannel interference
(b) adjacent channel interference
(c) near-to-far interference
(d) multipath fading

S11.19 The frequency-hopping algorithm employed in GSM provides change in the transmission frequency once every
(a) 6.12 s (b) 120 ms
(c) 4.615 ms (d) 577 μs

Answers to Self-Test Quiz

S11.1 (b); S11.2 (c); S11.3 (c); S11.4 (b); S11.5 (b); S11.6 (d); S11.7 (c); S11.8 (a); S11.9 (a); S11.10 (d); S11.11 (d); S11.12 (b); S11.13 (b); S11.14 (c); S11.15 (c); S11.16 (c); S11.17 (b); S11.18 (d); S11.19 (c)

Review Questions

Q11.1 What is meant by GSM Public Land Mobile Network (PLMN)? What are its objectives?

Q11.2 Draw the GSM reference architectural model and explain the following GSM subsystem entities:
(a) Mobile Station (MS) including its classification based on power
(b) Base Station Subsystem (BSS)
(c) Operation and Maintenance Subsystem (OMSS).

Q11.3 Why are so many different identifiers/addresses needed in GSM? Distinguish between user related and system related identifiers.

Q11.4 Describe the functions of MS and SIM. Why does the GSM separate MS and SIM? How and where is user related data represented/stored in the GSM system? How is the user data protected from unauthorised access, especially at the air interface?

Q11.5 Explain the VLR/HLR database approach used in GSM. How does this approach limit the scalability among moving users? Where are they physically located?

Q11.6 What is the difference between a physical channel and a logical channel? Describe the important functions of various types of GSM logical channels.

Q11.7 Explain the following GSM Interfaces:
(a) Mobile Station (MS) to Base transceiver Interface (BTS)
(b) BTS to BSC
(c) BSC to MSC
(d) Interfaces between other GSM Entities

Q11.8 What are different protocols used in GSM? Explain the protocol architecture in GSM.

Q11.9 Explain how hand-off takes place in GSM. What are the problems associated with hand-off in GSM? Explain the different types of hand-off encountered in GSM.

Q11.10 What are the different types of services offered by GSM?

Q11.11 How is subscriber authentication and data encryption done in GSM? Explain the three algorithms used for providing security in GSM, that is, A3, A5 and A8.

Analytical Problems

P11.1 Considering the frequency allocation strategy of the GSM systems, determine

(a) the total number of traffic channels per 50 MHz of bandwidth used for two-way GSM communications
(b) the total number of GSM channels per MHz of bandwidth
(c) the number of channels per cell for frequency reuse factor $K = 4$, $K = 7$

P11.2 Compute the individual data rates for the speech coder; speech error protection; SACCH; guard time, ramp-up, and synchronisation. Prove that the GSM system allocates gross RF data rate of 33.854 kbps/user by summing up these individual data rates.

P11.3 Compute the longest time over which a mobile station would have to wait in order to determine the frame number being transmitted by a GSM cell-site.

P11.4 Each 20 ms of the coded speech forms a 260-bit packet. Show that the speech coding rate for GSM signal transmission is 13 kbps and the effective transmission rate to support one 13 kbps coded voice channel is 22.8 kbps.

P11.5 Consider the multiframe transmission in GSM. Draw the overall structure of the multiframe, frame, and time slot to show that the transmission channel rate of the GSM is indeed 270.833 kbps.

P11.6 In each GSM multiframe structure, 24 frames are used for traffic and two for associated control signaling. Considering the detailed burst frame and multiframe infrastructure, show that the effective transmission rate for each GSM voice traffic is 22.8 kbps.

P11.7 Considering coded data packets in GSM, compute the net data rate (data plus signaling) and the effective transmission rate of a 9,600 bps GSM data service.

P11.8 The GSM cellular system uses a 270.833 kbps data rate to support eight users per frame.

(a) What is the raw data rate provided for each user?
(b) If guard time and synchronisation occupy 10.1 kbps, determine the traffic efficiency.
(c) If (7, 4) code is used for error handling, what is the overall efficiency?

P11.9 If 8 speech channels are supported on a single radio channel in GSM, and if no guard band is assumed,

(a) what is the number of simultaneous users that can be accommodated?
(b) what is the duration of a bit?
(c) if a user is allocated one time slot per frame, what is the delay between successive transmissions in successive frames?
(d) what is the longest time over which a mobile station would have to wait in order to determine the frame number being transmitted by its base station?

P11.10 If the trailing bits, stealing bits, guard bits, and training bits in a GSM frame are considered as overhead, and the rest of the bits as data then what is the percentage overhead in a GSM frame?

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12

CDMA Digital Cellular Standards (IS 95)

Code Division Multiple Access (CDMA) is the fastest-growing digital cellular technology. CDMA allows users to differentiate from one another by a unique code rather than a frequency or time assignment and, therefore, offers several advantages over cellular mobile communication systems using FDMA and TDMA. CDMA is an advanced digital cellular technology that can offer higher capacity, far superior speech quality, and robust performance in the presence of multipath and interference. This chapter describes the concept of spread spectrum modulation, features of IS-95 CDMA systems for the transmission of multi-rate signals, and numerous techniques and features. The second-generation IS-95 CDMA digital cellular standard is the foundation for third-generation Cdma2000 cellular technoloogy.

12.1 THE CONCEPT OF SPREAD SPECTRUM

In Spread Spectrum (SS) systems, the complete bandwidth is available to each mobile subscriber. The bandwidth is many times larger than the bandwidth required to transmit baseband information. The primary advantage is its ability to tolerate a considerable amount of signal interference, and simplified frequency assignment, and flexibility in system design and deployment. There are three general approaches to implement SS systems:

(a) Direct Sequence Spread Spectrum (DSSS) A carrier signal is modulated by a digital code in which the code bit rate is much larger than the information signal bit rate. These systems are called PseudoNoise (PN) systems. Figure 12.1 depicts DSSS approach.

DSSS-based systems experience continuous but lower-level random errors because errors are dispersed randomly over time. DSSS spreads the information in both the frequency and time domains, thus providing frequency as well as time diversity. This minimises the effects of fading and interference.

(b) Frequency-Hopping Spread Spectrum (FHSS) The carrier frequency is shifted in a pattern generated by a code sequence in discrete increments. The signal frequency remains constant

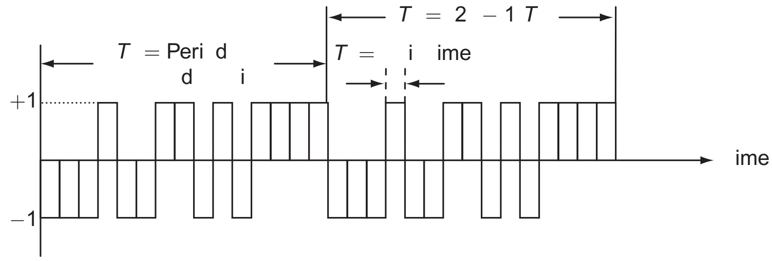

Fig. 12.1 *Direct Sequence Spread Spectrum (DSSS) approach*

for specified time duration, referred to as a time chip, T_c. The FHSS system can be either a fast-hop or a slow-hop system. In a fast-hop system, the frequency hopping occurs at a rate that is faster than the information bit rate. In a slow-hop system, the hop rate is slower than the information bit rate.

FHSS-based radio systems experience occasional strong bursty errors distributed in data blocks. Bursty errors are attributable to frequency interference or fading, which are both frequency as well as time dependent.

Figure 12.2 depicts a general FHSS approach.

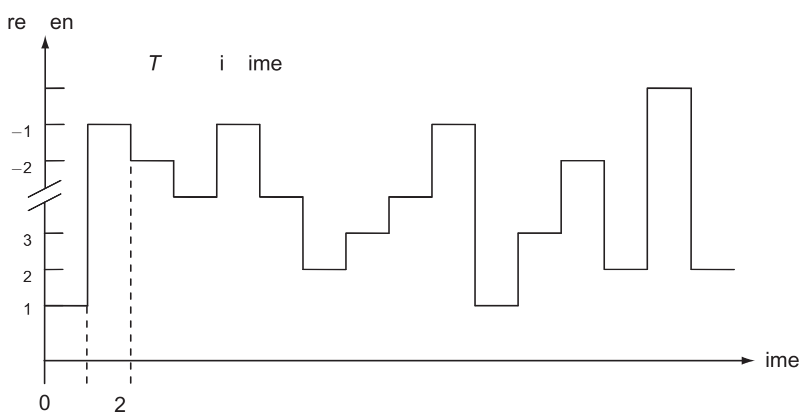

Fig. 12.2 *Frequency Hopping Spread Spectrum (FHSS) approach*

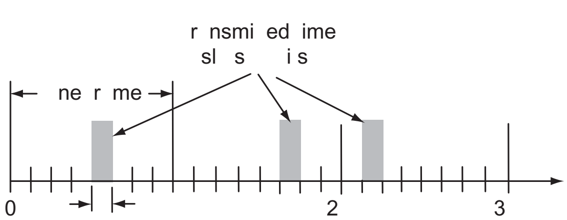

Fig. 12.3 *Time Hopping Spread Spectrum (THSS) approach*

(c) Time Hopping Spread Spectrum (THSS) The transmission time is divided into time intervals called frames, the time frame being T_f. Each frame is divided into M number of uniform time slots, each having duration, $t = T_f/M$. During each frame, one and only one time slot is modulated with an information signal having pre-determined k number of bits. All of the information bits accumulated in previous frames are transmitted. Figure 12.3 depicts the general THSS approach.

12.1.1 General Model of SS Communication System

Figure 12.4 illustrates the functional block diagram of a general model of the spread spectrum communication system. Input data is fed into a data encoder that produces an analog signal with a relatively narrow bandwidth around the centre frequency. The encoded signal is modulated using a sequence of digits (or chips) known as a *spreading code* or *spreading sequence*, generated by a pseudonoise code generator. The effect of spread spectrum modulation is to increase the bandwidth (spread the spectrum) of the signal significantly. On the receiving end, the same de-spreading code sequence is used to demodulate the spread spectrum signal. Finally, the demodulated signal is processed by a data decoder to recover the data output.

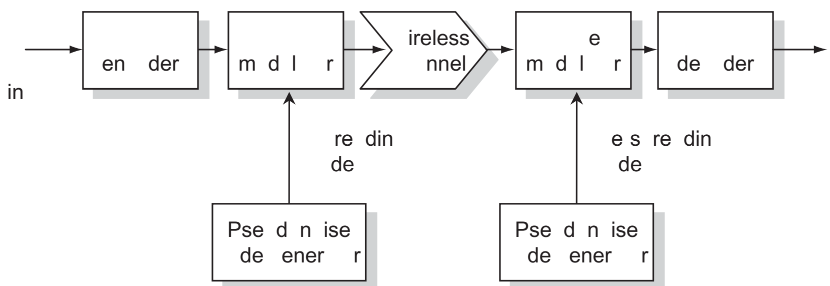

Fig. 12.4 *General model of spread spectrum digital communication system*

The salient features of spread spectrum digital communication system are the following:

(a) The use of orthogonal pseudonoise codes in spread spectrum communications makes wide bands and looks like noise. This results into immunity from various kinds of noise and multipath distortion, or simply immunity to jamming. SS signals are difficult to detect on narrow band receivers because the signal's energy is spread over a bandwidth of may be 100 times the information bandwidth. It is this very characteristic that makes spread spectrum signals possess the quality of low probability of intercept.

(b) The spread of energy over a wide band makes spread spectrum signals less likely to interfere with other narrowband radio communications. The correlator spreads out a narrowband interferer over the receiver's total detection bandwidth. This means that narrowband communications cause little or no interference to spread spectrum signals. Since the SNR at the correlators' input determines whether there will be interference or not, all spread spectrum systems have a threshold level of interference below which useful communication ceases. This threshold level is related to the spread spectrum processing gain. Processing gain is defined as the ratio of the transmitted RF bandwidth to the baseband information bandwidth.

(c) A typical commercial direct sequence spread spectrum communication radio might have a processing gain of 11–16 dB, depending on the data rate. It can tolerate total jammer power levels which may be 0–5 dB stronger than the desired signal. This means that SS system can work at negative SNR in the RF bandwidth. In fact, the SS system functions at positive SNR on the baseband data because of the processing gain provided by the receiver's correlator.

(d) Spread spectrum systems can be used for encrypting the signals. Only a receiver having the knowledge of spreading code can recover the encoded information.

(e) Spread spectrum signals are difficult to exploit or spoof. Signal exploitation is the ability of an undesired interceptor to use information. Spoofing is the act of maliciously sending misleading messages to the network. Spread spectrum signals are naturally more secure than narrowband radio communications.

(f) Several users can independently use the same higher bandwidth with very little interference. This property is used in CDMA cellular communication applications.

12.1.2 Direct Sequence Spread Spectrum

A Direct Sequence Spread Spectrum (DSSS) system spreads the baseband information data by directly multiplying it with a PseudoNoise (PN) sequence that is produced by a PN code generator. With DSSS, each bit in the baseband signal is represented by multiple bits in the transmitted signal, using a spreading code. A single pulse of the PN waveform is called a chip because it has extremely small time duration. The spreading code spreads the information signal across a wider frequency band in direct proportion to the number of chips used. Therefore, a 10-chip spreading code spreads the information signal across a frequency band that is 10 times greater than a 1-chip spreading code. Generally, the information data stream is combined with the spreading code bit stream using an Exclusive-OR (XOR) logical operation.

> **Facts to Know!**
>
> To solve the spreading task and to occupy the transmission band equally, the power spectrum of a single sequence should be like additive white Gaussian noise (AWGN) signal.

A functional block diagram of a direct-sequence SS transmitter system with binary phase modulation technique is shown in Fig. 12.5.

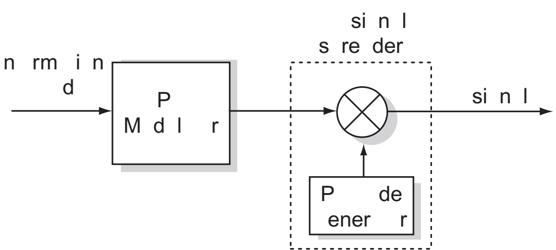

Fig. 12.5 *Block diagram of a DSSS transmitter system*

The information binary data is modulated with Binary Phase Shift Keying (BPSK). Each BPSK modulated information bit is spread into N chips by a DS spreader according to a random data pattern generated locally by a PseudoNoise (PN) code generator. The PN code has much higher data rate than the information data rate. The information data is simply logically modulo-2 added (an EX-OR operation) with the PN code. The bandwidth of any digital system is inversely proportional to the duration of the transmitted pulse. Because the transmitted DSSS chips are N times narrower than information data bits, the bandwidth of the transmitted DSSS signal is N times larger than a system without spreading.

At the DSSS receiver, the received chips are first processed through a DS de-spreader that de-spreads the signal. The de-spreader correlates the received signal with the same PN code sequence as used at the SS transmitted end. A functional block diagram of a direct-sequence SS receiver system with BPSK modulation technique is shown in Fig. 12.6.

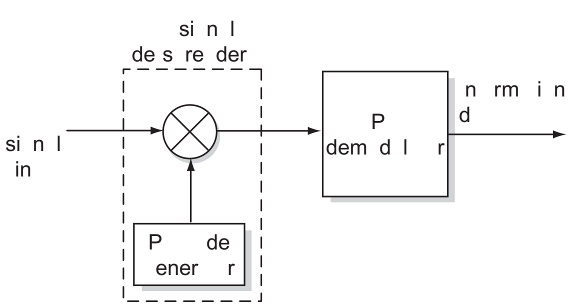

Fig. 12.6 *Block diagram of a DSSS receiver system*

A DSSS receiver uses a locally generated identical PN code generator and a DS de-spreader (also called receiver correlator) to separate the desired coded information from all possible signals. A DSSS correlator is a special matched filter that responds only to signals that are encoded with a PN code that matches its own PN code. It can be configured to different PN codes simply by changing its local PN code. This correlator does not respond to man-made or natural noise and interference. The peak of the autocorrelation function is used to detect the transmitted bit. Autocorrelation corresponds to correlating the bit pulse with itself. This involves multiplying the bit pulse with a delayed version of itself and integrating the product over the pulse duration. The DS de-spread signal is then demodulated with the BPSK demodulator.

There is an alternative method of the most widely used direct sequence implementation of a DSSS signal. Figure 12.7 shows a functional block schematic of a DSSS transmitter system with binary phase modulation.

The information bits or binary coded symbols are added in modulo-2 summer to the chips before being phase modulated.

A coherent or differentially coherent PSK demodulator is used in the receiver. Figure 12.8 shows a functional block schematic of a DSSS receiver system with binary phase modulation.

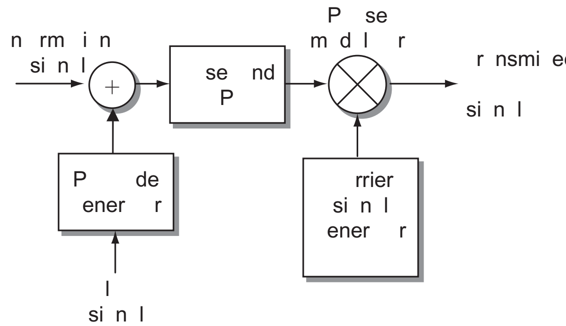

Fig. 12.7 *Block diagram of a DSSS transmitter system with binary phase modulation*

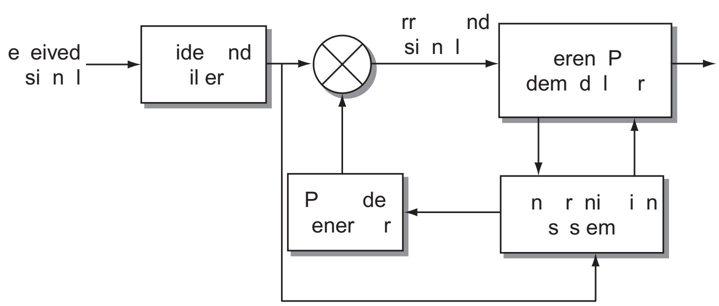

Fig. 12.8 *Block diagram of a DSSS receiver system with BPSK demodulation*

12.1.3 Frequency-Hopping Spread Spectrum

The Frequency-Hopping Spread Spectrum (FHSS) technique is a relatively simple concept to protect the transmitted signal from interference and jamming. With FHSS, the information data signal is transmitted over a large random sequence of radio frequencies, hopping from one frequency to another frequency at pre-determined fixed interval of time. The FHSS transmitter shifts the centre frequency of the transmitted signal in a pseudorandom manner by a hopping sequence generated by PN code generator. The sequence of frequency hops is known only to the intended transmitter and the corresponding receiver. A receiver, hopping between frequencies in synchronisation with the transmitter, decodes the information data. Other communication receivers or would-be eavesdroppers could decode the unintelligible blips only. If the FHSS signal is jammed on one frequency, then few bits transmitted at that particular frequency are corrupted, while all other bits are received satisfactorily at other hopping frequencies.

If the centre frequency of transmission is randomly hopped among 100 different frequencies generated by a frequency synthesiser in the same sequence as determined by the PN code generator then the required transmission bandwidth is 100 times more than the original information data bandwidth. So this technique is also referred to as a spread spectrum technique because the spectrum is spread over a band

Facts to Know!

The major functions of the PseudoNoise (PN) sequences used in wireless digital communications spread-spectrum code-division multiple access systems include spreading the bandwidth of the modulated signal to the larger transmission bandwidth, and distinguishing between the signals of different users utilising the same transmission bandwidth in a multiple-access scheme.

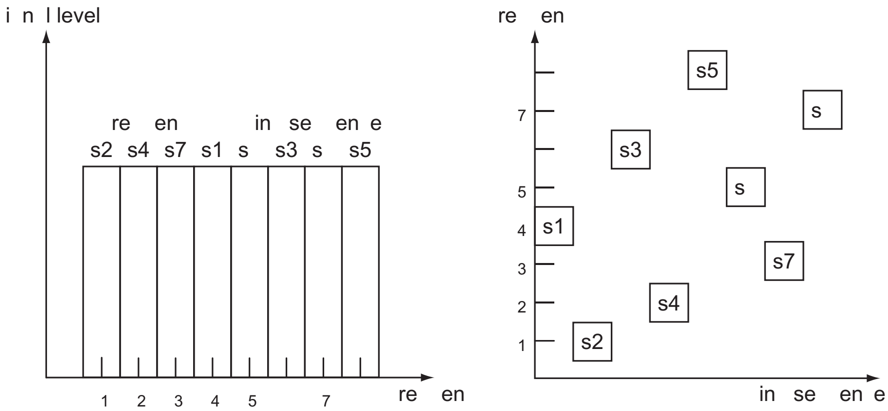

Fig. 12.9 *Basic concept of frequency-hopping*

that is many times greater than the conventional non-FHSS transceiver. FHSS is primarily applied to binary data for digital transmissions. Figure 12.9 shows the basic concept of a frequency-hopping process.

A number of different frequency channels are allocated for the FH signal. The spacing between frequencies usually corresponds to the bandwidth of the information signal. Let there be 2^k number of different carrier frequencies which form 2^k number of distinct channels, where k is the number of codes in PN code generator. The FH transmitter operates in one channel at a time for a fixed interval of time, during which some bits of information bits or a fraction of a bit are transmitted. The sequence of frequency channels used is determined by the PN code. Both transmitter and receiver use the same PN code in perfect synchronisation with a pre-determined sequence of channels. The gain in signal-to-noise ratio, or processing gain G_p, is 2^k. The jammer must jam all 2^k frequencies in a frequency-hopping system.

EXAMPLE 12.1 Number of PN bits in an FHSS system

An FHSS system employs a total bandwidth of 400 MHz and an individual channel bandwidth of 100 Hz. What is the minimum number of PN bits required for each frequency hop?

Solution

Step 1. *To determine processing gain, G_p*

Total bandwidth, $B_t = 400$ MHz (given)

Channel bandwidth, $B_c = 100$ Hz (given)

Processing gain, $G_p = B_t/B_c$

$= 400 \text{ MHz}/100 \text{ Hz} = 4 \times 10^6$

Step 2. *To compute the minimum number of PN bits, k*

In an FHSS system, we know that

Processing Gain, $G_p = 2^k$, where k is the number of PN bits

Therefore, $2^k = 4 \times 10^6 \approx 2^{22}$

Hence, the minimum number of PN bits, $k = 22$ bits

A typical block diagram for an FH transmitter system is shown in Fig. 12.10. The information data is modulated with an FSK/BPSK modulator. The centre frequency is changed rapidly in an FH spreader according to a random orthogonal hopping pattern generated by a known PN code generator. A code generator serves as an index into a set of channel frequencies, referred as the spreading code. Each k bit of the PN code generator specifies one of the 2^k carrier frequencies. At each successive k PN bits, a new carrier frequency is selected. The output is passed through a bandpass filter and resulting FHSS signal is finally transmitted.

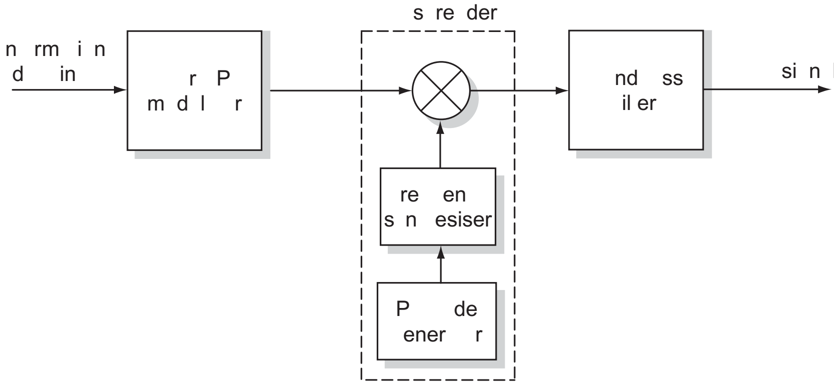

Fig. 12.10 | *Block diagram of an FHSS transmitter system*

In a fast frequency-hopping system, the frequency hops occur much more rapidly and in each hop a very short information data packet is transmitted. In fast frequency-hopping systems, the same bit is transferred using several frequencies. It is used in military applications. In a slow frequency-hopping system, long data packets are transmitted in each hop over the wireless channel.

EXAMPLE 12.2 | Slow frequency-hopping concept

In a slow frequency-hopping system, where long data packets are transmitted over the wireless channel at each hop, the sequence of frequencies programmed in the PN code generator is f_3, f_5, f_6, f_1, f_4, f_8, f_2, and f_7 before returning to the first frequency, f_3. Draw a suitable diagram to illustrate the concept of frequency hopping.

Solution

Figure 12.11 shows the given hopping pattern and associated frequencies for a frequency-hopping system. Each packet is transmitted using a different frequency.

A functional block diagram for an FH receiver system is shown in Fig. 12.12.

On reception, the FH spread spectrum signal is demodulated using the same sequence of PN-coded frequencies using a frequency synthesiser. This signal is then demodulated by an FSK or BPSK demodulator to produce the output data.

The performance of the FHSS systems in non-interfering wireless environments remains exactly the same as the performance of the non-FHSS systems. In an FHSS system, the interference or frequency selective fading corrupts only a fraction of the transmitted information, and transmission in the rest of the centre frequencies remains unaffected because the carrier frequency is constantly changing. This feature of the FHSS is used in the design of wireless networks to provide a reliable transmission in a frequency selective fading channel or in the presence of interfering signals.

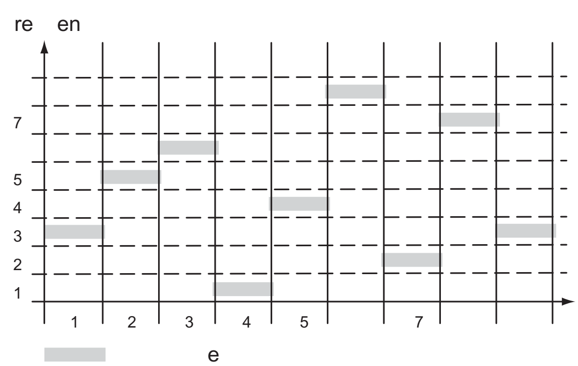

Fig. 12.11 | *Illustration of a slow FHSS system*

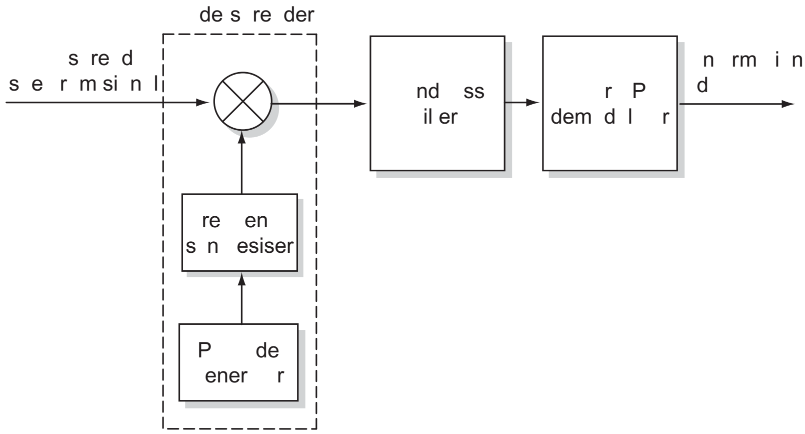

Fig. 12.12 *Frequency-hopping spread spectrum receiver system*

> **Facts to Know!**
>
> In order to generate 2^m frequencies in the transmitted hopped system, a digital frequency synthesiser is controlled by serial or parallel words, each containing *m* binary digits.

An FHSS system is less complex in design because the sampling rate is not very high, saving in the implementation costs and power consumption of the mobile subscriber units. On the other hand, a DSSS system needs much higher sampling rates and consequently a more complex hardware implementation. The FHSS is a narrowband system hopping over a number of frequencies in a wide spectrum, whereas the transmission bandwidth of the DSSS is always wide. A DSSS system is also anti-interference and resistant to frequency selective fading. Moreover, DSSS systems provide robust signal with better coverage area than FHSS. Therefore, DSSS systems are preferred in digital cellular systems such as IS-95 and Cdma2000.

12.2 ARCHITECTURE OF CDMA SYSTEM

> **Facts to Know!**
>
> CDMA systems are an extension of DSSS and FHSS systems, to provide multiple-access communications capabilities. In CDMA, each user is assigned with an individual unique PN code. Provided these codes are uncorrelated with each other, the number of independent users (same as number of distinct codes) can transmit at the same time and in the same radio bandwidth within the same cell.

The user data as well as the control channel and signaling information are transmitted at the same frequency at the same time in the CDMA system. All the CDMA systems employ direct sequence spread spectrum with processing gain, RAKE receiver diversity gain, a fast power-control mechanism to minimise interference, efficient error correcting coding techniques, variable-rate voice coders that provide a gain from pauses in natural voice conversation, and soft hand-off. This is made possible by reducing the required signal (bit energy)-to-noise ratio (E_b/N_0) for proper operation.

12.2.1 CDMA in a DSSS Environment

The DSSS can be employed for code division multiple access. In a multiuser DS-CDMA environment, different codes are assigned to different mobile subscribers. In other words, each mobile subscriber has its own unique key code which is used to spread and de-spread only that user's information. The codes assigned to other subscribers are selected so that during the despreading process at the receiver, they produce very small signal levels (like noise). In this manner, each subscriber is a source of noise for the detection of other subscribers' signals. As the number of subscribers increases, the multiuser interference increases for all of the subscribers. This phenomenon continues up to a point that mutual interference among all mobile subscribers stops the proper operation for all of them.

CDMA is a multiple-access strategy for wireless communications based on DSSS. The performance of a single-user DSSS signal is identical to that of the corresponding BPSK and QPSK signal in an AWGN environment. The spreading of the information signal provided by the code simply changes the spectrum of the signal used to transmit the information. The spreading codes are selected to be perfectly orthogonal with one another so as to achieve a single-user performance in the multiuser case. Therefore, the design of perfectly orthogonal codes for all users is the most critical system parameter.

The DSSS form of CDMA is generated by combining each of the baseband signals to be multiplexed with a PN sequence at a much higher data rate. Each of the signals to be multiplexed should use a different PN sequence. If the various sequences are mathematically orthogonal, the individual baseband signals can be recovered exactly without any mutual interference. However, the number of possible orthogonal sequences of codes is limited and depends on the length of the sequence. If the PN sequences are not orthogonal, CDMA is still possible. But there will be some mutual interference between the signals which may result in an increased noise level for all signals. As the number of non-orthogonal signals increases, the signal-to-noise ratio becomes too low and the bit-error rate too high rendering the operation of the system unacceptable.

Therefore, using orthogonal PN sequences for CDMA is highly desirable. A class of PN sequence called a Walsh code is used. Walsh codes are an important set of orthogonal codes. In CDMA, each subscriber is assigned a unique orthogonal code. Since the resultant waveforms are orthogonal, subscribers with different codes do not interfere with each other. Since the waveforms are orthogonal only if they are aligned in time, CDMA requires perfect synchronisation among all the subscribers. The cell-site uses 64 orthogonal Walsh codes; each repeats after 64 bits. This allows for 64 independent logical channels per RF channel. The Walsh code 0 is used for the pilot channel to keep mobile receivers phase-aligned with the cell-site. This is a requirement for coherent demodulation, which is the only way to avoid interference among channels using the same carrier frequency.

Facts to Know!

Theoretically, a zero cross-correlation is maintained by every set of orthogonal spreading signals such as Walsh functions. However, in practical wireless systems, the PN sequence has to be generated coherently on both the transmitter and receiver sides.

In addition to the Walsh codes, a short code for synchronising, and a long code for encryption of both voice and control-system data are also used at the CDMA cell-sites. These codes are not used for spreading of the information data.

Figure 12.13 depicts CDMA configuration as a multiplexing technique used with direct sequence spread spectrum. Let there be n number of mobile users, each transmitting the DSSS signal using its unique orthogonal PN code sequence. For each user, the information data stream to be transmitted, $d_n(t)$, is BPSK modulated to produce a signal with a bandwidth of W_s and then multiplied by the spreading code for that user, $c_i(t)$. All these DSSS signals, plus wireless channel noise, are received at the receiver antenna of the desired user.

Suppose that the CDMA receiver is attempting to recover the data of User 1. The received signal is demodulated and then multiplied by the spreading code of User code 1. This results in narrowing the bandwidth of that portion of the incoming signal corresponding to User 1 to the original bandwidth of the unspread signal, which is proportional to the information data rate. Because the rest of the received signal is orthogonal to the spreading code of User 1, that remaining signal still has the wider bandwidth W_s. Thus the undesired signal

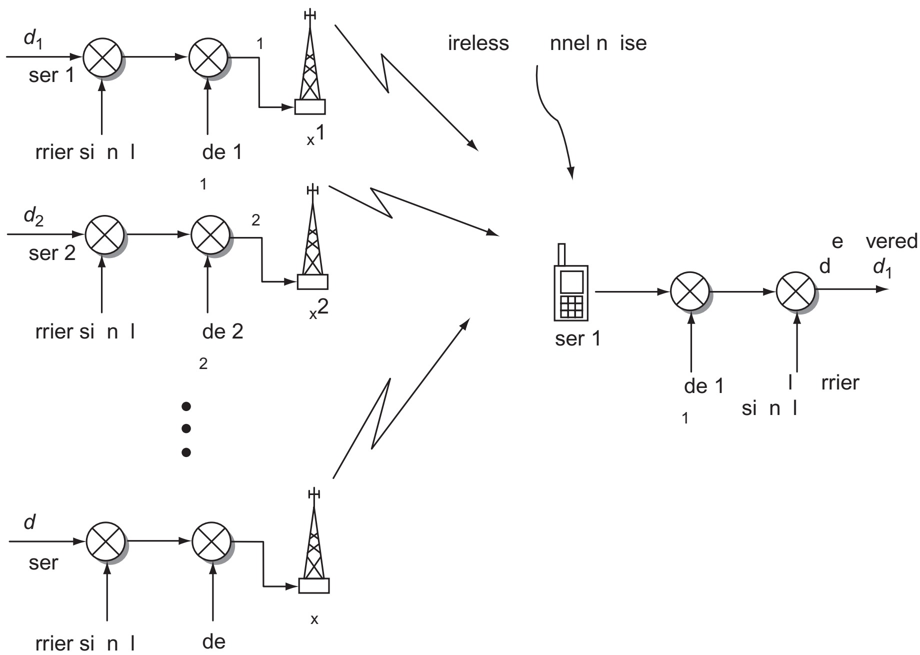

Fig. 12.13 *CDMA in a DSSS environment*

energy remains spread over a large bandwidth and the desired signal is concentrated in a narrow bandwidth. The bandpass filter used at the demodulator can therefore recover the desired signal. The principle benefits of spread spectrum transmission are as follows:

(a) The effect of multipath fading as well as interference can be reduced by a factor, known as the processing gain, which is the ratio of the spread bandwidth to the original bandwidth.
(b) The spectral density of the DSSS transmitted signal is reduced by a factor equal to the processing gain.
(c) Under ideal conditions, there is no difference in bit-error rate performance between spread and nonspread forms of BPSK or QPSK digital modulation schemes.
(d) Through proper receiver design, multipath can be used to advantage to improve receiver performance by capturing the energy in paths having different transmission delays.
(e) By using RAKE receiver concept, a spread-spectrum receiver can obtain an important advantage in diversity in fading channels.
(f) The choice of spreading codes is critical to reducing multipath self-interference (spreading-code autocorrelation) and multiple-access interference (spreading-code cross-correlations).
(g) Spread-spectrum signals can be overlaid onto frequency bands where other systems are already operating, with minimal performance impact to both systems.
(h) Spread spectrum is a wideband signal that has a superior performance over conventional communication systems on frequency selective fading multipath channel.
(i) Spread spectrum provides a robust and reliable transmission in urban and indoor environments where wireless transmission suffers from heavy multipath conditions.
(j) Cellular systems designed with CDMA spread spectrum technology offer greater system capacity and operational flexibility.

12.2.2 CDMA in a Cellular Environment

CDMA cellular systems are implemented based on the spread spectrum technology. In its most simplified form, a spread spectrum transmitter spreads the signal power over a spectrum which is as large as N times wider than the spectrum of the information signal. In other words, an information bandwidth of R_b occupies a transmission bandwidth of B_c, where:

$$B_c = N \times R_b$$

Or, $$N = B_c/R_b \tag{12.1}$$

Facts to Know!

The concept of self-interference can be well understood by an example. Let 10 mobile users be transmitting at the same time. The base station receiver will receive 10 spectrally overlapped as well as time-overlapped signals. If the received power of all 10 signals is the same, then any one of the desired signals will be interfered by nine other equal-power CDMA signals. Thus the desired C/I ratio will be 1/9, or –9.54 dB. This negative C/I is caused by self-interference which implies that interference caused by nine other DSSS signals that simultaneously occupy the user bandwidth as the tenth desired signal.

The spread spectrum receiver processes the received signal with a processing gain G_p equal to N. This means that during the processing at the receiver, the power of the received signal having the code of that particular receiver will be increased N times beyond the value before processing. In fact, the system processing gain (G_p) quantifies the degree of interference rejection. It is simply the ratio of RF bandwidth to the information bit rate and is given as

$${}_p = \quad /R \tag{12.2}$$

Typical processing gains for SS systems lie between 20 and 60 dB. With an SS system, the noise level is determined both by the thermal noise and by interference. For a given user, the interference is processed as noise alone. The input and output Signal-to-Noise Ratios (SNRs) are related as

$$(S/N)_o = G_p\,(S/N)_i \tag{12.3}$$

But $(S/N)_i$ is related to the (E_b/N_o), where E_b is the energy per bit and N_o is the noise power spectral density including both the thermal noise and interference. With SS systems, interference is transformed into noise.

$$(S/N)_i = (E_b \times R_b)/(N_o \times B_c) = (E_b/N_o)/(1/G_p) \tag{12.4}$$

Or, $$E_b/N_o = G_p \times (S/N)_i = (S/N)_o \tag{12.5}$$

The minimum E_b/N_o value required for proper system operation can be defined if the performance of the coding methods used on the signals, bit error rate, and the tolerance of the digitised voice signals are known. The number of CDMA channels in the network depends on the level of total interference that can be tolerated in the system. Thus, the CDMA system is interference limited. A well-designed system will have a required bit-error probability with a higher-level of interference. The Forward Error Correction (FEC) coding technique improves tolerance for interference and increase overall CDMA system capacity. It is assumed that at the cell-site, the received signal level of each mobile user is the same and that the interference observed by each receiver is modeled as Gaussian noise. Each modulation method has a relationship that defines the bit-error rate as a function of the E_b/N_o ratio. The best performance of the system can be obtained by maintaining the minimum E_b/N_o required for operation. The relationship between the number of mobile users, M, the processing gain, G_p, and the E_b/N_o ratio is therefore given as

$$= \quad {}_p \; (\quad / \quad) \tag{12.6}$$

For a given bit-error probability, the actual E_b/N_o ratio depends on the radio system design and error-correction coding technique used. The measured value of E_b/N_o may be closer, if not equal, to the theoretical value.

EXAMPLE 12.3 Processing gain of a DSSS system

A DSSS system has a 10-Mcps code rate and a 4.8-kbps information data rate. If the spreading code generation rate is increased to 50 Mcps, how much improvement in the processing gain of this DSSS system will be achieved? Is there any advantage in increasing the spreading code generation rate with a 4.8-kbps information data rate? Comment on the results obtained.

Solution

Step 1. *To find Processing gain, G_p at 10 Mcps code rate*

Code rate, $R_c = 10$ Mcps (given)

In the DSSS system, the RF bandwidth is same as spreading code rate.

Therefore, RF bandwidth, $B_c = 10$ Mcps or 10×10^6 cps

Information data rate, $R_b = 4.8$ kbps or 4.8×10^3 bps (given)

Processing gain, $G_p = B_c/R_b = (10 \times 10^6)/4.8 \times 10^3\ )$

Therefore, Processing gain, $G_p = 2.1 \times 10^3$

Expressing it in dB, $G_p = 10 \log (2.1 \times 10^3) = 33.1$ dB

Step 2. *To find processing gain, G_p at 50 Mcps code rate*

Code rate, R_c or RF bandwidth, $B_c = 50$ Mcps or 50×10^6 cps (given)

Processing gain, $G_p = B_c/R_b = (50 \times 10^6)/4.8 \times 10^3\ )$

Therefore, Processing gain, $G_p = 1.04 \times 10^4$

Expressing it in dB, $G_p = 10 \log (1.04 \times 10^4) = 40.2$ dB

Step 3. *Improvement in processing gain*

Thus, increase in processing gain = 40.2 – 33.1 = 7.1 dB

Comment on the results The improvement in processing gain is only 7.1 dB after enhancing the spreading code rate by 5 times (50 Mcps/10 Mcps). The circuit complexity needed to get five times the spreading code rate is too high for a marginal improvement of 7.1 dB in processing gain. So there is not much advantage in this case.

Consider the situation of a single cell operating in a CDMA cellular network. In its most simplified form, a spread spectrum transmitter spreads the signal power over a spectrum N times wider than the spectrum of the information signal. The spread spectrum receiver processes the received signal with a processing gain of N. This means that during the processing at the receiver, the power of the received signal having the code of that particular receiver will be increased N times beyond the value before processing.

Let there be M simultaneous users on the reverse channel of a CDMA network. It is assumed that there is an ideal power control employed on the reverse channel so that the received power of signals from all mobile subscriber units has the same value P_r. Then, the received power from the target mobile user after processing at the cell-site receiver is $N \times P_r$, and the received interference from $(M - 1)$ other mobile users is $(M - 1) \times P_r$. Assuming that a cellular system is interference limited and the background noise is dominated by the interference noise from other mobile users, the received signal-to-interference ratio, S_r for the target mobile receiver will be

$$S_r \approx (N \times P)/\ (M-1) \times P\ \approx N/(M\ \ 1) \tag{12.7}$$

All mobile users always have a requirement for the acceptable error rate of the received data stream. For a given modulation scheme and coding technique used in the system, that error rate requirement will be supported by a minimum S_r requirement that can be used to determine the number of simultaneous users or capacity of the system.

Using $N = B_c/R_b, \quad S_r = B_c/R_b\,(M \quad 1)$

Or, $M = \quad B_c/R_b \quad . \quad 1/S_r \quad + 1$

Or, $M \approx \quad B_c/R_b \quad \times \quad 1/S_r$ (12.8)

The number of simultaneous users or capacity of a single-cell CDMA system is inversely proportional to acceptable signal-to-interference ratio in the system.

EXAMPLE 12.4 Capacity of one carrier in a single-cell CDMA system

Given that the IS-95 CDMA digital cellular systems require 3 dB < S_r < 9 dB which employs QPSK modulation scheme and convolutional coding technique. The bandwidth of the channel is 1.25 MHz, and the transmission data rate is R_b = 9600 bps. Determine the capacity of a single IS-95 cell.

Solution

Channel bandwidth, B_c = 1.25 MHz or 1250 kHz (given)
The transmission data rate, R_b = 9600 bps or 9.6 kbps (given)

Step 1. *To determine maximum number of simultaneous users, M_{max}*

The minimum acceptable, S_r (min) = 3 dB (given)
Converting S_r (min) = 3 dB in S_r (min) ratio by using the expression
S_r (min) dB = 10 log S_r (min) ratio
Or, 3 dB = 10 log S_r (min) ratio
Or, S_r (min) ratio = $10^{3/10} = 10^{0.3} = 2$
The maximum number of simultaneous users can be determined by using the expression

$M_{max.} \approx \quad B_c/R \quad \times \quad 1/S_r\,(\text{min})$

Or, $M_{max.} \approx$ 1250 kHz/9.6 kbps $\times$ 1/2

Hence, $M_{max.} \approx$ 65 users

Step 2. *To determine minimum number of simultaneous users, M_{min}*

The maximum acceptable, S_r (max) = 9 dB (given)
Converting S_r (max) = 9 dB in S_r (max) ratio by using the expression
S_r (max) dB = 10 log S_r (max) ratio
Or, 9 dB = 10 log S_r (max) ratio
Or, S_r (max) ratio = $10^{9/10} = 10^{0.9} = 7.94$
The minimum number of simultaneous users can be determined by using the expression

$M_{min.} \approx \quad B_c/R_b \quad \times \quad 1/S_r\,(\text{max})$

Or, $M_{min.} \approx$ 1250 kHz/9.6 kbps $\times$ 1/7.94

Hence, $M_{min.} \approx$ 16 users

Hence, a single cell IS-95 CDMA digital cellular systems system can support from 16 users to 65 users.

With CDMA systems, the same frequency channel can be reused in the adjacent cell, as long as Multiple-Access Interference (MAI) is kept below a given threshold level necessary to meet the signal quality requirement. MAI comprises of two types of interference: intracell interference and intercell interference.

(a) *Intracell interference* is defined as the interference caused by other users operating within the same cell.
(b) *Intercell interference* is defined as the interference caused at the mobile user in a cell due to reuse of the same CDMA channel in the neighbouring cells.

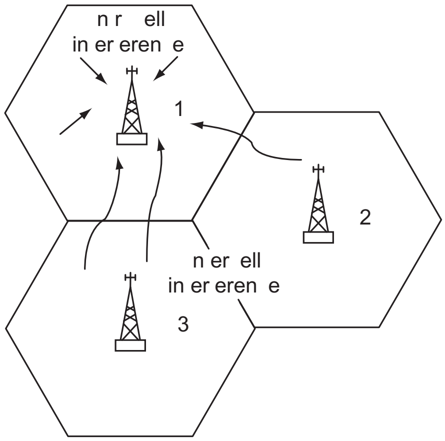

Fig. 12.14 Multiple-access interference

Figure 12.14 depicts intracell and intercell interference in hexagonal cells.

Intracell interference and intercell interference affect the multiple access performance in a cellular environment for the uplink from a mobile user to a cell-site located at the centre of the cell. If there is perfect power control on the reverse channel such that the signals from all users are received at equal power at the cell-site then the intracell interference is given by

$$I_{intracell} = (M - 1)/Q \times E_b \tag{12.9}$$

where M is the number of simultaneous users, Q is the number of chips per time period T, and E_b is the common received power level.

Since M 1, $$I_{intracell} \approx (M/Q) \times E_b \tag{12.10}$$

where M/Q is termed as channel loading or capacity.

Most of intercell interference occurs from the first and second tiers of the surrounding cells of the serving cell. The interference from more distant cells suffers more propagation attenuation, and hence can be ignored. The signals causing intercell interference are received at different power levels, because they are power controlled relative to other cell-sites. As a consequence, intercell interference depends on the propagation losses from a mobile user to two different cell-sites. In general, the relative power from mobile users in other cells will be attenuated relative to the power from the intracell mobile users due to larger distance. Let σ be the relative intercell interference factor, and is defined as the ratio of intercell interference to intracell interference, that is,

$$\sigma = I_{intercellular}/I_{intracell} \tag{12.11}$$

It is assumed here that the traffic loading in all cells is the same. The value of the intercell interference factor σ ranges from 0.5 to 20, depending upon the number of parameters such as the following:

The Propagation-Loss Exponent, γ The larger the propagation-loss exponent, the more attenuation adjacent-cell interference signals will suffer. Urban regions will tend to decrease intercell interference more quickly than suburban and rural regions. This is, in fact, advantageous as the urban regions tend to have the highest density of users.

The Variations in Signal Strength Due to Shadowing Shadowing loss between the mobile user and the serving cell-site will be neutralised by accurate power control. Shadowing of the same mobile signal relative to other cell-sites will not be neutralised and will cause a significant and variable effect on intercell interference. In fact, the diversity effect permits a significant reduction in the system margin allowed for shadow loss; the resulting decrease in transmitted power means that intercell interference will be considerably less.

The Handover Technique Between Cells Since the same frequencies are reused in every cell, it is possible to send and receive the same call from multiple cell-sites. This property becomes particularly important at the boundary of the cells, where there is possibility of call handover between cell-sites as a mobile user moves from one cell to the other cell. A soft handover can occur in CDMA, and the mobile user maintains communications with both cell-sites until one of them becomes significantly stronger than the other. A soft handover prevents the ping-pong behavior, and the dual cell-site capability is a form of diversity that can increase capacity in a heavily loaded system and also increase coverage in a lightly loaded system, in addition to reduction in intercell interference.

The total interference or MAI is the combination of intracell interference and intercell interference, that is

$$I_{MAI} = I_{intracell} + I_{intercellular} \tag{12.12}$$

Or, $$I_{MAI} = I_{intracell} + \sigma\, I_{intracell}$$

Or, $$I_{MAI} = (M/Q) \times E_b + \sigma\,(M/Q) \times E_b$$

Or, $$I_{MAI} = (1 + \sigma\,)\;\;(M/Q) \times E_b \tag{12.13}$$

Thus, the MAI is directly proportional to the channel loading or capacity, M/Q. Ultimately, the MAI on a cellular CDMA is more significant at the individual receiver. The signal-to-interference-plus-noise ratio (SINR) at the individual receiver is given by

$$\mathrm{SINR} = E_b/(N_0 + I_{MAI}) = E_b/\; I_{MAI}\,(1 + N_0/I_{MAI}) \tag{12.14}$$

Cellular CDMA systems are often interference limited; that is, the operating conditions are such that I_{MAI} N_0, typically 6 to 10 dB higher. The I_{MAI}/N_0 ratio depends upon the cell size. With large cells and battery-operated mobile phones, most of the transmit power is used to achieve the desired range. Thus, large cells tend to be noise limited. Smaller cells tend to be interference limited, and the interference level at the receiver is typically greater than the noise level of the receiver.

Substituting Eq. (12.13) into Eq. (12.14), we get

$$\mathrm{SINR} = 1/\;(1 \quad \boldsymbol{\sigma}\,)\;(\quad/\quad)\;(1 \qquad {}_0/ \qquad) \tag{12.15}$$

This expression shows the three system design factors that affect the SINR at the receiver, and limit spectral efficiency. The three factors are the intercell interference σ, the channel loading M/Q, and the operating I_{MAI}/N_0. The intercell interference depends on the environment as well as on the handover technique; channel loading is clearly a design parameter that needs to be maximised in a commercial cellular system; and, the third factor, I_{MAI}/N_0, is related to cell size. There is a trade-off between these three system design parameters. For example, for a constant SINR, moving from a noise-limited system ($I_{MAI}/N_0 = 0$ dB) to an interference-limited system ($I_{MAI}/N_0 = 10$ dB, say) increases the permissible channel loading or capacity. The channel loading must be significantly decreased to support a noise-limited system at the same SINR. Thus, large cells must have lighter load than small cells.

Similarly, for a constant SINR, increasing intercell interference significantly reduces the permissible channel loading. The methods of reducing the required SINR in CDMA systems are use of RAKE receivers and FEC coding. However, soft hand-offs are the key design aspect to reducing margins and keeping intercell interference low. On the other hand, reducing the SINR required by the receiver can significantly improve the permissible channel loading.

EXAMPLE 12.5 | Number of users per cell in an IS-95 system

Suppose an IS-95 CDMA cellular system is interference limited, with a ratio of total intracell-plus-intercell interference to receiver noise of 6 dB. Assuming continuously transmitting users, compute the average number of users allowed per cell if the relative intercell interference factor is 0.55 and the required SINR at the receiver is 8 dB.

Solution

Step 1. *To convert $I_{MAI}/N_0 = 6$ dB in ratio*

Intracell-plus-intercell interference to receiver noise, $I_{MAI}/N_0 = 6$ dB (given)

We know that $I_{MAI}/N_0\,(\mathrm{dB}) = 10 \log\; I_{MAI}/N_0\,(\mathrm{ratio})$

Or, $6\ \mathrm{dB} = 10 \log\; I_{MAI}/N_0\,(\mathrm{ratio})$

Or, $I_{MAI}/N_0\,(\mathrm{ratio}) = \mathrm{antilog}\,(6/10) = 3.98$

Step 2. *To find N_0/I_{MAI} (ratio)*

$$N_0/I_{MAI}\,(\text{ratio}) = 1/3.98 = 0.25$$

Step 3. *To convert SINR = 8 dB in ratio*

The required SINR at the receiver = 8 dB (given)

Expressing it in ratio, we get SINR = antilog (8/10) = 6.3

Step 4. *To compute average number of users, M*

The total spreading factor, $Q = 128$ (standard)

Relative intercellular interference factor, $\sigma = 0.55$ (given)

We know that

$$\text{SINR} = 1/\,(1+\sigma)\,(M/Q)\,(1 + N_0/I_{MAI})$$

The average number of users per cell, $M = Q/\,(1+\sigma)\,(1 + N_0/I_{MAI})\ \text{SINR}$

Therefore, $M = 128/\,(1 + 0.55)\,(1 + 0.25)\ 6.3$

Hence, $M = 10.5$ users per cell

In the practical design of digital cellular systems, there are three other system parameters that affect the number of simultaneous mobile users that can be supported by the system as well as the bandwidth efficiency of the system. These are the interference increase factor, the voice activity factor, and the number of sectors in each cell-site. These parameters are quantified as factors used in the calculation of the number of simultaneous mobile users that the CDMA system can support. The interference increase factor σ accounts for mobile users in other cells in the system. Because all neighbouring cells in a CDMA cellular network operate at the same frequency, they will cause additional interference. This interference is relatively small due to the processing gain of the system and the distances involved; a value of $\sigma = 1.6$ (2 dB) is commonly used in the practical CDMA cellular system.

The expression for SINR should include voice activity factor and cell sectorisation factor. Voice activity refers to the fact that the calling or called mobile user speaks only 40 of the time on an average in a two-way voice communication. If a mobile phone transmits only when the user speaks then it will contribute to the interference just 40 of the total talk time. Therefore, it is expected that a system consisting mainly of voice calls and using voice activity would see an average 40 reduction in interference, and could tolerate proportionately higher cell loadings. If voice activity is taken into consideration, the value of M increases by a factor of approximately 2.5 per cell. The voice activity interference reduction factor G_v is the ratio of the total connection time to the active talkspurt time. On the average, in a two-way conversation, each user talks roughly 50 of the time. The short pauses in the flow of natural speech reduce the activity factor further to about 40 of the connection time in each direction. As a result, the typical value used for voice activity factor, G_v is 2.5 (4 dB).

Cell sectorisation refers to the fact that cell-sites are sometimes installed with directional antennas instead of omnidirectional antennas in order to improve system capacity. Typically, directional antennas cover a sector of 120 , radiating and receiving signals only in that sector in a 3-sector cellular configuration. The use of sectored antennas is an important factor in maximising bandwidth efficiency too. Cell sectorisation using directional antennas reduces the overall interference, increasing the allowable number of simultaneous users by a sectorisation gain factor, which is denoted by G_A. With ideal sectorisation, the users in one sector of a base station antenna do not interfere with the users operating in other sectors, and $G_A = N_{sec}$ where N_{sec} is the number of sectors in the cell. This antenna directionality reduces the interference generated and received by the cell-site and could increase the cell loading by a factor close to three. In practice, antenna patterns of directional antennas cannot be designed to have ideal characteristics, and due to multipath reflections, users in general communicate with more than one sector. Three-sector base station antennas are commonly used in cellular systems, and a typical value of the sectorisation gain factor is taken as $G_A = 2.5$ (4 dB).

Incorporating these three factors, the number of simultaneous users that can be supported in a practical CDMA multicell can be approximated by

$$= \quad /R \quad . \quad 1/S \quad . \qquad /\boldsymbol{\sigma} \tag{12.16}$$

Where $G_V \,.\, G_A/\sigma$ is termed as the performance improvement factor, P_f in an IS-95 digital cellular system due to multi-user interference, voice activation, and cell-sectorisation factors in a cell.

Hence, $$M = B_c/R_b \;.\; 1/S_r \;.\; P_f \tag{12.17}$$

EXAMPLE 12.6 Performance improvement factor in an IS-95 system

Compute the performance improvement factor P_f in an IS-95 digital cellular system, considering typical values of voice activation, cell sectorisation, and multi-user interference in a cell.

Solution

The voice activity interference reduction factor, $G_V = 2.5$ (typical)

The antenna sectorisation gain factor, $G_A = 2.5$ (typical)

The interference increase factor, $\sigma = 1.6$ (typical)

The performance improvement factor, P_f in an IS-95 system is given by

$$P_f = (G_V \,.\, G_A)/\sigma$$

Therefore, $$P_f = 2.5 \times 2.5/1.6 = 3.9$$

Expressing it in dB, we get, (dB) $= 10 \log 3.9 = 5.9$ dB

EXAMPLE 12.7 Capacity of IS-95 in a multi-cell practical system

Using QPSK modulation and convolutional coding, the IS-95 digital cellular systems require 3 dB < S_r < 9 dB. Determine the multicell IS-95 CDMA capacity range if the performance improvement factor due to antenna sectorisation, voice activity, and interference increase parameter is approximately 6 dB.

Solution

Channel bandwidth, $B_c = 1.25$ MHz or 1250 kHz (standard)

Transmission data rate, $R_b = 9600$ bps (standard)

Step 1. *To convert the performance improvement factor in ratio*

Performance improvement factor, $P_f = 6$ dB (given)

Converting $P_f = 6$ dB in P_f(ratio) by using the expression

$$P_f \text{ dB} = 10 \log P_f(\text{ratio})$$

Or, $$6 \text{ dB} = 10 \log P_f(\text{ratio})$$

Or, $$P_f(\text{ratio}) = 10^{6/10} = 10^{0.6} = 4$$

Step 2. *To determine maximum number of simultaneous users, M_{max}*

The minimum acceptable, S_r(min) = 3 dB (given)

Converting S_r(min) = 3 dB in S_r(min) ratio by using the expression

$$S_r(\text{min}) \text{ dB} = 10 \log S_r(\text{min}) \text{ ratio}$$

Or, $$3 \text{ dB} = 10 \log S_r(\text{min}) \text{ ratio}$$

Or, $$S_r(\text{min}) \text{ ratio} = 10^{3/10} = 10^{0.3} = 2$$

The maximum number of simultaneous users can be determined by using the expression for the multicell IS-95 CDMA capacity,

$$M_{max} = B_c/R_b \;\; . \;\; 1/S_r(\text{min}) \;\; . \;\; P_f$$

$$M_{max} = 1250 \text{ kHz}/9.6 \text{ kbps} \times 1/2 \times 4 = 260 \text{ users}$$

Hence, $M_{max.} = 260$ users

Step 3. *To determine minimum number of simultaneous users, M_{min}*

The maximum acceptable, $S_r\,(\text{max}) = 9$ dB (given)

Converting $S_r\,(\text{max}) = 9$ dB in $S_r\,(\text{max})$ ratio by using the expression

$$S_r\,(\text{max}) \text{ dB} = 10 \log \; S_r\,(\text{max}) \text{ ratio}$$

Or, $$9 \text{ dB} = 10 \log \; S_r\,(\text{max}) \text{ ratio}$$

Or, $$S_r\,(\text{max}) \text{ ratio} = 10^{\,9/10} = 10^{\,0.9} = 7.94$$

The minimum number of simultaneous users can be determined by using the expression for the multicell IS-95 CDMA capacity,

$$M_{min} = B_c/R_b \;\; . \;\; 1/S_r(\text{max}) \;\; . \;\; P_f$$

$$M_{min} = 1250 \text{ kHz}/9.6 \text{ kbps} \times 1/7.94 \times 4 = 64 \text{ users}$$

Hence, $M_{min.} = 64$ users

Hence, a single-cell IS-95 CDMA digital cellular systems system can support from 64 users to 260 users or 64 M 260.

12.3 CDMA AIR INTERFACE

The air interface in CDMA systems is not symmetrical on the forward and reverse channels as in TDMA systems, which makes it the most complex of all systems. The means of applying spread spectrum modulation and error-control coding techniques are different on the forward and reverse channels. The CDMA channels are defined in terms of an RF frequency and code sequence. Sixty-four Walsh functions are used to identify the forward channels, whereas 64 long PN codes are used for the identification of the reverse channels.

The IS-95 standard defines a variety of communication channels for both forward-channel and reverse-channel transmission. All of the different channel types use the CDMA multiple access technique. Each carrier of the IS-95 occupies a 1.25-MHz band, whereas carriers of AMPS and IS-136 each occupy 30 kHz of bandwidth. Figure 12.15 shows the frequency spacing for two adjacent CDMA channels.

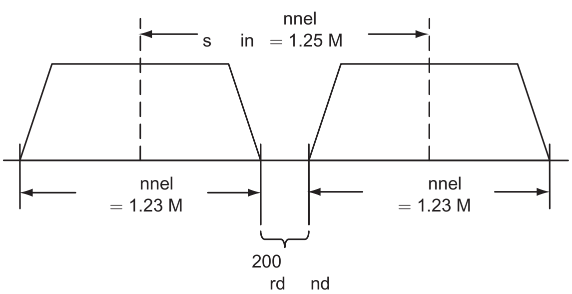

Fig. 12.15 *CDMA channel bandwidth, guard band, and channel spacing*

As the figure shows, each CDMA channel is 1.23 MHz wide with a 1.25-MHz channel spacing between adjacent carrier channels, providing a 200-kHz guard band between two adjacent CDMA channels. Guard bands are necessary to ensure that the CDMA carrier channels do not interfere with one another.

Each RF channel at a cell-site supports up to 64 orthogonal CDMA channels, using direct-sequence spread-spectrum. The CDMA system requires spreading of the spectrum using a PN sequence. In IS-95A CDMA standards, the rate of this PN sequence (called the chip rate) is 1.2288 Megachips per second (or Mcps). Many mobile users can share common transmit and receive channels with a baseband data rate of 9.6 kbps. The user information data is spread by a factor of 128 to a channel chip rate of 1.2288 Mcps. The resulting bandwidth of the spread signals is about 1.25 MHz.

The modulation and coding features of the IS-95 CDMA system are listed in Table 12.1. Modulation and coding details for the forward and reverse channels differ. The strong coding technique enables CDMA receivers to operate effectively at E_b/N_0 value of 5- to 7-dB range.

Table 12.1 *Modulation and coding features of the IS-95 CDMA system*

Channel bandwidth	1.23 MHz
Chip rate	1.2288 Mcps
Modulation scheme	Quadrature Phase Shift Keying (QPSK)
Nominal data rate	Rate Set 1: 9,600 bps; Rate Set 2: 14,400 bps
Coding type	Convolution coding
Interleaving procedure	With 20-ms span

The CDMA system uses voice activation and power-control features to minimise system interference. Voice activation is provided by using a variable-rate decoder that operates at a maximum rate of 8 kbps to a minimum rate of 1 kbps for Rate Set 1 (RS1), depending on the level of voice activity. A coding algorithm at 13.3 kbps for Rate Set 2 (RS2) is also supported. With the reduced data rate, the power control feature reduces the transmitter power to achieve the same bit-error rate. A precise power control as well as voice activation is critical to avoid the excessive transmitter signal power responsible for contributing the overall interference in the system. To overcome the effects of rapid multipath fading and shadowing, a time interleaver with a 20-ms time span, same as the time frame of the voice compression algorithm, is used with error-control coding.

EXAMPLE 12.8 Bandwidth Efficiency in IS-95 CDMA

The IS-95 standard uses QPSK modulation for its spread spectrum modulated baseband signal. The transmission bandwidth per carrier is 1.25 MHz, and the chip rate supported by the system is 1.2288 Mcps. Calculate the normalised bandwidth occupancy or bandwidth efficiency of the system.

Solution

RF channel bandwidth = 1.25 MHz (given)

Chip rate = 1.2288 Mcps (given)

We know that bandwidth efficiency = Chip rate/bandwidth

Therefore, bandwidth efficiency = 1.2288 Mcps/1.25 MHz

Hence, bandwidth efficiency = 0.98 chips/s/Hz

The PN-spreading codes are M-sequences generated by linear feedback shift registers of length 15 with a period of 32,768 chips. The IS-95A system uses two different types of PN codes: The short PN code is a pair of periodic binary PN sequences with a period of 2^{15}. These sequences are used for spreading and de-spreading signals into in-phase and quadrature components. The same short PN code is used by multiple base stations using the same frequency band by using different timing offsets in the code cycle. The long PN code is a sequence with a period of 2^{42}-1. It is used for spreading signals on the reverse channel as well as for power control burst randomisation and data scrambling purpose.

The orthogonal codes are used to isolate the transmissions between different channels within a cell, and

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The most suitable best-described PN sequences are maximal-length sequences (m-sequences) which are widely used for single-user spread-spectrum systems in military applications. Walsh sequences, Gold sequences, or Kasami sequences are popularly used in multi-user cellular CDMA systems because of their cross-correlation properties.

the PN spreading codes are used to separate the transmissions between different cells. The PN sequences are used to differentiate between several base stations in the service areas employing the same frequency. The same PN sequence is used in all base stations, but the PN sequence of each base station is offset from those of other base stations by some value. For this reason, base stations in IS-95 have to be synchronised on the forward channel. Such synchronisation is achieved using GPS.

In addition to the PN codes, there is a set of 64 mutually orthogonal codes called the Walsh codes. This is used for ensuring orthogonality between the signals for different users receiving from the same base station. The Walsh code is also used for modulation on the reverse channel of IS-95A. Thus, the logical channel on the forward channel is determined by the short PN code offset, the Walsh code assigned, and the assigned frequency of operation. On the reverse channel, the logical channel is determined by the short PN code offset and the long code offset, and the assigned frequency of operation.

On the forward channel, the base station simultaneously transmits the user data for all mobile users in the cell by using unique spreading sequence for each mobile user. A pilot code is also transmitted simultaneously at a higher power level, thereby allowing all mobile users to use coherent carrier detection while estimating the channel conditions. In the forward channel, transmissions originate at a single base station and transmissions for all mobile users are synchronised. It is thus possible to employ orthogonal spreading codes to minimise the interference between mobile users.

On the reverse channel, mobile users transmit whenever they have to transmit the data. As the transmissions from the mobile users are not synchronised, the way spreading is employed is to use the same orthogonal codes for orthogonal modulation to reduce the error rate. On the reverse channel, all mobile users respond in an asynchronous fashion and have ideally a constant signal level due to power control applied by the base station.

The logical channels of IS-95 CDMA can be classified as the control and traffic channels, as shown in Fig. 12.16. The control channels are the pilot channels, paging channels, sync channels, and access channels. The forward and reverse traffic channels are used to carry user data along with signaling traffic between the BS and the MS.

In traffic channels, variable-data-rate user information is carried. When the complete user information is replaced by the associated signaling and control data, it is called *blank and burst*. When part of the user information is replaced by signaling and control data, it is called *dim and burst.* The reverse channel and forward channel has a power control subchannel that allows the mobile phones to adjust its transmitted power by 1 dB every 1.25 ms.

The forward and reverse logical channels are quite different in its implementation and functions. The pilot, sync, paging, and traffic channels are combined on the same physical RF channel using CDMA. CDMA thus uses a bandwidth of 1.25 MHz for 55 traffic channels, which works out to be about 22.7 kHz per channel. This is similar

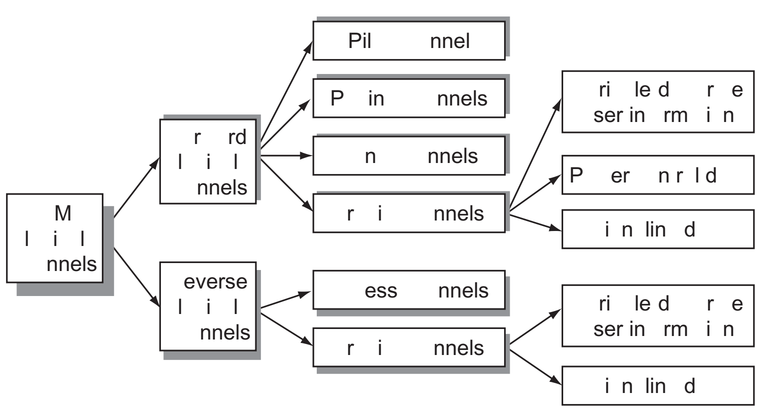

Fig. 12.16 *Classification of logical channels in IS-95*

to GSM (about 25 kHz per channel) system and apparently not as efficient as IS-54/136 TDMA (about 10 kHz per channel) system. However, the fact that all channels can be used in all sectors of all cells makes CDMA more spectrally efficient as much as ten to twenty times that for GSM. Since the signal quality in CDMA system degrades gracefully with increasing traffic, it is difficult to arrive at a definite maximum value for its capacity.

12.4 THE IS-95 CDMA FORWARD CHANNELS

Figure 12.17 depicts the IS-95 forward channel structure.

It consists of four types of logical channels—pilot channel, synchronisation channel, paging channel, and forward traffic channels. Each forward carrier channel contains one pilot, one synchronisation channel, up to seven paging channels, and a number of forward traffic channels. The pilot channel W_0 is always required. These channels are separated from one another using different spreading codes. The assignment of different Walsh codes to various logical channels is also shown in the figure. Some of the traffic channels are designated as fundamental data channel, supplementary data channel and mobile power control subchannel, depending on the nature of signaling and traffic data carried by them.

The main functions of the logical channel on the IS-95 forward channel are described now.

(a) Pilot Channel It is a downlink reference channel for synchronisation and tracking purposes. The pilot channel is used by the base station as a reference for all MSs. It does not carry any information and is used for signal strength comparisons and to lock onto other channels on the same RF carrier. It is typically the strongest channel, with 10 to 20 of the total combined power transmitted by a base station. It provides the capability for soft handover and coherent detection. There is one pilot channel which carries the phase reference for the other channels. It modulates a constant symbol and is used for channel estimation, which allows for coherent demodulation of the other channels that carry information data bits. Various Walsh codes are used for spreading various logical channels in IS-95. The pilot channel employs the Walsh code W_0 comprising of all 0s.

(b) Sync Channel It is used for providing synchronisation and configuration information to the mobile phones. There is one sync channel, which carries accurate timing information (synchronised to the GPS satellite system) that allows mobile users to decode the other channels. This channel provides the mobile users with system information such as the PN short sequence offset, the system time, the PN long code state,

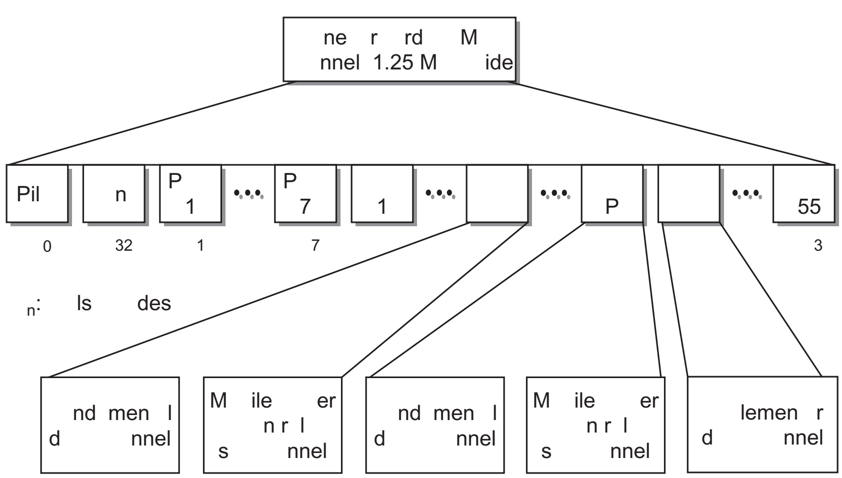

Fig. 12.17 *IS-95 forward channel structure*

paging channel data rate, system ID, network ID, and other related information. The synchronisation channel is assigned the Walsh code W_{32}.

(c) Paging Channel It is used for control information and sending paging messages to the mobile users in the system. There can be up to seven paging channels. This channel is used for sending short messages such as broadcast messages, details of registration procedures, pages for the called mobile users, the traffic channel information, the temporary mobile subscriber identity, response to access requests, and lists of neighbouring cell-sites and their parameters, and various other short messages for individual mobile users. The seven paging channels are assigned the Walsh codes W_1 to W_7.

(d) Forward Traffic Channel It is the main forward traffic-bearing channel which provides a dedicated link between the cell-site and the mobile user. It carries the speech or user data. The remaining fifty-five logical channels are used for traffic channels. The forward traffic channels are assigned the Walsh codes W_8 to W_{31} and W_{33} to $W_{63.}$ Supplement traffic channels are added dynamically to meet the required data rate. The modulation scheme employed for transmission of spread signal in the forward channel is QPSK.

The channel-coding operations in the forward channel use 20 ms frames for all logical channels except for the sync channel, which is coded using 26.666 ms frames. For error protection, a 1/2-rate convolutional code is used. The receiver can use the Viterbi algorithm for optimal decoding of the encoded data. Because the system uses variable data rates, the number of bits generated by the vocoder in one frame changes depending on the voice activity. The symbols are repeated for the lower data rate frames to ensure constant symbol rate at the modulation stage. For protection against bursts of errors, the frame data is interleaved to improve the overall BER on the link.

In the forward channel, all logical channels that are transmitted by the same cell-site together can be synchronised. Consequently, the system uses Walsh–Hadamard codes of length 64 for spreading and combining the logical channels to reduce the forward channel intracell multiple-access interference to negligible. The Walsh–Hadamard functions are combined with rate-1/2 FEC encoding leading to the overall chip rate of 1.2288 Mcps. Any information contained in the form of symbols (after encoding and interleaving) is modulated by Walsh codes which are obtained from Hadamard matrices.

Each Walsh code identifies one of the 64 forward channels. After the channel symbols are spread using the orthogonal codes, they are further scrambled into the in-phase and quadrature phase signals, also called the short PN-spreading codes. These codes are not orthogonal, but possess excellent autocorrelation and cross-correlation properties to minimise interference among different channels.

12.4.1 Pilot Channel

One key feature of the IS-95 CDMA transmission technique is the provision of a pilot channel in the forward link. The pilot channel allows very fast synchronisation and reliable channel tracking in the system. To make efficient use of the channel resources, the mobile user must be able to synchronise quickly which may not be easy with long spreading codes. The unmodulated pilot channel is always present which simplifies the synchronisation and channel-tracking process. In the IS-95 standard, the data are FEC encoded to allow operation at low signal-to-noise values. As a result, the data estimates prior to decoding at the mobile phone can be unreliable. If the mobile phone uses the pilot channel to track the channel variations, there is no need to estimate the data because of unmodulated pilot channel. Since the mobile phone receives a number of different synchronised forward channels, switching from one channel to the other channel need not resynchronise. Consequently, the mobile phone tracks only the pilot channel.

The way the pilot channel is generated by a base station is shown in Fig. 12.18. The pilot channel is intended to provide a reference signal for all mobile users within a cell that provides the phase reference for coherent demodulation.

The pilot-signal level for all base stations is about 4–6 dB stronger than all other channels including the traffic channel with a constant value. The pilot channel is used to lock onto all other logical channels on the same RF carrier. It is also used for comparing the received signal levels. The pilot signal does not carry any information and is assigned the all-zero Walsh code (W_0). It carries an unmodulated DSSS signal that is transmitted continuously by all base stations. It is also spread using the PN-spreading code to identify the base station. The pilot signals are quadrature pseudorandom binary sequence signals with a period of 32,768 (2^{15}) chips. Since the chip rate is 1.2288 Mcps, the pilot pseudorandom binary sequence corresponds to a period of 26.667 ms. This is equivalent to 75 times the pilot-channel code repetitions every 2 seconds. The pilot signals from all base stations use the same pseudorandom binary sequence, but each base station is identified by a unique time offset of its pseudorandom binary sequence. The way to identify the cell-site is to offset the PN sequence by some number of chips. In IS-95, the PN sequences are used with offsets of 64 chips that provide 512 possible unique spreading code offsets relative to the 0 offset code, providing for unique cell-site identification in dense microcellular areas as well. The short PN offsets are illustrated in Fig. 12.19.

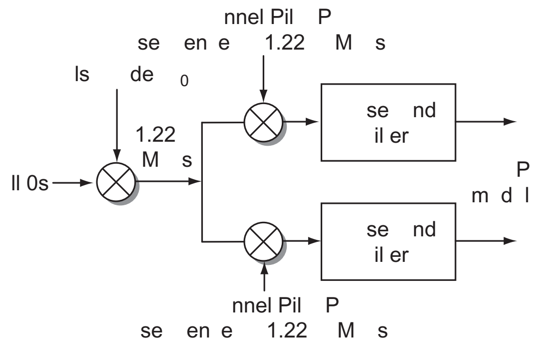

Fig. 12.18 *Pilot channel processing in IS-95 CDMA system*

The pilot channel is transmitted at a power approximately 20 of the total transmitted power for all channels from a particular cell-site. To minimise interference, all other forward channels are transmitted at very low power, and FEC coding is used to achieve acceptable performance. At these low operating powers, unreliable receiver synchronisation could actually degrade the performance of the channels. Since the same pilot channel is shared by all mobile user terminals, the average increase in power per mobile user required to achieve this performance is very small. This results into fast and reliable synchronisation.

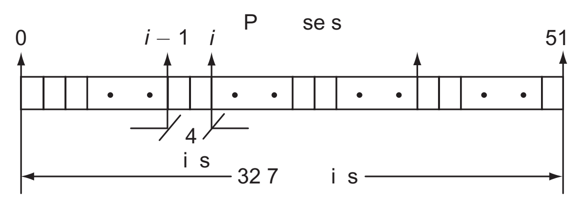

Fig. 12.19 *Short PN offsets used in a pilot channel*

A mobile user processes the pilot channel to find the strongest signal. The processed pilot signal provides an accurate estimation of time delay and the magnitude and phase of the three multipath components. These signal components are tracked in the presence of fast fading. For this purpose, coherent detection with combining technique is used at the mobile receiver. The chip rate on the pilot channel and on all other forward logical channels is locked to a precise system time provided by the Global Positioning System (GPS). Once the mobile user identifies the strongest pilot offset by processing the multipath components from the pilot channel correlator, it examines the signal on its sync channel which is locked to the pseudorandom binary sequence signal on the pilot channel. Since pilot channels are transmitted by each cell-site continuously, a mobile user can monitor the pilot channels of not only its serving base station but also from all the surrounding base stations and select the strongest one.

This procedure is often implemented by having a tracking receiver in the mobile phone that both monitors the strongest pilot channel and provides the synchronisation information necessary for demodulating the traffic and other channels. A second search receiver is continuously testing the pilot channels from adjacent cells and measuring their signal strength to determine whether and when a handover should occur. Because the users are mobile and the mobile users will occasionally cross cell boundaries. So they need to handover operations to a new cell-site. From a performance viewpoint, it is highly desirable that handover process should be transparent to the user; that is, users should be unaware that they have crossed a cell boundary.

12.4.2 Sync Channel

The sync channel is used to acquire initial time synchronisation. The way the sync channel is generated by the base station is shown in Fig. 12.20. It is assigned the Walsh code W_{32} for spreading. Note that it uses the same PN spreading codes for scrambling as used in the generation of the pilot channel.

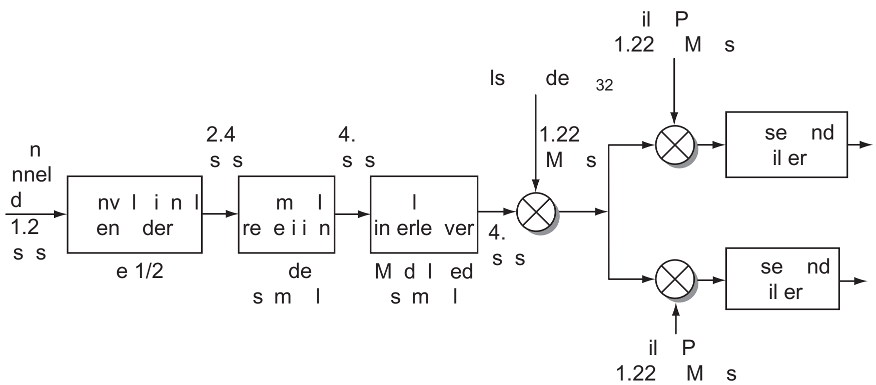

Fig. 12.20 *Sync channel processing in IS-95 CDMA system*

The sync channel is an encoded, interleaved, and modulated spread spectrum signal that is used along with the pilot channel to acquire initial time synchronisation by the mobile users in the system. The sync channel data always operates at a fixed rate of 1,200 bps. After a rate convolutional encoding, the data rate is increased to 2,400 bps, repeated to 4,800 bps, and then block interleaving over the period of the pilot pseudorandom binary sequence is employed. Each of the interleaved symbols uses four Walsh symbols. The sync channel message parameter includes the system identification, the network identification, the offset of the PN short code sequence, the state of the PN-long code, the system time, the leap seconds (number of leap seconds that have occurred since the start of system), local time offset, and the paging channel data rate (that is, 4800 bps or 9600 bps).

EXAMPLE 12.9 IS-95 sync channel modulation parameters

List the major IS-95 sync channel modulation parameters

Solution

Data rate	1200 bps
PN chip rate	1.2288 Mcps
Code rate	1/2
Code symbol repetition	2
Modulation symbol rate	4800 symbols per second (sps)
PN chips per modulation symbol	256
PN chips per bit	1024

The mobile user finds the information concerning the particular base station on the sync channel because the sync channel is time aligned with the pilot channel of its serving base station. The sync-channel message contains the time of day and long-code synchronisation to ensure that long-code generators at the base station and mobile user are aligned and identical. The sync channel message itself is long and may occupy more than one sync-channel frame.

12.4.3 Paging Channel

The paging channel is used to page the mobile user when there is an incoming call. It carries the control messages for call set-up to the mobile user. It provides the mobile users with system information and instructions, in addition to acknowledging messages following channel access requests on the reverse access channels by them. Figure 12.21 shows how a paging channel message is generated. It employs Walsh codes 1 to 7 ($W_0 - W_7$) so that there may be up to seven paging channels. There is no power control on a per-frame basis for the pilot, sync, and paging channels. The paging channel is additionally scrambled by the 42-bit PN long code mask.

The paging channel operates at a data rate of 4800 bps or 9600 bps. The long code is generated using a paging channel long-code mask of length 42. This means that the PN long code is generated by a linear feedback shift register of length 42 and has a period of 2^{42} chips. The major IS-95 paging channel parameters are listed in Table 12.2.

The paging channel is divided into 80-ms time slots called the *paging channel slots*. The IS-95 system accommodates two modes of paging — unslotted and slotted. In the *unslotted mode* of operation, the mobile user is required to monitor. all paging slots. However, in the *slotted mode*, a mobile user listens for pages only at pre-defined times during its page slots. This feature allows the mobile user to turn off its receiver for

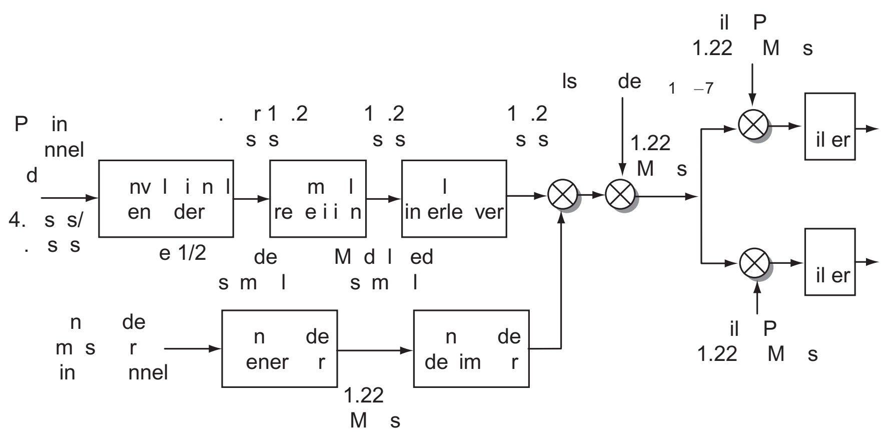

Fig. 12.21 *Paging channel processing in IS-95*

Table 12.2 *IS-95 paging channel modulation parameters*

S. No.	*Parameter*	*Value 1*	*Value 2*
1.	Data rate (bps)	4800	9600
2.	PN chip rate	1.2288 Mcps	1.2288 Mcps
3.	Code rate	1/2	1/2
4.	Code symbol repetition	2	1
5.	Modulation symbol rate (sps)	19,200	19,200
6.	PN chips per modulation symbol	64	64
7.	PN chips/bit	256	128

most of the time, saving the battery power and thereby increasing time between battery charging. The paging channel carries information to allow the network to identify the calling as well as called subscriber's number, convey information to the mobile user by means of alert signals, supply display information to be displayed by the mobile user, and indicate the number of wait messages.

The messages carried by the paging channel include system parameter, access parameters, CDMA channel list, neighbour list, unslotted or slotted page, page message, typical order message, channel assignment, data burst message, authentication challenge or shared secret data and feature notification message.

12.4.4 Forward Traffic Channel

The forward traffic channels carry the actual user information in digitally encoded voice or data form. The forward traffic channel has two possible rate sets called RS1 and RS2. RS1 supports data rates of 9600 bps, 4800 bps, 2400 bps, and 1200 bps. For voice traffic, the speech is encoded at a basic data rate of 8550 bps. After additional bits are added for error detection, the gross data rate is 9600 bps. The base station can select the data-transmission rate on a frame-by-frame basis. The full channel capacity is not used when the mobile user is not speaking. During quiet periods, the data rate is lowered to as low as 1200 bps. The 2400-bps rate and the 4800-bps rate is used to mix digitised speech and signaling data. A data rate of 9600 bps can support multiplexed traffic data and signaling data. Data rates of 4800 bps, 2400 bps, and 1200 bps can support only primary traffic information.

The forward traffic channel frame is 20 ms long. The forward channel frame structure for Rate Set 1 is shown in Figure 12.22.

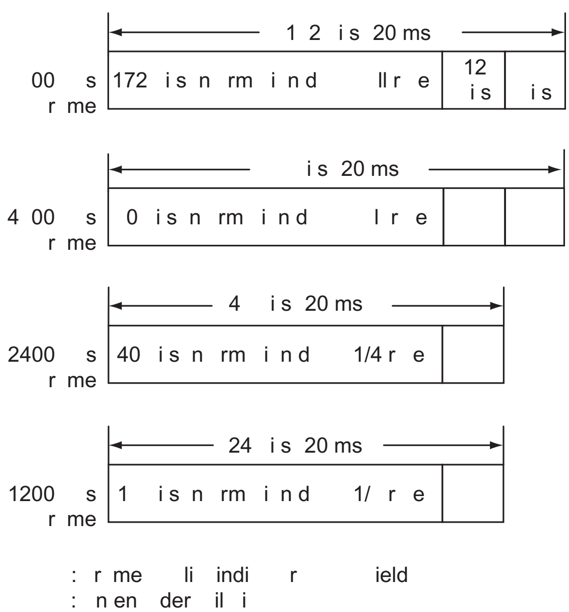

Fig. 12.22 *Forward-traffic-channel frame structure for Rate Set 1*

Table 12.3 shows the number of information bits, frame error control check (CRC) bits, and tail bits in Rate Set 1.

The procedure in which the information data symbols are processed for transmission on a forward traffic channel for the rate set RS1 is shown with the help of a functional block schematic of CDMA forward traffic channel in Fig. 12.23, depicting how the spreading works for one voice signal.

On the forward traffic channels, the digitised speech or user data is transmitted in 20-ms blocks with forward error correction provided by a convolutional encoder with rate 1/2, constraint length $L = 9$ to protect against transmission errors. This doubles the effective data rate to a maximum of 19.2 kbps. For lower data rates, the encoder output bits (called code symbols) are replicated to yield the 19.2-kbps rate. Symbol repetition is used to increase the data rate to 19.2 kbps. There is no repetition for 9.6 kbps-encoded voice and a repetition of four times for 9.6 kbps-encoded voice. The vocoder produces a voice signal with a maximum bit rate of 9.6 kbps.

Following the encoding block, is a symbol repetition/puncture block. The purpose of this block is to either repeat or puncture (remove) some of the bits produced by the convolutional encoder. The requirement is to produce a constant symbol rate for the subsequent processing stages, regardless of the input data rate. For the fundamental channel rate of 9.6 kbps, the second block has no effect. However, the systems can also support

Table 12.3 *Forward-traffic-channel frame structure for rate set 1*

Data rate (bps)	*Information (bits)*	*CRC (bits)*	*Tail (bits)*	*Total (bits)*
9600	172	12	8	192
4800	80	8	8	96
2400	40	0	8	48
1200	16	0	8	24

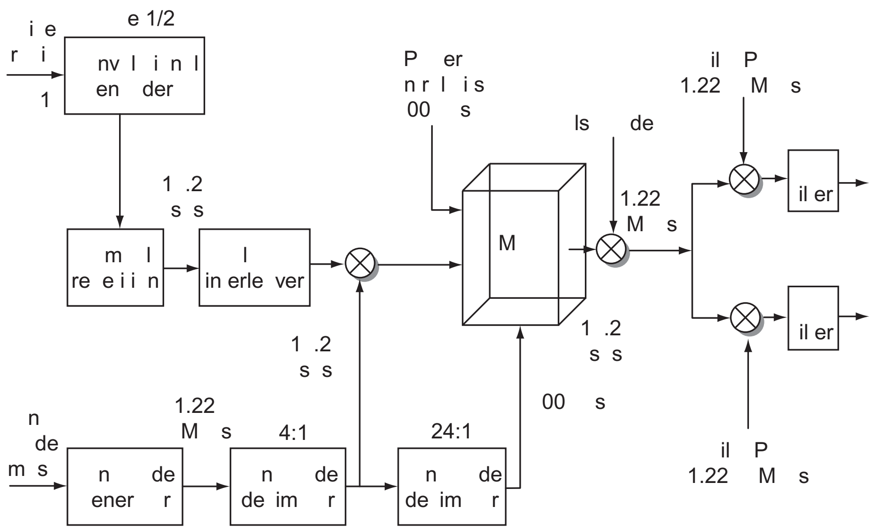

Fig. 12.23 *Forward-traffic-channel processing in IS-95 (Rate Set 1)*

data rates of 1200 bps, 2400 bps, 4800 bps, and 9600 bps. If the data rate is 4800 bps then each bit would be repeated twice.

With the addition of these error-correction bits, the samples are then interleaved over time in blocks to reduce the effects of errors by spreading them out—this process increases the bit rate for one voice channel to 19.2 kbps. This process improves the robustness of the communication link and reduces its sensitivity to multiple-access interference and fading. The output of the symbol repetition block is fed to the interleaver, which has a time span of 20 ms and contains 384 channel bits. It is a row-column interleaver, with 64 rows and six columns. The array is filled in column order, but the rows are read out in a permuted order. The 20-ms interleaver size matches the size of the voice frame generated by the vocoder in this system.

Following the interleaver, the data bits are scrambled. The purpose of this is to serve as a privacy mask and also to prevent the sending of repetitive patterns, which in turn reduces the probability of users sending at peak power at the same time. The scrambling is accomplished by means of a long code that is generated as a pseudorandom number from a 42-bit long shift register. A long code of $2^{42} - 1$ ($= 4 \times 10^{12}$) is generated containing the mobile user's equipment serial number embedded in the long-code mask (with speech privacy, the user long-code mask does not use the equipment serial number). The shift register is initialised with the user's equipment serial number. The scrambled data is multiplexed with power control information that steals bits from the scrambled data. The multiplexed data remains at 19.2 kbps and is changed to 1.2288 Mcps by the Walsh code W_i assigned to the ith user traffic channel. The output of the long-code generator is at a rate of 1.2288 Mbps, which is 64 times the rate of 19.2 kbps, so only one bit in 64 is selected (by the decimator function). The resulting data stream is XORed with the output of the block interleaver.

The IS-95 standard defines a very long maximal-length sequence of length 2^{42}-l bits $= 4 \times 10^{12}$ bits for this purpose. This scrambler is referred to as the long code; it is synchronised and used systemwide by all base stations and all mobile users. The base station applies modulo-2 operations, constituting a different mask to the state of this scrambler for each mobile user to generate the scrambler bits. The mask is determined by the mobile user's phone number. This can be shown to be equivalent to using a different initial state of the scrambler for each mobile user. Consequently, all communications that are intended for a specific mobile user are scrambled with a sequence known only at that mobile user. The scrambler does not spread the data. The long-code generator is clocked at the chip rate and thus runs 64 times faster than the channel bit rate. For this reason, the output of the long-code mask is decimated before being used as a scrambling sequence. The long code repeats only after 2^{42}-l bits and is used for encryption, not for spreading. The signal remains at 19.2 kbps after this process.

The power-control information is inserted in the forward traffic channel. A power-control subchannel is continuously transmitted on the forward traffic channel. The power-control function of the base station robs the traffic channel of bits at a rate of 800 bps. These are inserted by stealing modulated data symbols. The 800-bps channel carries information directing the mobile user to increment, decrement, or keep stable its current output-transmitting power level. A single power-control bit replaces 2 data symbols. A logic 0 specifies that the mobile increases its mean output power level by 1 dB (nominal), and a logic 1 indicates a decrease in mean output power level by 1 dB (nominal). The power-control data stream is multiplexed into the 19.2 kbps by replacing some of the code bits, using the long-code generator to encode the bits. This is done to randomise the location of the power control bits to avoid any spikes due to periodic repetition. The forward traffic channels are multiplexed with power control information for the reverse channel power control.

The DSSS process spreads the 19.2 kbps to a rate of 1.2288 Mbps using one row of the 64×64 Walsh matrix. One row of the matrix is assigned to a mobile user during call set-up. If a 0 bit is presented to the XOR function then the 64 bits of the assigned row are sent. If a 1 is presented then the bitwise XOR of the row is sent. Thus, the final bit rate is 1.2288 Mbps. This digital bit stream is then modulated onto the carrier using a QPSK modulation scheme. QPSK involves creating two bit streams that are separately modulated. In the IS-95 scheme, the data are split into I (in-phase) and Q (quadrature-phase) channels and the data in each channel are XORed with a unique short code. The short codes are generated as pseudorandom numbers from a 15-bit long shift register. It may be noted here that the traffic channels are scrambled with both the PN long code and the PN short codes to further reduce interference among channels. The power level of the traffic channel depends on its data transmission rate.

Forward logical channels which are not used for sync or paging channels can be used for traffic. Thus, the total number of traffic channels at a base station is 63 minus the number of sync plus paging channels in operation at that base station. Information on the forward traffic channels include the primary traffic (voice or data), secondary traffic (data), and signaling in frames of 20-ms length.

The major IS-95 forward link traffic channel modulation parameters for Rate Set 1 are listed in Table 12.4.

Table 12.4 *IS-95 forward-traffic-channel modulation parameters for RS1*

S. No.	*Parameter*	*Value 1*	*Value 2*	*Value 3*	*Value 4*
1.	User data rate (bps)	1200	2400	4800	9600
2.	Code-symbol repetition period	8	4	2	1
3.	Coding rate (bits per code symbol)	1/2	1/2	1/2	1/2
4.	Modulation symbol rate (sps)	19,200	19,200	19,200	19,200
5.	PN chips per modulation symbol	64	64	64	64
6.	PN chips per bit	1024	512	256	128
7.	PN chip rate (Mcps)	1.2288	1.2288	1.2288	1.2288

When the forward traffic channel is used for signalling, typical messages that can be sent include order message, authentication challenge, mobile registration confirmation, alert with information, data burst, in-traffic system parameters, power control, neighbour list update, hand-off direction, shared secret-data update, extended hand-off direction, and send burst DTMF message in case of three-way conference calls.

EXAMPLE 12.10 Processing gain in IS-95

In an IS-95 system, calculate the processing gain in dB if the baseband data rate is 9.6 kbps, 4.8 kbps, 2.4 kbps, and 1.2 kbps in Rate Set 1 (RS1). If the error-correction codes increase the data rate to 19.2 kbps, recalculate the processing gain. Comment on the results obtained.

Solution

The processing gain, G_p is given by $G_p = B_c/R_b$, where B_c is RF (transmitted) bandwidth or channel bandwidth, and R_b is the baseband (before spreading) bandwidth or baseband data rate.

In IS-95, the RF bandwidth, $B_c = 1.2288$ Mcps (standard)

Step 1. *To calculate the processing gain, G_p at 9.6 kbps*

The baseband data rate, $R_b = 9.6$ kbps (given)

We know that processing gain, $G_p = B_c/R_b$

Therefore, $G_p = 1.2288$ Mbps/9.6 kbps $= 128$

Expressing it in dB form, we get

$$G_p\,(\text{dB}) = 10 \log 128 = 21 \text{ dB}$$

Step 2. *To calculate the processing gain, G_p at 4.8 kbps*

The baseband data rate, $R_b = 4.8$ kbps (given)

We know that processing gain, $G_p = B_c/R_b$

Therefore, $G_p = 1.2288$ Mbps/4.8 kbps $= 256$

Expressing it in dB form, we get

$$G_p\,(\text{dB}) = 10 \log 256 = 24 \text{ dB}$$

Step 3. *To calculate the processing gain, G_p at 2.4 kbps*

The baseband data rate, $R_b = 2.4$ kbps (given)

We know that processing gain, $G_p = B_c/R_b$

Therefore, $G_p = 1.2288$ Mbps/2.4 kbps $= 512$

Expressing it in dB form, we get

$$G_p\,(\text{dB}) = 10 \log 512 = 27 \text{ dB}$$

Step 4. *To calculate the processing gain, G_p at 1.2 kbps*

The baseband data rate, $R_b = 1.2$ kbps (given)

We know that processing gain, $G_p = B_c/R_b$

Therefore, $G_p = 1.2288$ Mbps/1.2 kbps $= 1024$

Expressing it in dB form, we get

$$G_p\,(\text{dB}) = 10 \log 1024 = 30 \text{ dB}$$

Step 5. *To calculate the processing gain, G_p at 19.2 kbps*

If the error-correction codes are considered a form of spreading as well, then they increase the data rate.

Data rate after error-correction, $R_b = 19.2$ kbps (given)

We know that processing gain, $G_p = B_c/R_b$

Therefore, $G_p = 1.2288$ Mbps/19.2 kbps $= 64$

Expressing it in dB form, we get

$$G_p\,(\text{dB}) = 10 \log 64 = 18 \text{ dB}$$

Comment on the results A signal-to-noise ratio of about 7 dB is required at the receiver output for a reasonable bit-error rate. This means that the signal-to-noise ratio in the RF channel can be about (7 – 18) = –11 dB for satisfactory operation; that is, the signal power can be 11 dB less than the noise power. This is typical of spread spectrum systems.

Similarly, RS2 supports 14400 bps, 7200 bps, 3600 bps, and 1800 bps. The forward-traffic-channel frame is 20 ms long. Table 12.5 shows the number of information bits, frame error Control Check (CRC) bits, and tail/reserved bits in rate set 2.

Table 12.5 *Forward-traffic-channel frame structure for rate set 2*

Data rate (*bps*)	*Reserved/ lag bit*	*Information* (*bits*)	*CRC* (*bits*)	*Tail* (*bits*)	*Total* (bits)
14400	1	267	12	8	288
7200	1	125	10	8	144
3600	1	55	8	8	72
1800	1	21	6	8	36

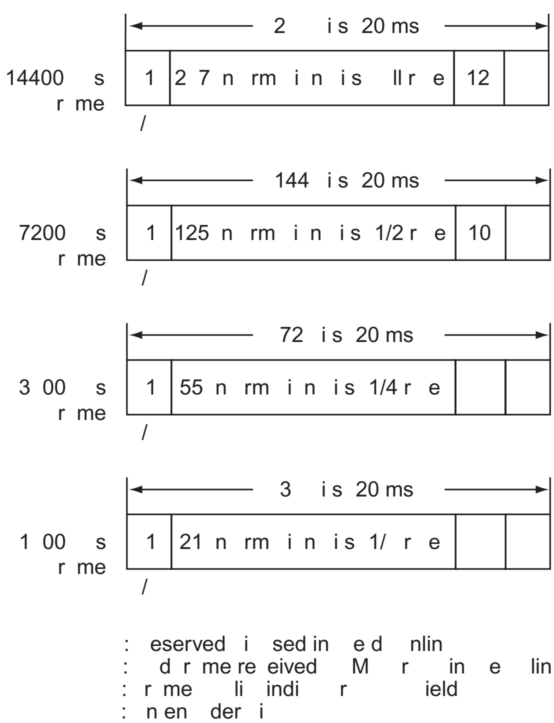

Fig. 12.24 *Forward-traffic-channel frame structure for Rate Set 2*

The forward-channel frame structure for Rate Set 2 is shown in Fig. 12.24.

The procedure in which the symbols are processed for the Rate Set RS2 in the forward traffic channel is shown in Fig.12.25. If the higher rate of 14.4 kbps is used, then the second block punctures one out of every three channel bits. In the case of RS2, the rate at the output of the symbol repeater is 28.8 kbps that is punctured by selecting only four out of every six bits. This reduces the data rate to 19.2 ksps at the input of the block interleaver.

Walsh codes that can be assigned to forward traffic channels are available at a cell-site or sector (W_2 through W_{31}, and W_{33} through W_{63}). Only 55 Walsh codes are available for forward traffic channels if all the paging channels as well as the sync channel is used in the system. The speech is encoded using a variable rate encoder to generate forward traffic data depending on voice activity. The power-control subchannel is continuously transmitted on the forward traffic channel. Spreading occurs when the 19.2 kbps baseband data stream is multiplied by one of the 64 Walsh codes. During the multiplication process, if the data bit is zero, the Walsh code bits are transmitted unchanged, and if the data bit is one, all Walsh code bits are inverted. Each of the Walsh codes has a bit rate of 1.2288 Mbps. The output bit stream is at 1.2288 Mbps, which is 64 times as great a data rate as for the baseband signal at 19.2 kbps. Therefore, the transmitted signal bandwidth is 64 times as great as it would be for the original signal, assuming the same modulation scheme for each transmitted signal.

The major IS-95 forward link traffic channel parameters for Rate Set 2 are listed in Table 12.6.

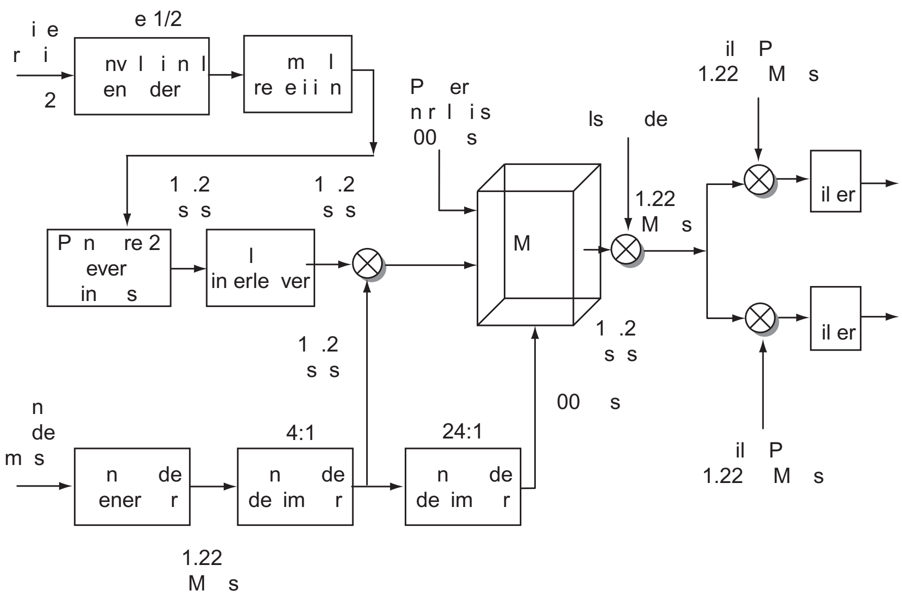

Fig. 12.25 *Forward-traffic-channel processing in IS-95 (Rate Set 2)*

Table 12.6 *IS-95 forward-traffic-channel parameters for RS2*

S. No.	*Parameter*	*Value 1*	*Value 2*	*Value 3*	*Value 4*
1.	Data rate (bps)	1800	3600	7200	14400
2.	Code symbol repetition period	8	4	2	1
3.	Coding Rate (bits per code symbol)	1/2	1/4	1/2	1/2
4.	Puncturing rate	4/6	4/6	4/6	4/6
5.	Effective code rate	3/4	3/4	3/4	3/4
6.	Modulation symbol rate (sps)	19,200	19,200	19,200	19,200
7.	PN chips per modulation symbol	64	64	64	64
8.	PN chips per bit	682.67	341.33	170.67	85.33
9.	PN Chip Rate (Mcps)	1.2288	1.2288	1.2288	1.2288

12.4.5 Processing of Forward Channels

Figure 12.26 summarised how the four forward channels—pilot channel, paging channel, sync channel, and forward traffic channels—are generated, prior to combining them in the cell-site transmitter. This illustration enables to figure out the salient processes involved in generation of each of the logical channels.

The four forward channels are configured as follows:

The Pilot Channel (Channel 0) It is an unmodulated channel that is spread by the Walsh–Hadamard code W_0, as shown at the top of the figure. It is a continuous signal on a single channel. This channel allows the mobile phone to acquire timing information, provides phase reference for the demodulation process, and provides a means for signal strength comparison for the purpose of hand-off determination. The pilot channel consists of all logical zeros.

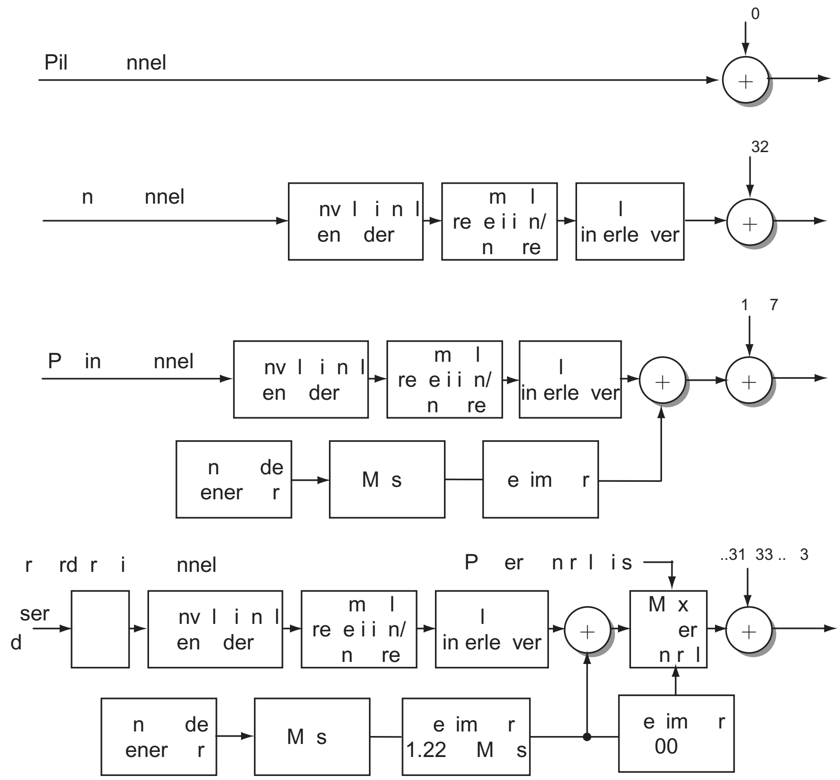

Fig. 12.26 *Processing of IS-95 forward channels*

The Sync Channel (Channel 32) It has low-rate data that is convolutionally encoded and interleaved before being spread by the Walsh–Hadamard code W_{32}. It is used with the pilot channel to acquire initial time synchronisation. It is a 1200-bps channel used by the mobile user to obtain identification information about the system (system time, long-code state, protocol revision, etc.).

The Paging Channel (Channels 1 to 7) It has low-rate data that is also encoded and interleaved. Prior to being spread, the paging data is randomised with a scrambler that is specific to the mobile user for which the page is intended. The data is then spread by Walsh–Hadamard code W_1–W_7 (for up to 7 paging channels). The paging channel provides system information and instructions to the MSs. It may contain messages for one or more mobile users at the same time. When the MS is to receive a call, it will receive a page from the BS on an assigned paging channel. There is no power control for the paging channel on a per-frame basis.

The Forward Traffic Channel (Channels 8 to 31 and 33 to 63) It is FEC encoded and interleaved and then scrambled with a sequence that is specific to the intended mobile user. The signaling and control bits are multiplexed with the forward traffic channel which are meant for adjusting the power of the mobile transmitter. The traffic channel is then spread with an assigned Walsh–Hadamard sequence that is orthogonal to all other Walsh–Hadamard codes. The forward channel supports 55 traffic channels. The traffic channel supports data rates of up to 9600 bps in RS1 and up to 14,400 bps in RS2.

All the forward logical channels use the same bandwidth. The Walsh code is used to distinguish among the different channels which are the 64 orthogonal 64-bit codes derived from a 64 $\times$ 64 Walsh matrix. The fundamental format of spreading procedure for all forward channels in IS-95 is shown in Fig. 12.27.

Power Control in Forward Traffic Channel The transmission format for the forward traffic channel includes bits that are used for closed-loop power control. These bits control the transmit power of the user terminal. To transfer the power-control information from the cell-site to the mobile user, power-control bits are multiplexed with the traffic bits at a low rate. This is done after scrambling and FEC encoding, to minimise the delay in the power-control loop. Power-control bits are included by replacing traffic bits at a very low data rate. In IS-95, the power-control bits are added at a rate of 800 Hz. Since the traffic bits are FEC encoded and the discarded traffic bits are rare, a minimal effect on the traffic channel performance is experienced. At the mobile user terminal, a power control bit that is a binary 1 means 'decrease the power', while a 0 means 'increase the power'. Since these bits are not protected by FEC encoding, they can have a relatively high error rate. So it is desirable that the power-control bits should be averaged over time to determine the direction (increase or decrease) in which the power should be adjusted.

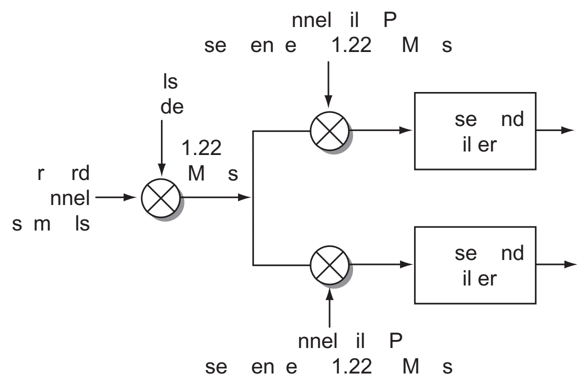

Fig. 12.27 *Basic spreading procedure on the forward channels in IS-95*

Multiplexing of CDMA Forward Logical Channels The signals from each forward logical channel (pilot, sync, paging, and traffic) are modulo-2 added to I and Q PN short-code sequences. The I and Q spread signals are baseband filtered and sent to a linear adder with gain control. The gain control allows individual channels to have different power levels assigned to them. The CDMA system assigns different power levels to different channels using appropriate algorithms depending on the quality of the received signal at a mobile terminal. The I and Q baseband signals are modulated by I and Q carrier signals, combined together, amplified, and transmitted on one RF carrier through the base station antenna, as shown in Fig. 12.28. The net signal from the CDMA modulator is a complex quadrature signal that looks like noise.

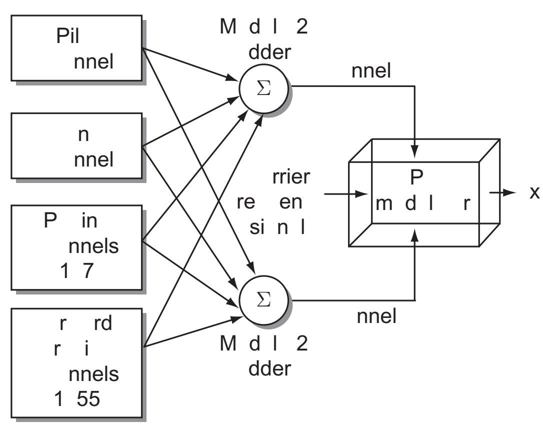

Fig. 12.28 *Multiplexing of CDMA channels*

All the forward logical channels can be added directly because all of them are synchronous in nature and use the orthogonal Walsh–Hadamard functions for spreading. This does not generate any multiple-access intercell or intracell interference. The output of the modulo-2 adder is no longer bipolar, as it is for the logical addition of a number of spread signals. Each forward traffic channel has a unique Walsh code assigned to it. These Walsh codes are orthogonal to each other, and the different symbol streams are spread by their respective Walsh codes in bipolar form and added together in a manner such that the weight given to each corresponds to the intended power in that channel. The forward traffic channel has a variable data rate. In the case of lower data rates, the symbols are repeated and the power of the traffic channel is reduced by the same repetition factor. This reduction ensures that the power transmitted for each information data bit before encoding and repetition remains identical.

The output of the modulo-2 adder is then I- and Q-modulated, with the same signal transmitted on both the in-phase (I) and quadrature (Q) channels. After the combined channels are I-and Q-modulated, the resulting bit sequence is further scrambled with the short code. The IS-95 standard defines two distinct short codes of length 2^{15}, one each for the in-phase and quadrature channels. The purpose of the short codes is to differentiate the signals radiated by different cell-sites. The period of the short code is 26.66 milliseconds. All of the cell-sites use the same short code, but with different timing offsets relative to a common reference. The receiver attempts to synchronise to this short code, and the strongest correlation identifies the nearest cell-site. There are 512 different offsets, corresponding to approximately 52 microseconds each; this offset is longer than any expected multipath delay, so channel effects are unlikely to cause synchronisation to the wrong cell-site.

12.5 THE IS-95 CDMA REVERSE CHANNELS

The CDMA reverse channel is fundamentally different from the forward channel. The reverse link supports up to 32 access channels and up to 62 traffic channels. The CDMA reverse channel employs OQPSK rather than the QPSK digital modulation scheme used in the forward channel. The OQPSK is closer to a constant envelop modulation scheme which provide for a more power-efficient implementation of the transmitter at the mobile phone. The QPSK modulation is easier for demodulation at the mobile phone.

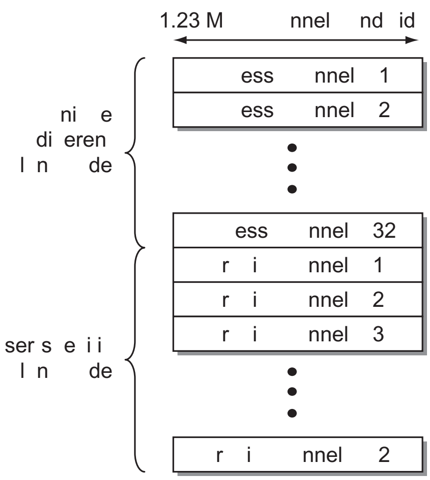

Fig. 12.29 *IS-95 channel structure—reverse channels*

In the reverse channel, there is no spreading of the data symbols using orthogonal codes. Instead, the orthogonal codes are used for waveform encoding. This means that the reverse channel employs an orthogonal modulation scheme that consumes bandwidth but reduces the error rate performance of the system. The reverse channel consists of up to 94 logical CDMA channels, each occupying the same 1.23-MHz bandwidth as shown in Fig. 12.29.

Figure 12.30 depicts the IS-95 reverse channel structure. It consists of two types of logical channels—the access channel and reverse traffic channels. These logical channels are used by the mobile users to access the system and transfer user data.

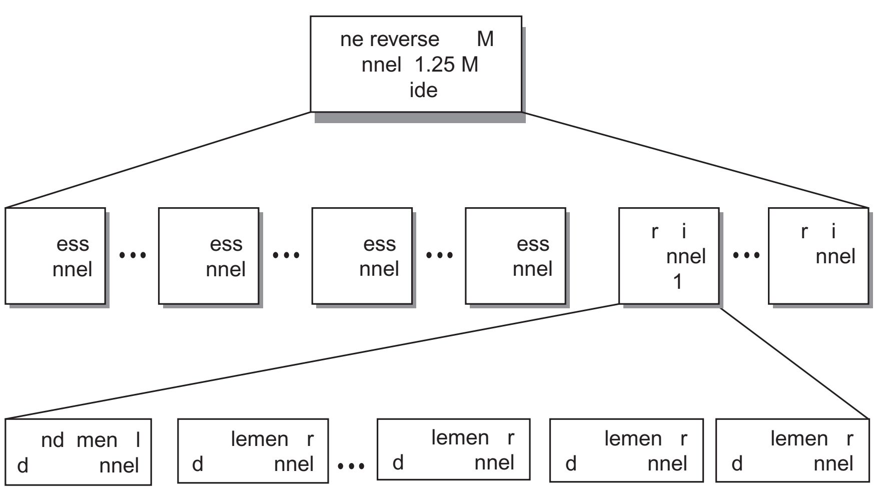

Fig. 12.30 *IS-95 reverse channels structure*

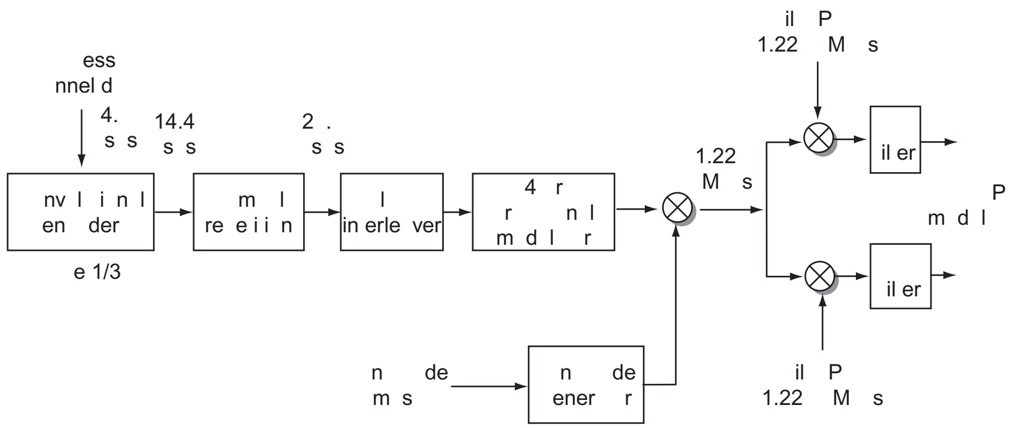

Fig. 12.31 Access channel processing in IS-95

12.5.1 Access Channel

An access channel is meant for signaling and control information, on which mobile users communicate short messages such as information on registration, call originations, and responses to pages, or providing other data to the cell-site. The system includes many of these reverse channel random-access channels. All the mobile users accessing a system share the same frequency assignment. When any mobile user places a call, it uses the access channel to inform the serving base station. The access channel is also used to respond to a page received from the base station.

The access channel uses a prearranged long-code offset. The access-channel data rate is 4,800 bps and each access channel message is composed of several access-channel frames lasting 20 ms. Thus an access-channel frame is 96 bits long. An access-channel preamble always precedes an access-channel message, and it consists of several 96-bit frames with all bits in the frame equal to zero. The actual message itself is fragmented into 96-bit frames that have 88 bits of data and 8 tail bits set to zero. Figure 12.31 shows a functional schematic, depicting access channel processing in IS-95 system.

The baseband information is error protected using a robust convolutional encoder of code rate 1/3 that increases the data rate to 14.4 kbps. Symbol repetition is employed to increase the data rate to 28.8 kbps. The data is then interleaved to combat fading. Every six bits is now mapped into 64 bits using the 64-ary orthogonal modulator. The orthogonal modulation is used due to the noncoherent nature of the reverse channel. The long PN code sequence, having an access number, a paging-channel number associated with the access channel, and other system data, is used to distinguish between different access channels. It spreads each of the bits at the output of the 64-ary orthogonal modulator by a factor of four that yields a chip rate of 1.288 Mcps. The actual start of transmission on the access channel is randomised to minimise collisions between multiple mobile users accessing the channel at the same time. All access channels corresponding to a paging channel have the same slot length. Different base stations may have different slot lengths.

EXAMPLE 12.11 | IS-95 access channel parameters

List the major IS-95 access channel parameters.

Solution

Data rate 4800 bps
Code rate (bits per code symbol) 1/3
Symbol rate before repetition 14,400 sps
Code symbol repetition (symbols per code symbol) 2
Symbol rate after repetition 28,800 sps
Transmit duty cycle ( ) 100

Code symbols per modulation symbol 6
Modulation symbol rate 4800 sps
Walsh chip rate 307.2 kcps
Modulation symbol duration 208.33 s
PN chips per code symbol 42.67
PN chips per modulation symbol 256
PN chips per Walsh chip 4

The messages carried by access channels include registration message (to inform the BS about MS location, status, identification, and other parameters), order message (BS challenge, shared secured data update registration and confirmation, MS acknowledgement, local control response, MS reject), data burst message, origination message, page response message, authentication challenge response message.

12.5.2 Reverse Traffic Channel

Reverse traffic channel carries the user speech or data. It is similar to the forward traffic channel, and is intended to transfer dedicated user data. The reverse traffic channel supports variable-data-rate operation. There are two sets of traffic-channel data rates. Rate Set 1 has a maximum data rate of 9.6 kbps, whereas Rate Set 2 has a maximum data rate of 14.4 kbps. Both rate sets further support full, half, quarter, and one-eighth rates with respect to the maximum data rate.

The traffic channels in the reverse channel are unique to each mobile user. Each mobile user has a unique long-code mask based on its electronic serial number. The long code mask is a 42-bit number, so there are $2^{42}–1$ different masks. The access channel is used by a mobile user to initiate a call, to respond to a paging channel message from the base station, and for a location update.

For Rate Set 1, the reverse traffic channel uses 9600 bps, 4800 bps, 2400 bps, or 1200 bps data rates for transmission. The duty cycle for transmission varies proportionally with the data rate being 100 at 9600 bps to 12.5 at 1200 bps. The reverse traffic channel processing is similar to the access channel except for the fact that the reverse channel uses a data burst randomiser. In the reverse channel, the mobile phone transmitters are not synchronised, and as a result, there is more multiple-access interference.

On the reverse CDMA channel in IS-95, all the transmission uses 20 ms frames. The mobile users cannot use truly orthogonal channels because they lack a phase-coherent pilot channel. Each mobile user would need its own pilot channel, which would use too much bandwidth. Therefore, they use a more robust error-control system.

Figure 12.32 shows a functional block schematic of CDMA reverse traffic channel, depicting how the spreading works for reverse traffic signal.

Most functions in the reverse traffic channel are qualitatively similar to the forward traffic channel, but there are two important differences:

– A rate-1/3, constraint length-9 convolutional code is used. While this code has the same asymptotic coding gain as the rate-1/2 code used in the forward channel, it has slightly greater gain over a large S/N value range. The rate-1/3 code implies that the basic channel bit rate is 28.8 kbps for the reverse channel. Thus, it outputs data at three times the input data rate, which is 3×9.6 kbps $= 28.8$ kbps. The data rate on the reverse traffic channel is variable; hence each frame may have a different data rate.
– The system does not use Walsh–Hadamard sequences for orthogonal spreading. This particular use would not be practical on the reverse channel, since all mobile users are not synchronised. The reverse traffic channel uses a form of orthogonal modulation. In particular, every six bits at the interleaver output are used to select one of 64 Walsh–Hadamard orthogonal sequences for transmission. In this case, the orthogonality is utilised to assist detection over fading wireless channels rather than to minimise multiple-access interference.

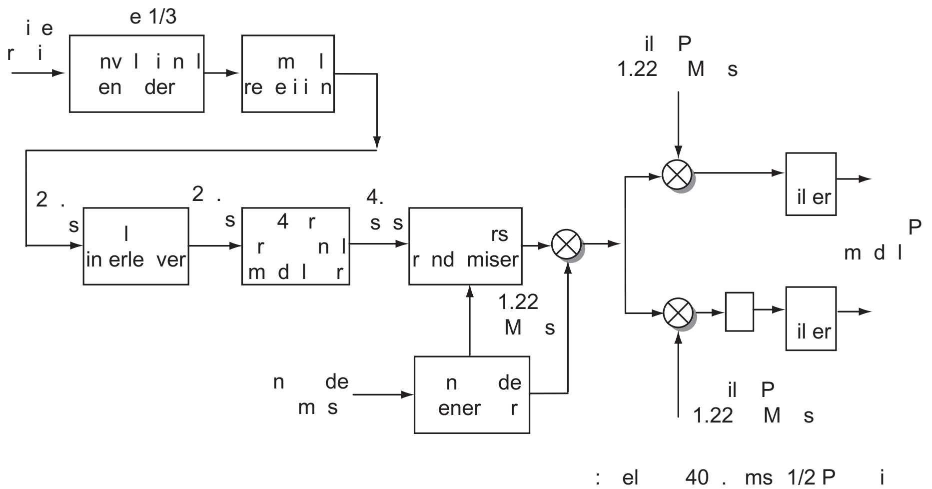

Fig. 12.32 CDMA reverse traffic channel processing

The purpose of Walsh codes is different on the reverse traffic channel. The long code is used to distinguish one mobile user from another mobile user, as each one uses a unique, though not necessarily orthogonal, long code. The Walsh codes are used to help the cell-site decode the message in the presence of interference. Each block of six information bits ($2^6 = 64$ different possible combinations) is associated with one of the 64 Walsh codes, and that code, rather than the actual data bits, is transmitted. Since each Walsh code is 64 bits long, this in itself does some spreading of the signal. The bit rate is increased by a factor of 64/6. The Walsh code mapping thus increases the data rate by 28.8 kbps $\times$ 64/6 = 307.2 kbps. The purpose of this encoding is to improve reception at the base station. Because the 64 possible codings are orthogonal, the block coding enhances the decision-making algorithm at the receiver and is also computationally efficient. Walsh modulation can be viewed as a form of block error-correcting code with $(n, k) = (64, 6)$ and $d_{min} = 32$.

Facts to Know!

The mobile transmitter interleaves all code symbols on the reverse traffic channel and the access channel prior to modulation and transmission. A block interleaver of 20-ms duration is used. It is an array with 18 columns and 32 rows. Code symbols corresponding to information data rate of 9600 bps (or repeated code symbols for lower data rates) are written into the block interleaver by columns filling the complete 32 $\times$ 18 matrix.

The data-burst randomiser is implemented to help reduce interference from other mobile users. The operation involves using the long-code mask to smooth the data out over each 20-ms frame. In the process of the DSSS function, the long code unique to the mobile is XORed with the output of the data burst randomiser to produce the 1.2288-Mcps final data stream, the same as for the forward channel. Each mobile user transmits at the same rate to produce the spread-spectrum signal received at the cell-site. The mobile user convolutionally encodes the data transmitted on the reverse traffic channel. The reverse traffic channel, like the forward traffic channel, supports voice data at two rate sets, RS1 and RS2. The mobile user uses a 1/3-rate convolution code when the traffic channel uses Rate Set I, and a 1/2-rate code when the traffic channel uses Rate Set 2. In either case, the data burst after coding and interleaving is at a rate of 28.8 kbps just before the 64-ary orthogonal modulation.

Prior to modulation, symbols are repeated in the case of lower data rates, and the frame data is interleaved for protection against burst errors. The modulation used is a 64-ary orthogonal modulation. The output of the

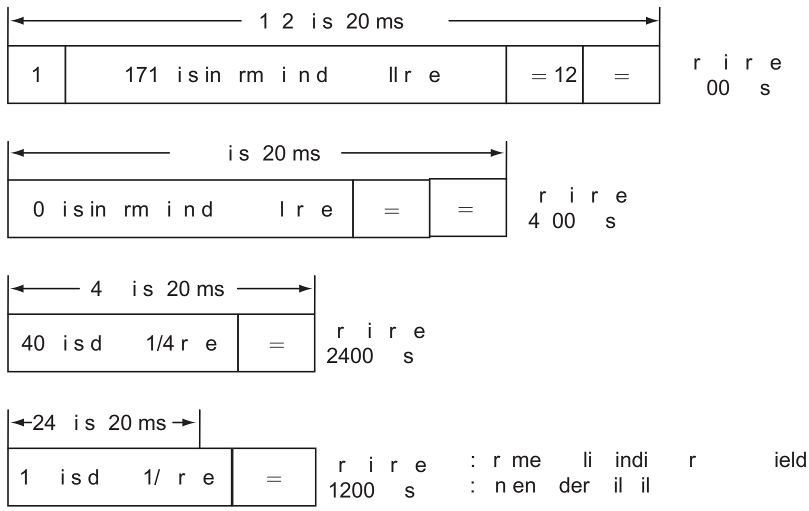

Fig. 12.33 *Reverse traffic channel frame structure for Rate Set 1*

64-ary orthogonal modulator is $28.8 \times 64/6 = 307.2$ kcps. After spreading by the long PN code by a factor of four, the final chip rate is $307.2 \times 4 = 1.2288$ Mcps. Randomised gating is used in the case of lower data rate frames to control the transmit power. A data randomiser is used in the fundamental code channel to mask out redundant data in case of symbol repetition. The reverse traffic channel sends information related to the signal strength of the pilot and frame error rate statistics to the base station. It is also used to transmit control information to the base station such as a hand-off completion message and a parameter response message. Because the power control is done by gating, the power level of the transmitted symbols is kept constant. The Viterbi algorithm provides optimal decoding at the receiver.

The reverse traffic channel is also broken into 20-ms traffic channel frames. The frame is further divided into 1.25-ms Power Control Groups (PCGs). There are thus $20/1.25 = 16$ PCGs in one frame. A data-burst randomiser randomly masks out individual PCGs depending on the data rate that result in less interference on the reverse traffic channel. For instance, at 4.8 kbps (half the data rate), eight PCGs are masked. In addition to voice traffic, the traffic channel can also be used to transfer signaling or secondary data. In the blank and burst case, the entire frame carries data. In the dim and burst case, part of the frame carries voice and part of it data. The frame structures for the reverse traffic channel are very similar to that of the forward traffic channel. The reverse channel frame structure for Rate Set 1 is shown in Fig. 12.33.

EXAMPLE 12.12 Long-code repetition time

Calculate the time for one complete repetition of the long code in CDMA forward channel.

Solution

$$\text{Number of bits in long code} = (2^{42} - 1) \text{ bits} = 4.4 \times 10^{12} \text{ bits} \quad \text{(standard)}$$

On the forward channel, the long code is exclusive-OR'd with the data bits and is sent at the data rate, not the channel bit rate. Therefore,

$$\text{Transmission data rate for long code} = 19.2 \text{ kbps or } 19{,}200 \text{ bits} \quad \text{(standard)}$$

$$\begin{aligned}\text{Time taken to transmit the complete long code} &= (4.4 \times 10^{12})/19200 \\ &= 229 \times 10^{6} \text{ s} \\ &= \mathbf{7.26\ years}\end{aligned}$$

That is why it is called the long code!.

The major IS-95 reverse link traffic channel parameters for Rate Set 1 are listed in Table 12.7. The major IS-95 reverse link traffic channel parameters for Rate Set 2 are listed in Table 12.8.

Table 12.7 *IS-95 Reverse traffic channel modulation parameters for RS1*

S. No.	*Parameter*	*Value 1*	*Value 2*	*Value 3*	*Value 4*
1.	Data rate (bps)	1200	2400	4800	9600
2.	Code rate (bits per code symbol)	1/3	1/3	1/3	1/3
3.	Symbol rate before repetition (sps)	3600	7200	14,400	28,800
4.	Code-symbol repetition	8	4	2	1
5.	Code-symbol rate after repetition (sps)	28,800	28,800	28,800	28,800
6.	Transmit duty cycle ( )	12.5	25.0	50.0	100.0
7.	Code symbols per modulation symbol (bits per Walsh symbol)	6	6	6	6
8.	Modulation symbol rate (sps)	4800	4800	4800	4800
9.	Walsh chip rate (kcps)	307.2	307.2	307.2	307.2
10.	Modulation symbol duration (μs)	208.33	208.33	208.33	208.33
11.	PN chips per code symbol	42.67	42.67	42.67	42.67
12.	PN chips per modulation symbol	256	256	256	256
13.	PN chips per bit	128	128	128	128
14.	PN chips per Walsh chip	4	4	4	4
15.	PN chip rate (Mcps)	1.2288	1.2288	1.2288	1.2288

Table 12.8 *IS-95 reverse traffic-channel parameters (RS2)*

S. No.	*Parameter*	*Value 1*	*Value 2*	*Value 3*	*Value 4*
1.	Data rate (bps)	1800	3600	7200	14400
2.	Code rate (bits per code symbol)	1/2	1/2	1/2	1/2
3.	Symbol rate before repetition (sps)	3600	7200	14,400	28,800
4.	Code-symbol repetition	8	4	2	1
5.	Code-symbol rate after repetition (sps)	28,800	28,800	28,800	28,800
6.	Transmit duty cycle ( )	12.5	25.0	50.0	100.0
7.	Code symbols per modulation symbol (bits per Walsh symbol)	6	6	6	6
8.	Modulation symbol rate (sps)	4800	4800	4800	4800
9.	Walsh chip rate (kcps)	307.2	307.2	307.2	307.2
10.	Modulation symbol duration (s)	208.33	208.33	208.33	208.33
11.	PN chips per code symbol	42.67	42.67	42.67	42.67
12.	PN chips per modulation symbol	256	256	256	256
13.	PN chips per bit	128	128	128	128
14.	PN chips per Walsh chip	4	4	4	4
15.	PN chip rate (Mcps)	1.2288	1.2288	1.2288	1.2288

The modulation and scrambling of the chip sequence are very similar to their use in the forward channel, except for the use of offset-QPSK modulation. The near-constant envelope properties of offset-QPSK make transmission more efficient for a power-limited mobile user. This digital bit stream is then modulated onto the carrier using an orthogonal QPSK modulation scheme. This differs from the forward channel in the use of a delay element in the modulator to produce orthogonality. The reason the modulators are different is that in the forward channel, the spreading codes are orthogonal, all coming from the Walsh matrix, whereas orthogonality of the spreading codes is not guaranteed in the reverse channel.

The modulation scheme is slightly different on the forward and reverse channels. Both use a form of Quadrature Phase-Shift Keying (QPSK). The cell-site uses conventional QPSK. With this system, the transmitter power has to go through zero during certain transitions. The mobile users delay the quadrature signal by one-half a bit period to produce offset QPSK. This has the advantage that the transmitter power never goes through zero, though the amplitude does change somewhat. Linear amplifiers are still used in the mobile transmitter, but the linearity requirements are not as strict for offset QPSK as they are for conventional QPSK. Offset QPSK would have no advantage for the base station because a single transmitter is used for all the multiplexed signals.

The output from either the access channel or the reverse traffic channel is processed through two modulo-2 adders—one for the I-channel and the other for the Q-channel. Two different short-code PN sequences are modulo-2 added to the data and filtered by a baseband filter. For a Q-channel, a delay of PN symbol (406.9 ns) is added before the filter. Thus, the reverse channel uses OQPSK.

The reverse traffic channel carries the typical messages which include order messages, authentication challenge response, flash with response, data burst, pilot signal strength measurement, power measurement report (FER statistics), status messages, origination continuation, hand-off completion, parameter response, and send burst DTMF message containing dialed digits.

12.6 CDMA CALL PROCESSING

A mobile user in CDMA goes through several states prior to getting a traffic channel assigned. The call processing from a mobile user can be categorised into four states such as system initialisation state, system idle state, system access state, and traffic channel state.

State 1: System Initialisation State In the system-initialisation state, the mobile user selects a system to use and then synchronise to a CDMA carrier by acquisition of pilot and sync channel along with its long-code and system timing. The mobile user searches all the PN-I and PN-Q possibilities to acquire a pilot channel and selects the strongest pilot channel. The processed pilot signal provides an accurate estimation of time offset of the pilot channel by using GPS. Once the pilot channel is acquired, the sync channel is acquired using the W_{32} Walsh function. Then the mobile user obtains the system configuration parameters such as the spreading code of the access channel and the paging channel.

State 2: System Idle State The mobile user then enters the system-idle state where it monitors the paging channel. In this state, the mobile user can receive the messages and orders from the base station containing necessary parameters to initiate or to receive a call, initiate a registration process, and initiate a message transmission. An idle hand-off, or change of paging channel, occurs when a mobile user has moved from the coverage area of one base station to the coverage area of another base station during the idle state. The mobile user determines that a hand-off should occur when it detects a new pilot that is sufficiently stronger than the current pilot. In the idle state, three pilot sets (active, neighbour, and remaining) identified by short PN offsets are maintained.

State 3: System-Access State If a call is being initiated or received, the mobile user enters the system-access state where the necessary parameters are exchanged. The mobile phone transmits its response on the reverse access channel and the base station transmits its response on the paging channel.

State 4: Traffic-Channel State When the access attempt is successful by the mobile user, it enters the traffic-channel state. In this state, two-way speech communication takes place, with associated control messages replacing digital speech by either blank-and-burst signalling (in which the complete speech packet is replaced with signalling) or dim-and-burst signalling (in which the part of the speech packet is replaced with signalling). In addition, power control messages are sent by bit puncturing on the forward traffic channel in which two gross-data bits are replaced by a single power control bit.

12.7 POWER CONTROL IN CDMA SYSTEM

The system-level objectives of maximum utilisation of the available radio spectrum in the cellular communication systems often translates into the maximum number of simultaneous mobile users served by the system with acceptable signal quality. Most of the cellular systems are interference limited. This implies that the number of users in the system are not limited by channel parameters such as fading and receiver noise, but seriously limited by interference. Subsequently, to achieve improvement in spectral efficiency and increase in system capacity requires minimising the interference. This is directly related to minimising the transmitted power of each mobile user at all times of its operation.

In other words, power control is simply the technique of controlling the mobile transmit power so as to affect the cell-site received power, and hence the overall carrier-to-interference (C/I) value. Like all other cellular communication systems, CDMA is also an interference limited system. But in a CDMA system, it is not cochannel and adjacent channel interference which pose serious bottlenecks, but rather the interference from other mobile users transmitting in the same frequency band at the same time in the system.

Facts to Know!

Adaptive power-control schemes reduce the near–far interference problem, and optimise the system capacity and spectral efficiency. An ideal power control scheme ensures that the signals received at the cell-site from all the mobile users within a cell remain at the same power level, independent of the propagation path loss, location and/or movement of the mobile users.

It is desirable for maximum efficiency that the power received at the base station from all the mobile users served by it must be nearly equal. If the received signal power is too low, there is a high probability of bit errors or frame errors. If the received signal power is too high, interference increases. Power control is applied at both the mobile users as well as the base station. There are several power-control mechanisms that can be based on the signal strength perceived by the base station or can depend on other system parameters. Accordingly, the base station or the mobile user can either initiate the power control.

On one hand, it is important to implement good power control in order to minimise the near–far effect. On the other hand, the transmit power should be adequate enough so as to encounter the effects of fading and shadowing in order to maintain a good link quality. In the case of CDMA, an important factor is that the received signal strength may be reasonably good but data frames are still received in error due to interference. Therefore, it is preferred to use the Frame Error Rate (FER) instead of the received signal strength for power-control decisions. It is safer to allow a range of 0.2 per cent to 3 per cent FER, with error bursts of up to four frames. It is also assumed that an FER of 1 per cent with maximum error bursts of two frames will give optimum results.

In IS-95, a slow mobile-assisted power control is employed on the forward channel. Since non-coherent detection is employed on the reverse channel, therefore implementation of power control is a must on the reverse channel. There are mainly two types of power-control mechanisms: an open-loop power control and a closed-loop power control. Because all the voice channels occupy the same frequency and time slots, the received signals from multiple mobile users located anywhere within the periphery of the serving cell

Facts to Know!

In a DS-CDMA system, all traffic channels within one cell share the same radio channel simultaneously. Neighbouring cells may also be assigned the same or adjacent channel. Practically, some mobile users are near to the base station, while others are far from it. A strong signal received at the base from a near-in mobile user masks the weak signal from a far-end mobile user.

must all have the same received signal strength at the base station for proper detection. A mobile user that transmits unnecessarily at a large power may jam or interfere with the received signals of all the other mobile users.

The advantage of implementing strict power control is that the mobile user can operate at the minimum required E_b/N_0 for adequate performance. This increases battery life and reduces the size and weight of the mobile-user phone equipment.

12.7.1 Open-Loop Power Control

Open-loop power control refers to the procedure whereby the mobile user measures its received signal level and adjusts its transmit power accordingly. The cell-sites are not involved in power-control mechanism. The reference channel on the forward channel used to determine the transmit power of the mobile user is the pilot channel. The principle of operation of the open-loop power control scheme is that a mobile user closer to the cell-site should transmit less power because its propagated signal encounters a small path loss. Mobile users that are far away from a cell-site or in deep fade should transmit at a larger transmit power to overcome the greater path loss in signal strength.

– In open-loop power control at the mobile user, the mobile user senses the received signal strength of the pilot channel and can adjust its transmit power based on that. If the signal level of the pilot channel is very strong, it can be assumed that the mobile user is very close to the base station. Therefore, the mobile-user transmit power level should be reduced.
– In open-loop power control at the base station, the base station decreases its transmit power level gradually and waits to receive the Frame Error Rate (FER) message from the mobile user. If the FER is beyond a specified level, the base station increases its transmit power level on the corresponding forward traffic channel.

Facts to Know!

An adaptive open-loop power control is based on the measured signal strength of the received pilot by the mobile, and then setting its transmit power after estimating the path loss between the base transmitter and mobile receiver. It is assumed that the forward and reverse links have the same propagation path loss. However, this arrangement may not be precise and accurate.

The fundamental advantage of open-loop power control is that its operation is quite fast. It does not need to wait for a round-trip delay between the cell-site and the mobile user to control its transmit power. The main disadvantage of open-loop power control is the limited correlation between received power levels on the reverse channel and the forward channel. There is reasonable correlation between the average power levels of the reverse channels and forward channels. The variations in the average power levels are due to shadowing effects (slow log-normal fading) caused by nature of the terrain, dense foliage, and high-rise building structures. For example, if the mobile user goes into the shadow region of a building, both the reverse-channel and forward-channel signals may be significantly attenuated for improper operation of the open-loop power-control mechanism. Hence, open-loop power control is good for tracking median power levels and slow variations such as those due to shadowing. It can provide good coarse correction factors in the mobile transmit levels for these types of signal variations.

For fast Rayleigh fading conditions, open-loop power control is not effective. The reverse channel and forward channel transmissions usually occur in separate frequency bands and there is virtually no correlation between fast fading effects on these transmissions. Even small variations in the phase of the carrier signal may cause large differences in the propagation characteristics. In some cases, open-loop power control is implemented in an asymmetric fashion. The purpose of the asymmetric open-loop power control is to minimise the interference caused to other mobile users in the event of an error in the power measurement. A high received-signal level means that the transmit power should be decreased immediately, while a low received-signal level means that the transmit power should be increased slowly.

At the time of switching on the mobile user, it adjusts its transmit power based on the total received power from all cell-sites on the pilot channel. If the received signal strength of the pilot channel is very strong, the mobile user transmits a weak signal to the cell-site. Otherwise it transmits a strong signal to the cell-site. The power received at the cell-site from all mobile users must be nearly equal, say within 1 dB, for the system to work properly. The power level is first set approximately by the mobile user, and then tightly controlled by the cell-site. When the mobile user is switched on for the first time, it measures the received power from the cell-site, assuming that the signal losses on the forward and reverse channels are equal, and sets the transmitter power accordingly.

As a first approximation, the transmitted power of the mobile, P_{tm} (dBm) is set as

$$P_{tm}\text{ (dBm)} = -76\text{ dB} - P_{rm}\text{ (dBm)} \tag{12.18}$$

where P_{rm} is the received power by the mobile.

The mobile user begins by transmitting at the power level determined by the above expression and increases its transmit power if it does not receive acknowledgement from the cell-site. This could happen if a substantial amount of the received power at the mobile user is actually from adjacent cells. Once a link with the nearest base station is established, the open-loop power setting is adjusted in 1-dB increments after every 1.25-ms by commands from the cell-site, to keep the received power from all mobile users at the same level.

EXAMPLE 12.13 | Open-loop power control at CDMA mobile phone

A CDMA mobile phone measures the received signal level from its serving cell-site as -85 dBm.
(a) What should the mobile transmitter power be set to as a first approximation?
(b) Once the mobile transmitter power is set as computed in Part (a), its serving cell-site needs the mobile user to change its transmit power level to +5 dBm. How long will it take to make this change?

Solution

(a) Received signal strength by mobile, $P_{rm} = -85$ dBm (given)

As a first approximation, the transmitter power of the mobile is given by the expression

$$P_{tm}\text{ (dB)} = -76\text{ dB} - P_{rm}\text{ (dB)}$$

Therefore, $P_{tm}\text{ (dB)} = -76\text{ dB} - (-85\text{ dBm})$

Hence, $P_{tm}\text{ (dB)} = +9\text{ dBm}$

(b) Required transmitter power of the mobile = +5 dBm

Difference in mobile transmitter levels = +9 dBm – (+5 dBm) = 4 dB

The mobile transmitter power level is adjusted by 1-dB step after every 1.25 ms. That is, time taken to adjust 1-dB power level = 1.25 ms

Number of steps required to adjust mobile transmitter level to +5 dBm = 4

Time needed to adjust mobile transmitter level to +5 dBm = 4 × 1.25 ms

Hence, time needed to adjust mobile transmitter level = 5 ms

12.7.2 Closed-Loop Power Control

Within a cell, multiple mobile users employ orthogonal codes, and the primary source of interference is from mobile users of other cells. A mobile-assisted power control on the forward channel is implemented to reduce intercell interference. The mobile user periodically reports the FER on the reverse channel to the cell-site, which will then adjust its transmit power accordingly. Maximum transmit power values are preset to prevent excessive interference and minimum transmit power values are preset to avoid voice quality to degrade.

With closed-loop power control of the mobile user, the cell-site measures its received signal strength and then sends a corresponding command to adjust its transmit power to the desired level. In closed-loop power control at the mobile user, power-control information is sent to the mobile users from the cell-site. This message indicates either an increment or decrement in the transmit power.

Facts to Know!

In adaptive closed-loop power control, the base-station receiver detects the received signal power from the mobile user on the reverse link, and then transmits control bits to it on the forward link to adjust the transmit power of the mobile to the desired level.

The delay between measurement of the received signal level and implementation of power control is a critical parameter in closed-loop power control. This delay can occur due to a number of reasons. To provide a reasonably accurate measurement of the received signal level, the measurement must be averaged over several symbol periods. Moreover, the power-control adjustment message must be multiplexed with the ongoing transmission, which means a processing delay. To minimise this delay, power-control adjustments are not forward error-correction encoded or interleaved. Generally, encoding would add an additional delay of 10 to 20 milliseconds. The correction in the transmit power incurs the signal transmission delay corresponding to the distance between the mobile user and the cell-site. At the mobile receiver, the power-control message data should also be averaged over several symbols because the power-control adjustments are uncoded and thus less reliable. This process further incurs delay.

Typically, power-control adjustments are sent as a single bit command, for example, a logical 1 to decrease and a logical 0 to increase the transmit power by a predetermined value of about 1 dB. Sending these power-control bits on a frequent basis minimises the delay between received signal measurement and power-control implementation. In some systems, power-control bits are sent at the rate of 800 Hz to 1500-Hz. Although this improves response times, yet it is an added overhead for the system. Due to the delay in the closed power-control loop, power control will not be able to compensate for the fast Rayleigh fading effects that occur at normal vehicle speeds. In fact, at high vehicle speeds, closed-loop power control may degrade performance. But interleaving provides a significant benefit at these faster fading rates, which may in turn compensate for the shortcomings of closed-loop power-control mechanism.

Fast closed-loop power control is based on the measurements of the received signal strength at the cell-site and comparisons with an expected received signal strength, which is based on the likelihood of noise and interference on the received path according to the system design. There may be a slower closed-loop power control that is based on frame-error-rate measurements in order to ensure that no systematic errors are present in the comparison procedure. The FER measurements are often based on cyclic redundancy checks included in each frame and are used for error detection after error-correction decoding. Since the system is designed to tolerate a specified frame error rate, the cell-site monitors the frame rate to ensure that it is within the predetermined threshold value. If the error rate is very high then the mobile user is not getting the acceptable quality. If it is very low then the mobile user is generating an unacceptably high level of interference. If the error rate is not in the desired range then the appropriate power-control adjustments are carried out taking into account this information.

Facts to Know!

The power-control error of practical closed-loop power-control mechanism is about 1.5 dB against ideal requirement of 0 dB, that is, all transmitted signals from various mobile users located at varying distances from the base station should be received simultaneously with 0-dB differences. This would have eliminated the near–far interference problem and maximise the capacity of the CDMA cellular system.

It is important to note that fading will not be correlated between a mobile user and two different cell-sites. With regard to intercell interference, power control limits the maximum power transmitted by mobile users in adjacent cells but does not control how this interference affects individual mobile users. This implies that a mobile user may have a weak signal at the desired cell-site momentarily; it may serve as a relatively strong interferer at another cell-site at the same time.

In a practical system, the combination of the open- and closed-loop power control techniques must be carefully considered. The power control should be based on a measured received signal level, not the S/N value. Using the signal-to-noise ratio as a measurement reference can result in an unpredictable situation because an increase in the signal level in interference-limited systems will also increase the noise level, resulting in the same S/N value.

If the number of mobile users in a particular service area increases, the system is interference limited, and increasing the transmit power will not benefit any mobile user or group of mobile users as the total interference also increases. It is quite possible that interference from many cells can raise the noise floor to such a level that coverage holes or weak signal spots may be created in the area where spreading/processing gain or the coding gain is not sufficient to overcome the interference levels.

There are some disadvantages of power control. Firstly, it may not be desirable to set transmission powers to higher values because battery power at the mobile user is a limited resource that needs to be conserved. Secondly, increasing the transmitted power on one channel, irrespective of the power levels used on other channels, can cause inequality of transmission over other channels. Finally, power-control techniques are restricted by the physical limitations on the transmitter power levels not to exceed the unwanted radiations.

The IS-95 specifies complex procedures for regulating the power transmitted by each mobile user. The minimum and maximum Effective Isotropic Radiated Power (EIRP) is specified instead of limiting the maximum transmit power only. The maximum radiated power of base stations is limited to 100 W per 1.23-MHz CDMA channel. Table 12.9 lists the maximum EIRPs for five classes of CDMA mobile phones.

Table 12.9 *CDMA mobile phone power levels*

Class of CDMA mobile phone	*Minimum EIRP*	*Maximum EIRP*
I	–2 dBW (630 mW)	3 dBW (2.0 W)
II	–7 dBW (200 mW)	0 dBW (1.0 W)
III	–12 dBW (63 mW)	–3 dBW (500 mW)
IV	–17 dBW (20 mW)	–6 dBW (250 mW)
V	–22 dBW (6.3 mW)	–9 dBW (130 mW)

12.8 SOFT HAND-OFFS

IS-95 uses both power control and soft hand-off as an interference reduction mechanism. Soft hand-off refers to the process by which a mobile user is in communication with multiple candidate cell-sites before finally deciding to transfer data through one of them. It is highly desirable to implement soft hand-off in power-controlled CDMA systems. Power control is closely related to soft hand-off. With soft hand-off, a conditional decision is made whether to hand off or not. In soft hand-off, a mobile user is temporarily connected to more than one base station at the same time. Whenever a mobile user enters a region in which the received signal levels from two adjacent base stations are comparable within some threshold level of each other, the mobile user enters the soft hand-off condition in which it is connected to both the base stations. The mobile user remains in this state until the received signal level from one base station clearly predominates. Then the mobile user is assigned exclusively to that base station to continue its operation.

The implementation of soft hand-off has its basis in the near–far problem and the associated power-control mechanism. If a mobile user moves far away from a cell-site and continues to increase its transmit power to compensate for the near–far problem, there is a possibility that it may end up in an unstable situation of operation. It will also cause a lot of interference to mobile users operating in neighbouring cells. To avoid this situation and ensure that a mobile user is connected to the cell-site with the maximum received signal strength level, a soft hand-off strategy is implemented.

A mobile user will continuously track all nearby cell-sites and communicate with multiple cell-sites for a short while, if necessary. Then it decides which cell-site to select as its point of connection or getting services. In the soft hand-off state, the transmissions from the mobile user arriving at both base stations are forwarded to the mobile switching centre, which estimates the quality of the two signals and selects the better one.

In general, the soft hand-off procedure involves several cell-sites. A controlling primary serving cell-site coordinates the addition or removal of other cell-sites to the on-going call during soft hand-off. The primary cell-site uses a hand-off direction message to indicate the pilot channels to be used or removed as part of the soft hand-off process. After hand-off, the primary cell-site is also changed. The signals from multiple cell sites are combined either in the base station controller or mobile switching centre and processed as a single call. The mobile user detects a reasonably strong pilot signal from a neighbouring cell-site and informs the primary serving cell-site. After a traffic channel is set up with the neighbouring cell-site, the signal is selected from both cell-sites at the BSC/MSC.

When the signal strength of the pilot signal from the primary serving cell-site starts reducing, the mobile user will request its change. This avoids the dropping of calls that sometimes occurs when a hand-off is unsuccessful in other systems, perhaps because there are no available channels in the new cell. The disadvantage of soft hand-off is a considerably increased load on the cell-sites and the switching network.

In the IS-95 standard, three types of soft hand-offs are defined.

(a) Softer Hand-off In the softer hand-off or intersector hand-off as shown in Fig. 12.34(a), the mobile user communicates with two sectors of the same cell at the same time. A RAKE receiver at the base station combines the best versions of the traffic frame from the diversity antennas of the two sectors into a single traffic frame.

(b) Soft Hand-off In the soft hand-off, or intercell hand-off, as shown in Fig. 12.34(b), the mobile user communicates with two (or three) sectors of different cells at the same time. A three-way soft hand-off may end by first dropping one of the sectors of any base station and becoming a two-way soft hand-off. A soft hand-off uses considerably more network resources than the softer hand-off.

(c) Soft-softer Hand-off In the soft-softer hand-off as shown in Fig. 12.34(c), the candidate sectors for hand-off include two sectors from the same cell and a third sector from a neighbouring cell. Network resources

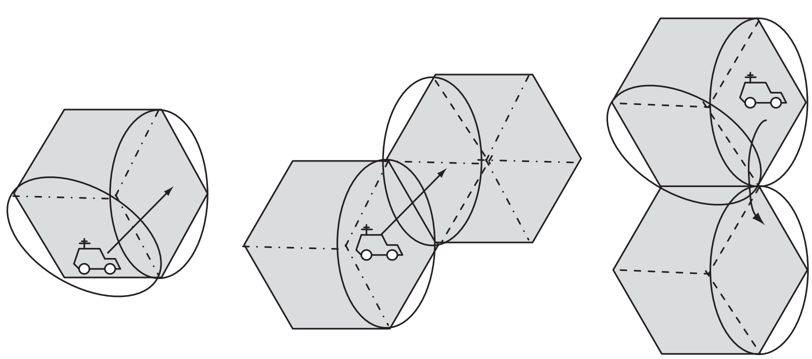

Fig. 12.34 *Type of soft hand-offs in IS-95*

required for soft-softer hand-off include the resources for a two-way soft hand-off between two cells plus the resources for a softer hand-off of the cell involving two sectors. In all cases, the hand-off decision mechanism is more or less the same.

The pilot channels of each cell are involved in the hand-off mechanism because the pilot channel only has constant transmit power and is not subjected to power control. So it provides a reliable measure of the received signal strength level. The mobile user maintains a list of pilot channels that it can receive and classifies them into the four different categories.

- The active set consists of pilot channels that are being continuously monitored or used by the mobile user. The mobile user is equipped with three RAKE fingers that allows it to monitor up to three pilot channels.
- The candidate set can have at the most six pilot channels different from those included in the active set but that have sufficient received signal levels to be demodulated and used in demodulating the associated traffic channels.
- The neighbour set contains pilot channels that belong to neighbouring cells and are intimated to the mobile user by a system-parameters message on the paging channel.
- The remaining set contains all other possible pilot channels in the system. Because the receiver uses a RAKE to capture multipath components, it employs search windows to track each of the sets of pilot channels. The search windows are large enough to capture all the multipath components of the pilot channel from a cell-site but small enough to minimise searching time. The multipath delays are a function of the distance between the mobile user and the cell-site, and, consequently, the search windows are also affected.

If there is an isolated three-sector cell, most of the cell has an E_b/N_o larger than 7 dB. In the soft hand-off region, the E_b/N_o value from each base station is about 3 dB which provides sufficient diversity gain. This situation is illustrated in Fig. 12.35(a). If numbers of cells or sectors are deployed, as shown in Fig. 12.35(b), there may be some regions where it is not possible to communicate due to high noise level.

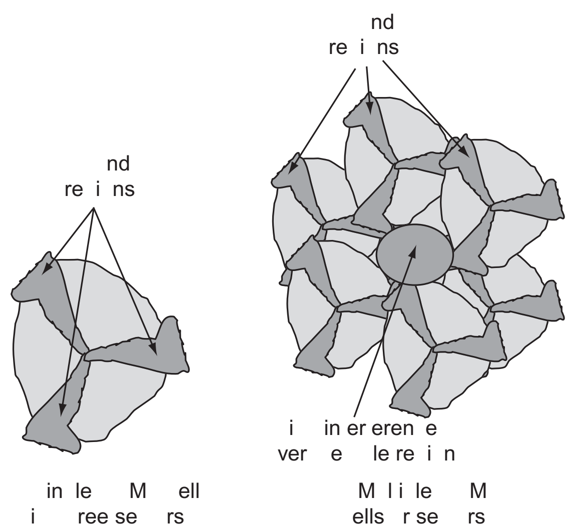

Fig. 12.35 *(a) Single CDMA cell (b) Multiple CDMA cells*

It is often possible to cover the same area with fewer cells or sectors to reduce the total interference levels. It is recommended that more than three cells or sectors should not cover such a region. Proper selection of the cell-site, down tilting the cell-site antennas with minimum radiated power levels are necessary to lower the overall interference level.

12.9 RAKE RECEIVER CONCEPT

In a multipath propagation environment, the multiple versions of a signal arrive more than one chip interval apart from each other. The mobile receiver can recover the signal by correlating the chip sequence with the dominant received signal. The remaining signals are treated as noise. In order to achieve better performance, the receiver attempts to recover the signals from multiple paths and then combine them with suitable delays. This is the principle of operation of the RAKE receiver used in IS-95 systems.

RAKE receivers are commonly used in DSSS receivers in CDMA cellular mobile phones which enables to provide a robust signal reception in a hostile mobile radio environment. There is no pilot channel available for the reverse channel transmission. A pilot channel is useful for obtaining good carrier-channel estimation, making it possible to perform coherent detection and combining of multipath components. The absence of estimation of carrier phase and amplitude necessitates either noncoherent or differential coherent detection. It is needed to acquire and track timing of all signal paths. The effect of multipath interference can be reduced by combining direct and reflected signals in the receiver.

EXAMPLE 12.14 | Principle of RAKE receiver

Illustrate the principle of a RAKE receiver.

Solution

Figure 12.36 illustrates the principle of the RAKE receiver. The original information signal in the binary form to be transmitted is spread by the exclusive-OR (XOR) operation with the PN spreading code. The spread bit sequence is then modulated for transmission over the wireless channel.

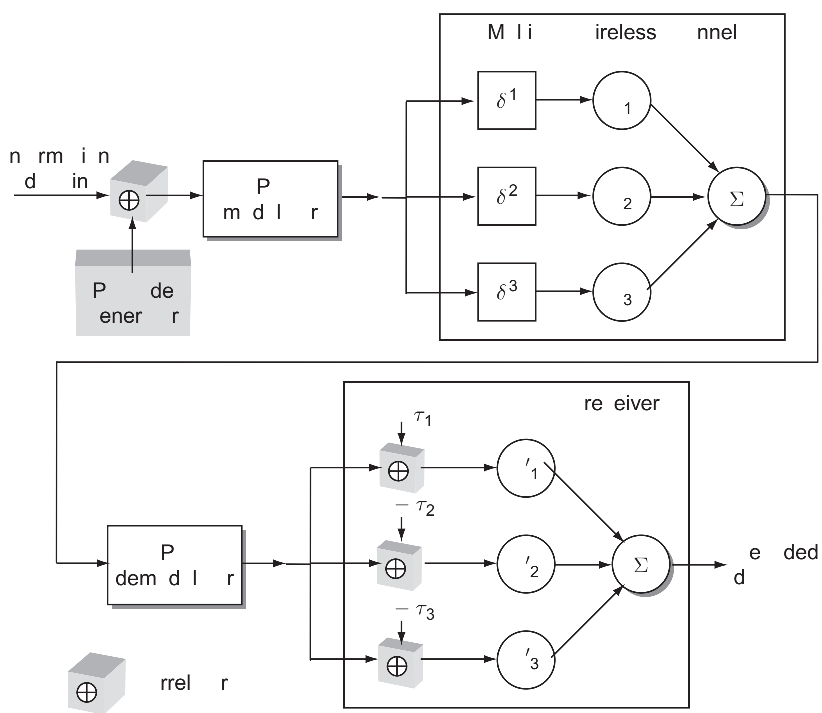

Fig. 12.36 | *Principle of RAKE receiver*

The wireless channel generates multiple copies of the same signal due to multipath effects, each with a different amount of time delay (δ_1, δ_2, etc.), and attenuation factors (a_1, a_2, etc.). At the receiver, the combined signal is demodulated. The demodulated chip stream is then fed into multiple correlators, each delayed by a different amount. These signals are then combined using weighting attenuator factors (a_1', a_2', etc.) estimated from the channel characteristics.

The mobile user receives the signal transmitted from the serving base station through several paths with different propagation delays in a mobile radio environment. The mobile unit has the capability of combining up to three RF signals—one signal which is received directly from the serving cell-site. The other two signals may be copies of the same signals received after reflections from the structures between the cell-site transmitter and mobile receiver or may be received from neighbouring cell-sites operating in the same frequency. The cell-site receiver can combine up to four signals—the direct signal from the mobile user and three copies of the signals received after reflection from closeby buildings.

The received signal is not only corrupted by noise but is also distorted by the channel fading. For a basic receiver design, the delay-spread results in a loss of signal. The coherent RAKE receiver first de-spreads the received signals with the corresponding Walsh code and the short PN code, and then demodulates the desired signal to detect the transmitted data from each delayed-path component and combines the results. The RAKE receiver uses the direct-sequence spreading of the coded signal to separate the components of the received signal corresponding to different propagation-delay paths. It derives diversity gain from a potentially weak signal path channel.

> **Facts to Know!**
>
> The PN code is the key of each user to his or her intended signal in the receiver. Due to this, the complete set of PN sequences has to be chosen with a small cross-correlation between the several sequences in order to keep minimum adjacent channel interference.

The Walsh modulator has $\log_2 64 = 6$ data bits and transmits one of 64 orthogonal Walsh functions. The receiver demodulator tracks a finite number of independent signal paths. Assume that each signal path has a separate demodulator and their output signals are noncoherently combined. For a single path, the demodulator makes its decision by selecting the largest of the 64 output magnitudes. In the case of a finite number of independent paths, the final decision is made by the demodulator after adding the individual noncoherent correlator outputs for each of the independent signal paths.

It is possible to make a decision for each bit separately instead of making decisions for all six bits at once. This may be done by 'soft decisions' (decisions whose difference value serves as a metric for decision reliability) and by finding the difference between the highest correlation values that correspond to the two possible values (0 and 1) of the bit of interest. Soft decisions can be used in the Viterbi convolutional decoder to improve performance.

EXAMPLE 12.15 Rake receiver delay calculations

A rake receiver in a CDMA mobile phone receives a direct signal from a cell-site A located 1 km away and a reflected signal from a building 0.5 km behind the mobile. It also receives a direct signal from another cell-site B located 3 km away, as shown in Fig. 12.37.

Calculate the amount of time delay each 'finger' of the CDMA mobile receiver needs to be applied.

Solution

Distance traveled by direct signal from cell-site A, $d_1 = 1$ km (given)

Distance between cell-site A and building, $d_{11} = 1.5$ km (from the figure)

Distance between mobile and building, $d_{12} = 0.5$ km (given)

Therefore, distance traveled by the reflected signal, $d_2 = d_{11} + d_{12}$

Hence, distance traveled by the reflected signal, $d_2 = 1.5 + 0.5 = 2$ km

Distance traveled by the direct signal from cell-site B, $d_3 = 3$ km (given)

The RAKE-receiver finger receiving signals from the cell-site B which is 3 km away need not delay the signal. But the two signals (direct as well as reflected) received from the cell-site A needs to be delayed enough to allow all the three signals to be synchronised.

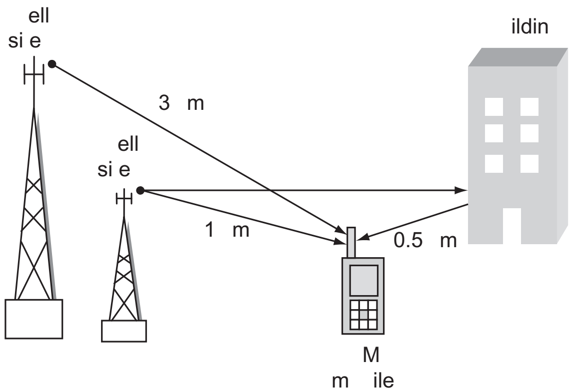

Fig. 12.37 *CDMA mobile rake receiver operation*

Step 1. *To calculate time delay for direct signal from the cell-site A, t_1*

The direct signal from the cell-site A has to be delayed by a time equal to the propagation time for a distance of (3 km – 1 km =) 2 km. That is,

Time delay for direct signal from the cell-site A, $t_1 = (2 \times 10^3)/(3 \times 10^8)$

Hence, $t_1 = 6.67\ \mu$s

Step 2. *To calculate time delay for reflected signal from the cell-site A, t_2*

The reflected signal from the cell-site A has to be delayed by a time equal to the propagation time for a distance of (3 km – 2 km =) 1 km. That is,

Time delay for reflected signal from the cell-site A,

$$t_2 = (1 \times 10^3)/(3 \times 10^8)$$

Hence, $t_2 = 3.33$ s

Usually, the delay spread is less than the information bit interval. But if the delay spread is greater than bit duration then intersymbol interference occurs. To avoid interference between detected information symbols, it is desirable that the bit duration must be kept larger than the multipath spread of the channel. This also implies that the symbol-transmission rate should be kept below the coherence bandwidth of the channel.

When consecutive signal paths arrive with delay differences greater than the chip duration or width of the autocorrelation function, the receiver output will exhibit different peaks. If the delay between two consecutive paths is significantly less than the chip duration, the two paths will merge and appear as one path equivalent to the phasor sum of the two actual paths. The fluctuations in their amplitudes and phases produce an overall fluctuation in the phasor sum, which results in fading. As the transmission bandwidth is reduced, the chip duration becomes correspondingly longer, and fewer isolated signal paths can be resolved at the receiver output.

The IS-95 CDMA system uses a chip rate of 1.25 Mcps and a symbol transmission rate of 4,800 sps (equivalent to 9,600 bps at two-bits per symbol in QPSK modulation). Therefore, it can resolve multipath components of the order of 1/1.25 Mcps = 800 ns apart. Moreover, a multipath spread of up to 1/9600 bps = 104 microseconds cannot cause intersymbol interference in the system. The multipath delay spread in the outdoor microcellular environment is of the order of several tens of microseconds. Therefore, IS-95 does not suffer from intersymbol interference in either indoor or outdoor microcellular environments. The multipath delay spread in the indoor picocell areas is of the order of several hundreds of nanoseconds. So, in an indoor picocellular environment, it is unlikely that the system resolves the multipath components and results in intersymbol interference.

If the system operates with chip durations small enough to resolve individual paths, it is desirable to design a smart receiver that can take advantage of the multiple paths to provide diversity and thus enhance the reliability of the decision on each received data symbol. In a DSSS system, a RAKE receiver optimally combines the multipath components as part of the decision process.

A RAKE receiver is capable of combining the received signal paths using any standard diversity combiner technique such as a selective, equal-gain, square-law, or maximal-ratio combiner. A maximal-ratio combining RAKE receiver that resolves all the paths and does not introduce intersymbol interference provides optimum system performance in the presence of time diversity. The maximal-ratio combiner weighs the received signal from each branch by the signal-to-noise ratio at that branch.

The received signal is passed through a tapped-delay line, and the signal at each tap is passed through a correlator similar to the one used for standard DSSS receivers. The outputs of the correlators are then brought together in a diversity combiner whose output is the estimate of the transmitted information symbol. A typical RAKE receiver structure for a DSSS system is shown in Fig. 12.38.

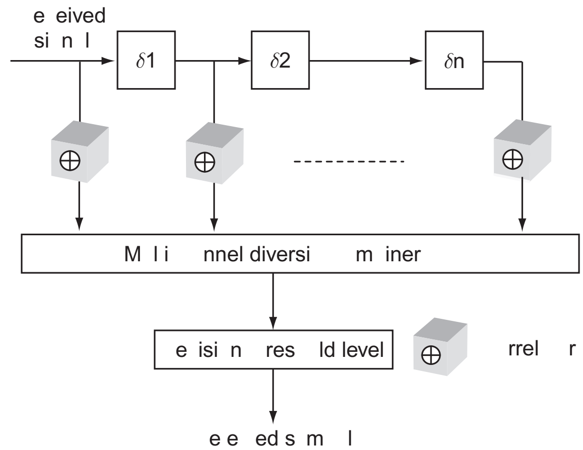

Fig. 12.38 *The RAKE receiver operation*

In the original RAKE receiver, the delays between the consecutive fingers of the RAKE receiver are fixed at half of the chip duration. This provides two samples of the overall correlation function for each chip period. For a rectangular-shaped chip pulse with triangular correlation function, there will be four samples. It is not possible to capture all the major peaks of the correlation function because the peaks are not aligned precisely at multiples of the sampling rate. But a RAKE receiver implemented with a sufficiently large number of fingers will provide a good approximation of all major peaks. An algorithm is used in digitally implemented RAKE receivers having few fingers to search for some major peaks of the correlation function and then adjust the finger locations accordingly.

In the operation of a DSSS system, if the information symbol transmission rate is less than the coherence bandwidth of the channel, multipath signals do not cause intersymbol interference. If the isolated signal paths are not utilised properly in a diversity combining receiver, the signal received from each path is wideband interference to the signals received from other paths. This leads to degraded performance of the DSSS system. The wider the transmission bandwidth, the greater is the order of implicit time diversity that can be implemented. The isolated signal paths will provide a source of implicit diversity to a DSSS receiver. It is possible to design a receiver which takes advantage of the isolated signal paths to improve system performance.

12.10 CDMA FEATURES

With IS-95, frequency reuse is available in all cells with a reuse factor of one. Each mobile user within a given cell, and mobile users in adjacent cells use the same radio-frequency channels. In essence, this is made possible because IS-95 specifies a direct-sequence spread spectrum CDMA system. Instead of dividing the allocated frequency spectrum into narrow-bandwidth channels, one for each mobile user, user data is transmitted over a very wide frequency spectrum with as many as 20 mobile subscribers simultaneously using the same carrier frequency. As such, there is no limit to the number of simultaneous active mobile subscribers in the system at any time provided the interference is within acceptable limits. As more and more active mobile subscribers use the system, there is a graceful degradation of received signal quality.

The 1S-95 specifies a different modulation and spectrum spreading technique for the forward and reverse channels. On the forward channel, the base station simultaneously transmits user data from all current mobile users in that cell by using different spreading codes for each user's transmissions. A pilot code is transmitted along with the user data at a higher power levels at all times, which provide all mobile users to use coherent detection. On the reverse channel, all mobile users respond in an asynchronous manner with a constant received signal level at the cell-site under the control of the base station.

EXAMPLE 12.16 IS-95 air-interface standards

List the important IS-95 CDMA air-interface standard specifications.

Solution

The IS-95 standard specifications are given in Table 12.10.

Table 12.10 *IS-95 CDMA standard air-interface specifications*

S. No.	*Parameter*	*Specification*
1.	Frequency band	IS-95A : 800-MHz band IS-95B : 1900-MHz band
2.	Spectral allocation	IS-95A : 50-MHz IS-95B : 120-MHz
3.	Forward and reverse channel duplex spacing	IS-95A : 45-MHz IS-95B : 80-MHz
4.	RF channel bandwidth	2.46-MHz total bandwidth; 1.23-MHz forward channel bandwidth 1.23-MHz reverse channel bandwidth
5.	Multiple-access method	Direct sequence spread spectrum (DSSS) CDMA-based accessing method
6.	Modulation scheme	*Uplink* digital OQPSK *Downlink* digital QPSK
7.	Number of channels per CDMA channel bandwidth	Total channels : 64 Voice channels : 55
8.	Voice bandwidth	8-kHz
9.	Frequency assignment	Dynamic
10.	Simultaneous voice channels per carrier	Up to 20

CDMA allows the use of all frequencies in all cells. This gives a considerable increase in system capacity. Because of the spread-spectrum system, cochannel interference simply increases the background noise level, and a considerable amount of such interference can be tolerated. However, because of the frequency reuse advantage of CDMA, CDMA offers approximately a 10-to-l capacity enhancement over standard analog AMPS and a 3-to-l capacity enhancement over USDC digital AMPS.

One CDMA RF channel has a bandwidth of 1.25 MHz, using a single carrier, modulated by a 1.2288 Mbps bit stream using QPSK. The speech coder used with IS-95 is the Qualcomm 9600-bps Code-Excited Linear Predictive (QCELP) coder. The vocoder converts an 8-kbps compressed data stream to a 9.6-kbps data stream. The design of the vocoder detects voice activity and automatically reduces the data rate to 1200 bps during silent periods. Intermediate mobile user data rates of 2400 bps and 4800 bps are also used for special purposes. Subsequently, a 14,400-bps vocoder that transmits 13.4 kbps of compressed digital voice information is also available for use with a CDMA system.

Frequency diversity is inherent in any spread-spectrum system. This is especially beneficial in a mobile radio environment due to multipath propagation. The GSM system can use a limited amount of frequency diversity by hopping among typically three discrete channels. The CDMA system uses the full 1.25-MHz bandwidth for all voice channels on a given RF channel. If a small part of this spectrum suffers a deep fade due to reflections, the only effect will be a slight increase in the error rate. The error rate can be compensated for by the error correction mechanism built into the coding of the control signals and voice data stream.

Space diversity is also built into a spread-spectrum system. Other cellular systems typically employ two receiving antennas per cell or sector at the base station to provide space diversity, but they use only one antenna at the mobile subscriber unit. Multiple receiving antennas are also used with CDMA cell-sites. It is possible to receive the signals from the same mobile phone at two or more cell-sites because the same frequencies are used in all cells. Similarly, a mobile user can receive signals from more than one cell-site. The mobile user then can make a decision about the strongest signal received and can, in fact, combine multiple received signals to obtain an even stronger signal.

CDMA uses a variable rate vocoder. Four different data rates are possible: 9600 bps, 4800 bps, 2400 bps, and 1200 bps. The full data rate of 9600 bps is used when the mobile user is talking during conversation. During speech pauses, the data rate is reduced to 1200 bps. It has been observed that each mobile user typically talks for less than fifty per cent of the time during a conversation. Theoretically, the bandwidth allocated to that mobile user can be reassigned during the pauses while the other user is still speaking.

On the forward traffic channel, data bits are repeated when the coder is running at less than the maximum rate of 9.6 kbps. For instance, if the coder operates at 1.2 kbps, as it does during pauses in speech, each block of data is transmitted eight times to get the equivalent 9.6-kbps data rate. The power in the transmit channel can be reduced under these circumstances because the error rate at the receiver depends on the energy per received data bit. Instead of reducing power, the mobile user simply transmits only one-eighth of the time, reducing interference and increasing battery life at the mobile phone.

Facts to Know!

The problem of near–far interference can be illustrated with a scenario in which a nearby and a far-off mobile user transmit with the same power, say +30 dBm. Let the respective propagation-path loss be 35 dB and 95 dB. Therefore, the received power at base station from these two mobile users will be (+30 dBm – 35 dB) = –5 dBm, and (+30 dBm – 95 dB) = –65 dBm. Thus, there is an in-band interference power, caused by the near-in mobile user, of 60 dB higher than the received signal power from the far-end mobile user. This is near–far interference due to masking effect.

12.10.1 Near–Far Effect

Using spread spectrum has a disadvantage in that the near–far effect becomes predominant. The near–far problem originates from a wide range of signal levels received at the cell-site from different mobile users located very close or far away from it within the service area of the cell. In order to prevent the signal from one mobile user (operating nearer to the cell-site) overtaking that of another mobile user (operating far away from the cell-site at the same time), strict power control at the mobile-transmitter end needs to be implemented.

Consider a cellular system in which two Mobile Users (MUs) are communicating with a base station, as illustrated in Fig. 12.39. If it is assumed that the transmission power of each MU to be the same, received signal levels at the BS from the MU_1 and MU_2 are quite different due to the difference in the propagation signal path-lengths. If r_1 r_2 then the received signal level from MU_1 will be much larger that the received signal level from MU_2 at the base station.

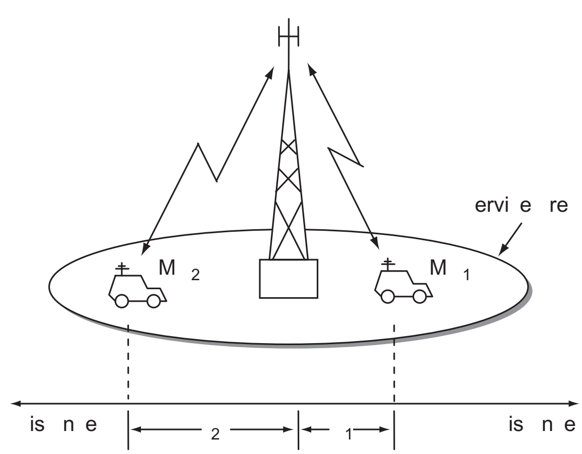

Fig. 12.39 *Illustration of near–far problem*

Assume that the MUs are using adjacent channels. Out-of-band radiation of the signal from MU_1 interferes with the signal from MU_2 in the adjacent channel. This effect, called *adjacent-channel interference*, becomes serious when the difference in the received signal strength is large. For this reason, the out-of-band radiation must be kept small. The tolerable relative adjacent channel interference level can be different depending on the system characteristics.

EXAMPLE 12.17 The near–far problem in a cellular system

In a given cellular system, the distance of a mobile user from the cell-site may range from 100 m to 10 km. Given a fourth-power propagation-path loss in a mobile radio environment, what is the expected difference in received power levels at the cell-site if both mobile users transmit at the same power level? Comment on the results obtained.

Solution

Let both mobile users transmit at P_{tm} power level. The received power level, P_{rm}, at the cell-site from the mobile user at distance '*d*' away is given by the expression:

$$P_{rm} = \beta P_{tm}/(d)^{\gamma}$$

where β is proportionality constant.

The propagation path-loss constant, $\gamma = 4$ (given)

Step 1. *To calculate the received power level,* P_{rm1}

The distance of the cell-site from Mobile 1, $d_1 = 100$ m (given)

Therefore, $P_{rm1} = \beta P_{tm}/(100)^4$

Step 2. *To calculate the received power level,* P_{rm2}

The distance of the cell-site from Mobile 2, $d_2 = 10$ km or 10,000 m (given)

Therefore, $P_{rm2} = \beta P_{tm}/(10000)^4$

Step 3. *To determine the difference in* P_{rm1} *and* P_{rm2}

The difference in received power levels at the cell-site or relative power level by these two mobile users is

$$P_{rm1}/P_{rm2} = (100 \text{ m})^4/(10{,}000 \text{ m})^4 = (0.01)^4$$

Expressed it in dB, P_{rm1}/P_{rm2} (dB) $= 10 \log (0.01)^4$

$$P_{rm1}/P_{rm2} \text{ (dB)} = 40 \log (0.01) = \quad 80 \text{ dB}$$

Comment on the results Hence, the difference in received power levels at the cell-site will be as large as 80 dB if both mobile users transmit at the same power level. This means that even in a CDMA system with a code spreading rate of 512, which is equivalent to a processing gain of 27 dB, the stronger mobile user (located nearer to the cell-site) would prevent detection of the weaker mobile user (located far away from the same cell-site). This is the near–far problem.

The near–far problem in CDMA system is quite serious and a typical example of system interference limits. The near–far problem leads to significant degradation in the quality performance of the system, especially where spread spectrum signals are multiplexed on the same frequency using low cross-correlation codes. So there is a need to implement special techniques in order to overcome the near–far problem. The possible solution is effective power control. With the use of power-control technique, the system can tolerate higher relative adjacent-channel interference levels as noticed in the above example.

12.10.2 Cell Breathing

In CDMA, the boundary of a cell or sector depends on the threshold value of E_b/N_o and is not fixed. As the number of traffic channels on the reverse channel is increased by more number of simultaneous mobile users; E_b/N_o value that is observed at a base station also increases. As a result, the hand-off boundary shifts closer to the base station. This effect is called cell breathing. In order to ensure that a proper hand-off is performed, the transmit power of the pilot channel of the base station must also be reduced so that the forward channel hand-off boundary is also maintained at the same level as the reverse channel hand-off boundary. In some cases, cell breathing can have serious implications on the system performance. The system should be properly designed either by deploying more number of cells or offloading some capacity to other carrier channels in the same service area.

12.10.3 Mobile Data over CDMA Networks

In FDMA and TDMA systems, the mobile data users may use free available channels and free available time slots respectively as they become available. In a CDMA system, all active mobile users use the entire bandwidth-time space simultaneously. So it is the signal power which has to be managed effectively. With the application of efficient power-control algorithms, the signal levels transmitted by mobile users and the base station are continuously adjusted in response to the changing locations of mobile users and the number of simultaneous mobile users on the system at any given time. In a CDMA network, the integration of data calls with voice calls is straightforward because each call accesses the channel with its unique user code. Therefore, it is possible to accommodate integration of voice and data channels without any modification to the channel access scheme. The information data rate for voice or data traffic in any one channel can be varied by a variable-rate scheme.

Since CDMA systems take advantage of the voice activity factor, the integration of bursty data with voice is very simple with the same infrastructure and mobile phones. Data services can be easily overlaid on voice traffic. If higher data rates are needed, several parallel channels for one data link can be assigned or the processing gain of the data channel can be reduced.

CDMA offers excellent security for the information transferred. In order to decode a call, it is necessary to have a spread-spectrum receiver as well as the correct de-spreading code. Since the long code is $2^{42} - 1$ bits long before it repeats and a new code is generated for every call, the probability of eavesdropping is extremely negligible.

By employing smart receiver antennas, signals from mobile users are isolated so as to reduce the near–far interference experienced in CDMA systems. Smart antennas reduce the multiple access interference, which is the primary cause for capacity limitations, by focusing the signals in one direction only. Smart antenna systems use an adaptive antenna array that can form a beam towards the target mobile users. Because the antenna is highly directive, the gain is substantially increased which enables better building penetration and longer ranges. Indeed, the natural flexibility of CDMA to accommodate a variety of data services is one of the main reasons for selection of the CDMA for 3G systems.

12.10.4 Frequency Hopping and CDMA

Frequency-Hopping Spread Spectrum (FHSS) allows the coexistence of several transmissions in the same frequency band using CDMA. Frequency hopping is another method of spreading a relatively low-data-rate digital signal over a large bandwidth. In some respects, a frequency-hopped spread spectrum technique is similar to the narrowband modulation technique except that the carrier frequency f_c of the transmitted signal is changed at regular predefined intervals. To implement CDMA, a different random hopping pattern is assigned to each mobile user. In this situation, multiuser interference can occur only when two different mobile users transmit on the same hop frequency. The codes are synchronised and the hopping patterns are selected so that two mobile users never hop to the same frequency at the same time, thereby eliminating multiple-user interference. However, the number of frequency slots limits the number of simultaneous mobile users. The advantages of FHSS are as follows:

(a) High Tolerance of Narrowband Interference Since the FH signal varies its centre frequency as per the hopping pattern, a narrowband interference signal will cause degradation only when it aligns with the carrier frequency in every hop.

(b) Relative Interference Avoidance If it is already known that a particular band of the radio spectrum contains interference, the hopping frequencies can be selected to avoid this band.

(c) Large Frequency-Hopped Bandwidths Current technology permits frequency-hopped bandwidths much greater than that can be achieved with DSSS systems.

(d) Higher Interference Tolerance With FHSS, a single interfering frequency will cause degradation only at one hop frequency, regardless of its signal strength relative to the desired signal. The interference tolerance of FHSS thus is more than that of DSSS.

The disadvantages of FHSS are as follows:

(a) Noncoherent Detection The implementation of a frequency synthesiser often limits the continuity of the signal across frequency hops. Therefore, M-ary FSK modulation techniques with noncoherent detection are employed which does not provide comparable performance as offered by coherent detection techniques.

(b) Higher Probability of Detection The higher probability of detection depends upon the spread bandwidth of the frequency-hopped signal. Since a single hop period is similar to a narrowband carrier, it is simpler to detect than a DS signal. However, if the hopping frequency is relatively fast, it is still difficult to detect the transmitted data without a prior knowledge of the hopping pattern.

12.11 THE PERFORMANCE OF A CDMA SYSTEM

The CDMA systems provide excellent immunity to interference. The CDMA system can be designed to operate at much lower S/N values since the additional channel bandwidth can be used to achieve good performance at a very low signal-to-interference value. A DSSS system offers more than ten times improvement in the system capacity. The improved performance is due to many factors such as use of the voice activity, the concept of orthogonality for multiple users on a common frequency channel, synchronise transmission for all base stations, implementation of soft hand-offs, and reduction in the level of interference.

The number of simultaneous traffic channels in the CDMA network depends on the level of total interference that can be tolerated in the system. The introduction of each additional active mobile user increases the overall level of interference to the cell-site receivers, receiving CDMA signals from several mobile users. Each mobile user adds a unique level of interference that depends on its received power level at the cell-site, timing synchronisation relative to other signals and specific cross-correlation with other CDMA signals. Thus, the CDMA system is limited by interference only. Forward error-correction coding techniques improve tolerance for interference and increase the overall system capacity.

The IS-95 standard CDMA system improves voice quality by employing a better voice coder, has resistance to multipath interference and fading, implements soft hand-offs, has less power consumption due to implementation of power control, and does not require frequency planning. Instead of defining an acceptable signal-to-interference value, it is necessary to define the signal quality parameter. Usually, this is expressed in terms of the acceptable energy per bit to total noise ratio E_b/N_o. The value of

Facts to Know!

CDMA presents unique features. In FDMA and TDMA systems, the mobile users operating in one channel are completely isolated from the mobile users operating in other channels. The cochannel interference and adjacent-channel interference are the main concerns which can be reduced by intelligent system design. In CDMA system, all mobile users are operating on the same frequency channel at the same time, causing cochannel interference. On the reverse channel, a combination of convolutional coding, spreading, and orthogonal modulation are employed to combat the effects of cochannel interference. This problem is overcome on the forward channel by employing time-synchronised orthogonal codes.

E_b/N_o is usually around 6 dB and depends on the speed of the mobile user, propagation conditions, the number of multipath signals that can be used for diversity, and so on. The value of N_o, depends on the number of interfering signals and the transmit powers of the interfering users. Consequently, power control and threshold levels decide the soft hand-off process as well as the radio coverage of a CDMA cell.

It is assumed that at the cell-site, the received signal level from each mobile user is the same and the interference seen by each receiver is modeled as Gaussian noise. The system processing gain G_p quantifies the degree of interference rejection and is defined as the ratio of channel bandwidth (B_c) to the information rate (R_b).

$$G_p = B_c/R_b \tag{12.19}$$

The system processing gain G_p is also related to input signal-to-noise ratio, $(S/N)_i$ and output signal-to-noise ratio, $(S/N)_0$ as:

$$G_p = (S/N)_0/(S/N)_i \tag{12.20}$$

It is imperative to relate $(S/N)_i$ ratio to the E_b/N_o ratio, where E_b is the energy per bit and N_o is the noise power spectral density.

$$(S/N)_i = (E_b \times R_b)/(N_o \times B_c)$$

Or, $$(S/N)_i = (E_b \times N_o\,) \times (R_b/B_c) = (E_b/N_o) \times (1/G_p)$$

Or, $$E_b/N_o = G_p \times (S/N)_i = (S/N)_0 \tag{12.21}$$

The relationship between the theoretical number of mobile users M, the processing gain G_p, and the E_b/N_o ratio is given as:

$$M = G_p/(E_b/N_o) \tag{12.22}$$

For a given bit-error probability, the actual E_b/N_o ratio depends on the radio system design and error-correction code. It may approach but never equal the theoretical calculations. The best theoretical performance that can be obtained is defined by the Shannon limit in Additive White Gaussian Noise (AWGN). As per this, $E_b/N_o = \log_e 2 = 0.69$ or –1.59 dB provides error-free communications.

EXAMPLE 12.18 Theoretical CDMA capacity

For the Shannon limit in Additive White Gaussian Noise (AWGN), calculate the theoretical number of mobile users, M in terms of processing gain in CDMA system. Also, compute M for a practical value of $E_b/N_o = 6$ dB.

Solution

Step 1. *To calculate theoretical number of mobile users, M*

As per Shannon limit in Additive White Gaussian Noise (AWGN),

$$E_b/N_o = -1.59 \text{ dB}$$

Expressing it in ratio form, E_b/N_o (in ratio) = antilog (–1.59/10) = 0.6934

We know that, $M = G_p/(E_b/N_o)$

Hence, $M = G_p/0.6934 = 1.44 \quad {}_p$

This theoretical Shannon limit shows that CDMA systems can have more users per cell than traditional narrowband systems.

Step 2. *To calculate practical number of mobile users, M*

In practice a cellular system is typically engineered such that $E_b/N_o = 6$ dB

Expressing it in ratio form, E_b/N_o (in ratio) = antilog (6/10) = 3.98

We know that, $M = G_p/(E_b/N_o)$

Hence, $M = G_p/3.98 = 0.25 \quad {}_p$

However, due to practical limitations on CDMA radio design, it is difficult to accommodate even these many users in a single cell. The upper bound theoretical capacity of an ideal noise-free CDMA channel is limited by the processing gain G_p.

In an actual system, the CDMA cell capacity is much lower than the theoretical upper-bound value. The CDMA cell capacity is affected by many other factors such as the receiver modulation performance, interference from other non-CDMA systems sharing the same frequency band, power-control accuracy, etc.

The CDMA transmissions in neighbouring cells use the same carrier frequency and therefore cause interference that can be accounted by introducing a factor ρ. This modification reduces the number of users in a cell since the interference is generated by the other mobiles in the user's cell. The typical value for ρ is taken as 0.4 – 0.55. The power–control accuracy is represented by a factor α. The practical value for α is 0.5 – 0.9. The reduction in the interference level due to voice activity can be represented by a factor 'v'. The practical range for 'v' is 0.45 – 1. If directional antennas are used rather than omnidirectional antennas at the cell site, the cell is sectorised with, say, 's' number of sectors. Each antenna used at the cell-site radiates into a sector of 360/s degrees, and this has an interference-improvement factor of 'Y'. For a 3-sector cell, the practical value of the improvement factor 'Y' is 2.55. The typical values for ρ, α, v and Y are usually taken as 0.5, 0.85, 0.6 and 2.55 respectively.

Introducing ρ, α, v and Y into Eq. (12.22), we get

$$= \ {}_p/(\ \ /\ \ ) \times 1/(1 \quad \boldsymbol{\rho}) \times \boldsymbol{\alpha} \times 1/ \quad \times \tag{12.23}$$

EXAMPLE 12.19 | Performance of a CDMA system

To illustrate the performance of a CDMA system, determine the number of mobile users that can be supported by a sector of three-sector cell of the CDMA system. The system uses an RF bandwidth of 1.25 MHz to transmit data at 9.6 kbps. Assume $E_b/N_o = 6$ dB, the interference from neighbouring cells $\rho = 50\%$, the power-control accuracy factor $\alpha = 0.85$, the voice-activity factor $v = 0.6$, and the improvement from sectorisation $Y = 2.55$.

Solution

Step 1. *To calculate processing gain, G_p*

CDMA channel bandwidth, $B_c = 1.25$ MHz or 1250 kHz (given)

Baseband data rate, $R_b = 9.6$ kbps (given)

We know that processing gain, $G_p = B_c/R_b$

Therefore, processing gain, $G_p = 1250$ KHz/9.6 kbps $= 130.2$

Step 2. *Converting E_b/N_o (dB) in ratio form*

$E_b/N_o = 6$ dB (given)

We know that E_b/N_o (dB) $= 10 \log$ E_b/N_o (ratio)

Or, E_b/N_o (ratio) $=$ Antilog E_b/N_o (dB) /10

Or, E_b/N_o (ratio) $=$ Antilog (6/10) $= 3.98$

Step 3. *To determine the number of mobile users per cell, M*

Interference factor, $\rho = 50$ or 0.5 (given)

Power control accuracy factor, $\alpha = 0.85$ (given)

Voice activity factor, $v = 0.6$ (given)

Improvement from sectorisation, $Y = 2.55$ (given)

We know that the number of mobile users per cell is given by the expression

$$M = G_p/(E_b/N_o) \times 1/(1 + \rho) \times \alpha \times 1/v \times Y$$

$$M = 130.2/(3.98) \times 1/(1 + 0.5) \times 0.85 \times 1/0.6 \times 2.55$$

$$M = 78 \text{ mobile users per cell}$$

Step 4. *To determine the number of mobile users per sector*

Number of sectors in a cell = 3 (given)

Hence, Number of mobile users/sector = 78/3 = 26

12.11.1 Advantages and Disadvantages

CDMA has a number of advantages for a cellular network.

(a) Frequency Diversity Because the transmission is spread out over a larger bandwidth, frequency-dependent transmission impairments, such as noise bursts and selective fading, have less effect on the signal.

(b) Multipath Resistance In addition to the ability of DSSS to overcome multipath fading by frequency diversity, the PN spreading codes used for CDMA exhibit low cross-correlation and low autocorrelation. Therefore, a copy of the signal that is delayed by more than one chip interval does not interfere with the dominant signal as much as in other multipath environments.

(c) Privacy Each mobile user has a unique code and spread spectrum is obtained by the use of noiselike signals, privacy is inherent.

(d) Graceful Degradation With FDMA or TDMA, a finite number of mobile users can access the system simultaneously. However, with CDMA, as more mobile users access the system simultaneously, the noise level and hence the error rate increases gradually only.

CDMA has a number of disadvantages for a cellular network:

(a) Self-Jamming Unless all of the mobile users are perfectly synchronised, the received signals from multiple mobile users will not be perfectly aligned on chip intervals. Thus, the spreading sequences of the different mobile users are not orthogonal and there is some level of cross-correlation.

(b) Near-far Problem Signals from the mobile users closer to the cell-site are received with less attenuation than signals from the mobile users far away from the cell-site. The transmissions from the more remote mobile users may be more difficult to recover. Thus, power-control techniques are very significant in a CDMA system.

(c) Soft Hand-off A smooth hand-off from one cell to the next cell requires that the mobile user acquires the new cell before it relinquishes the old one. This is referred to as a soft handoff and is more complex than the hard hand-off used in FDMA and TDMA systems.

12.11.2 Comparison of CDMA with GSM

Code Division Multiple Access (CDMA) is one of 2G digital cellular technologies, which is based on the principle of spread spectrum technique. Standardisation of CDMA technology has gone through several stages. IS-95 is the first accepted standard, followed by IS-95A, IS-95B and Cdma2000. IS-95 is mainly a single vendor (Qualcomm cdmaOne) specifications and it covers the air interface.

GSM is also a 2G digital cellular technology, which is based on TDMA technique. It is a relatively mature technology, and includes all the specifications of mobile communications. GSM standards are complete, open and proven with many equipment manufacturers and service providers.

Table 12.11 compares some of the key parameters and features of two key 2G digital cellular technologies — CDMA and GSM.

Like in GSM, a cell is normally divided into 3 sectors in CDMA, each sector covering 120 of a circular or regular hexagonal cell area. CDMA antennas are either of switched-beam system that uses multiple fixed beams

Table 12.11 *Comparison of CDMA and GSM technologies*

S. No.	*Parameter*	*CDMA (IS-95)*	*GSM*
1.	Allocated spectrum band	800 MHz; 1900 MHz	900 MHz; 1800 MHz (DCS 1800); 1900 MHz (PCS 1900)
2.	Multiple access technique	CDMA	TDMA
3.	RF and channel bandwidth	12 MHz with 1.25 MHz per carrier channel for spread spectrum	25 MHz with 200 kHz per carrier channel, 8 time slots per channel with frequency hopping
4.	Modulation technique	QPSK with DSSS	GMSK
5.	Multipath phenomenon	Used as an advantage with rake receivers	Causes fading and interference which degrade performance
6.	Use of SIM card at MS	No	Yes
7.	Data bit rate	9.6 kbps or 14.4 kbps	9.6 kbps
8.	Voice codec rate	8 kbps or 13 kbps	13 kbps
9.	SMS feature	Up to 120 text characters	Up to 160 text characters
10.	Nominal MS transmit power	2 mW to 200 mW	125 mW to 2 W
11.	Hand-off mechanism	Soft hand-off	Hard Hand-off
12.	System capacity	Flexible and better than GSM	Fixed and limited
13.	Network planning	PN code planning	Frequency planning with reuse concept

in a sector and a switch to select the best beam to receive a signal, or adaptive-antenna system in which the received signals are weighed and combined by multiple antennas to maximise the signal-to-noise ratio parameter.

In CDMA, a mobile user remains always connected to different base stations at the same time due to same frequency employed in all sectors/cells. So hand-off mechanism is based on just changing the point of attachment (base station) smoothly on move, without any break. Hence it is soft hand-off. In GSM, the attachment with the current serving cell is first broken and then a new connection is set up with another cell—hard hand-off.

12.11.3 Comparison of 2G Technologies

Table 12.12 lists some key characteristics of three of the most important second-generation digital cellular technologies.

Table 12.12 *Key characteristics of 2G cellular technologies*

S. No.	*Parameter*	*IS-54/IS-136*	*GSM*	*IS-95*
1.	Multiple access method	TDMA	TDMA	CDMA
2.	Base station Tx band	869–894 MHz; 1930-1990 MHz	935–960 MHz	869–894 MHz
3.	Mobile station Tx band	824–849 MHz; 1850–1910 MHz	890–915 MHz	824–849 MHz

(*Continued*)

Table 12.12 (*Continued*)

S. No.	*Parameter*	*IS-54/IS-136*	*GSM*	*IS-95*
4.	Duplexing technique	FDD	FDD	FDD
5.	Spacing between forward and reverse channels	45 MHz	45 MHz	45 MHz
5.	Channel bandwidth	30 kHz	200 kHz	1250 kHz
6.	Number of full-duplex RF channels	832	125	20
7.	Number of users per RF channel	3	8	Upto 35
8.	Maximum Tx power of mobile station	3 W	20 W	0.2 W
9.	Modulation scheme	π/4-DQPSK	GMSK with $B \times T_b = 0.3$	BPSK with quadrature spreading
10.	Speech coder	VSELP @ 7.95 kbps	RPE-LTP @ 13 kbps	CELP @ 13 kbps
11.	Frame size	40 ms	4.6 ms	20 ms
12.	Speech-coding bit rate	8 kbps	13 kbps	8, 4, 2, 1 kbps
13.	Channel data rate	48.6 kbps	270.833 kbps	1.2288 Mcps
14.	Error control coding	Convolutional rate	Convolutional rate	Convolutional: forward rate, reverse 1/3 rate

Key Terms

- CDMA
- 1/3 rate FEC
- Air interface
- Base station
- Closed loop power control
- Direct Sequence Spread Spectrum (DSSS)
- Forward channel
- Frequency Hopping Spread Spectrum (FHSS)
- Hand-offs
- Hard hand-off
- Interim Standard – 95 (IS-95)
- Open-loop power control
- Packet
- Power control
- Pseudo Noise (PN) code
- Rake receiver
- Reverse channel
- Roaming
- Soft hand-off
- Spread spectrum
- Spreading code

Summary

In this chapter, an overview of 2G digital cellular technology based on the CDMA technique as implemented in the IS-95 standard is presented. The air interface is by far the most complex of all similar systems, and it is not symmetrical on the forward and reverse channels unlike TDMA systems. With CDMA, the control channel and signaling information including user data is transmitted at the same frequency at the same time. It does not require any frequency planning because all cells employ the same frequency at the same time. CDMA system employs direct sequence spread spectrum technique and strong error-control coding schemes. The system implements soft hand-off, efficient power control, has resistance to multipath and fading, and consumes less power at the mobile unit. CDMA systems have demonstrated an

increase in system capacity as well as an improvement in quality of voice. However, IS-95 needs precise synchronisation of all base stations using GPS satellites, frequent power control, and typically, dual-mode mobile phones due to the limited coverage. Moreover, it still does not support high data-rate transmission. The demand for faster and high data rate transmission by the cellular users has created the need for third-generation and next-generation cellular networks that are discussed in the next chapter.

Important Equations

- ❑ $G_p = B_c/R_b$ (12.2)
- ❑ $M = G_p\ (E_b/N_o)$ (12.6)
- ❑ $\text{SINR} = 1/\ (1 + \sigma)\ (M/Q)\ (1 + N_o/I_o)$ (12.15)
- ❑ $M = B_c/R_b\ .\ 1/S_r\ .\ G_V . G_A/\sigma$ (12.16)
- ❑ $M = G_p/(E_b/N_o) \times 1/(1+\rho) \times \alpha \times 1/v \times Y$ (12.23)

Short-Answer Type Questions with Answers

A12.1 Why is DSSS preferred over FHSS technique for bursty data transmission

Bursty errors are attributable to frequency interference or fading, which are both frequency-as well as time-dependent. DSSS spreads the information in both the frequency and time domains, thus providing frequency as well as time diversity. This minimises the effects of fading and interference. FHSS-based radio systems experience occasional strong bursty errors distributed in data blocks.

A12.2 Distinguish between slow and fast FHSS systems.

In a slow-hop FHSS system, the hop rate is slower than the information bit rate. In this system, long data packets are transmitted in each hop over the wireless channel. In a fast-hop FHSS system, the frequency hopping occurs at a rate that is faster than the information bit rate. In this system, the frequency hops occur much more rapidly and in each hop a very short information data packet is transmitted, that is, the same bit is transferred using several frequencies.

A12.3 If the PN sequences are not orthogonal is CDMA still possible

It is highly desirable that various PN sequences should be mathematically orthogonal so that the individual baseband signals can be recovered exactly without any mutual interference. However, the number of possible orthogonal code sequences is limited. If the PN sequences are not orthogonal, CDMA is still possible. But there will be some mutual interference between the signals which may result in an increased noise level for all signals. As the number of non-orthogonal signals increases, the signal-to-noise ratio becomes too low and the bit-error rate too high rendering the operation of the system unacceptable.

A12.4 Why does CDMA require perfect synchronisation among all the subscribers

Using orthogonal PN codes for CDMA is highly desirable. Each subscriber is assigned a unique orthogonal code. Since the resultant waveforms are orthogonal, subscribers with different codes do not interfere with each other. Since the waveforms are orthogonal only if they are aligned in time, CDMA requires perfect synchronisation among all the subscribers.

A12.5 What is the significance of processing gain.

Processing gain is the ratio of the spread RF bandwidth to the original information bandwidth. The effect of multipath fading as well as interference can be reduced by a factor equivalent to the processing gain. In fact, it quantifies the degree of interference rejection.

A12.6 Define intracell interference and intercell interference.

Intracell interference is defined as the interference caused by other mobile subscribers operating within the same cell. Intercell interference is defined as the interference caused at the mobile subscriber in a cell due to reuse of the same CDMA channel in the neighbouring cells.

A12.7 What is meant by voice activity How can it contribute in increasing the system capacity

Voice activity refers to the fact that the calling or called mobile users speak only 40 of the time on an average in a two-way voice communication. If a mobile phone transmits only when the user speaks, then it will contribute to the interference just 40 of the total talk time. Therefore, it is expected that a system consisting mainly of voice calls and using voice activity would see an average 40 reduction in interference. This signifies that the system can accommodate more capacity proportionately.

A12.8 Why is the air-interface in CDMA systems the most complex of all other cellular systems

The air-interface in CDMA systems is not symmetrical on the forward and reverse channels as in FDMA and TDMA systems, which makes it the most complex of all other cellular systems. The CDMA channels are defined in terms of an RF frequency and code sequence. The means of applying spread spectrum modulation and error-control coding technique are different on the forward and reverse channels. Sixty-four Walsh functions are used to identify the forward channels, whereas 64 long PN codes are used for the identification of the reverse channels.

A12.9 What is the use of short PN code and long PN code

The short PN code is a pair of periodic binary PN sequences with a period of 2^{15}. These sequences are used for spreading and despreading signals as well as by multiple base stations to set different timing offsets in the code cycle. The long PN code is a sequence with a period of $2^{42}–1$. It is used for spreading signals on the reverse channel as well as for power control burst randomisation and data scrambling purpose.

A12.10 Differentiate between the PN spreading codes and the orthogonal codes.

The PN spreading codes are used to separate the transmissions between different cells, and the orthogonal codes are used to isolate the transmissions between different channels within a cell. The PN-spreading codes are *M*-sequences generated by linear feedback shift registers. The orthogonal codes is a set of 64 mutually orthogonal codes called the Walsh codes.

A12.11 Why is there need to have synchronisation in IS-95 forward channel

The PN spreading codes are used to differentiate among several base stations in the service areas employing the same frequency. The same PN sequence is used in all base stations, but the PN sequence of each base station is offset from those of other base stations by some value. For this reason, base stations in IS-95 have to be synchronised on the forward channel. Such synchronisation is achieved using GPS.

A12.12 What is the advantage of unslotted mode of paging over slotted mode

IS-95 system accommodates two modes of paging—unslotted and slotted. In the unslotted mode of operation, the mobile user is required to monitor all paging slots. However, in the slotted mode, a mobile user listens for pages only at pre-defined times during its page slots. This feature allows the mobile user to turn off its receiver for most of the time, saving the battery power and thereby increasing time between battery charging.

A12.13 How is power-control information inserted in the forward traffic channel

A power-control subchannel is continuously transmitted on the forward traffic channel by stealing equivalent bits of the traffic channel at a rate of 800 bps. The power-control information directs the mobile user to increment, decrement, or to change in its current transmitting power level. A single power-control bit (logic 0 or 1) replaces 2 data symbols to increase or decrease its mean output power level by a nominal 1 dB.

A12.14 List typical messages that can be sent over the forward traffic channel when it is used for signaling purpose.
The typical messages include order message, authentication challenge, mobile registration confirmation, alert with information, data burst, in-traffic system parameters, power control, neighbour list update, hand-off direction, shared secret data update, extended hand-off direction, and send burst DTMF message in case of three-way conference call.

A12.15 Bring out the difference in the modulation methods employed in the forward and reverse traffic channels.
The modulation method used in the forward traffic channel is QPSK and in the reverse traffic channel is offset-QPSK. The near-constant envelope properties of offset-QPSK make transmission more efficient for a power-limited mobile phone equipment. Moreover, in the forward traffic channel, the spreading codes are orthogonal generated with the Walsh matrix, whereas orthogonality of the spreading codes is not guaranteed in the reverse traffic channel.

A12.16 Why is it essential to implement power control on the reverse channel
A mobile user that transmits unnecessarily at a large power may jam or interfere with the received signals of all the other mobile users. Because all the voice channels occupy the same frequency and time slots, the received signals from multiple mobile users located anywhere within the periphery of the serving cell must all have the same received signal strength at the base station for proper detection. Since noncoherent detection is employed on the reverse channel, therefore implementation of power control is must on the reverse channel.

A12.17 Mention some disadvantages of power control.
Firstly, it may not be desirable to set transmission powers initially to higher values at the mobile phones because battery power is a limited resource that needs to be conserved. Secondly, increasing the transmitted power on one channel, irrespective of the power levels used on other channels, can cause inequality of transmission over other channels. And, power-control techniques are restricted by the physical limitations on the transmitter power levels not to exceed the specified unwanted radiations.

A12.18 What is the basis of implementing soft hand-off in CDMA systems
The implementation of soft hand-off has its basis in the near–far problem and the associated power-control mechanism. If a mobile user moves far away from a cell-site and continues to increase its transmit power to compensate for the near-far problem, there is a possibility that it may end up in an unstable situation of operation. It will also cause a lot of interference to mobile users operating in neighbouring cells. To avoid this situation and ensure that a mobile user is connected to the cell-site with the maximum received signal strength level, a soft hand-off strategy is implemented.

Self-Test Quiz

S12.1 The effect of spread spectrum modulation is that the bandwidth of the spreaded signal ________.
(a) remains constant
(b) increases significantly
(c) increases marginally
(d) decreases

S12.2 CDMA is a multiple-access strategy for wireless communications based on ________ technique.
(a) DSSS
(b) slow FHSS
(c) fast FHSS
(d) THSS

S12.3 The minimum E_b/N_o value required for proper system operation depends on
(a) the performance of coding method used
(b) bit error rate
(c) tolerance of the digitised voice signals
(d) all of the above

S12.4 A DSSS system has a 48-Mcps code rate and 4.8-kbps information data rate. The processing gain is computed to be
(a) 4.8 dB (c) 48 dB
(b) 40 dB (d) 60 dB

S12.5 Typical value of the sectorisation gain factor is taken as
(a) 6 (c) 3
(b) 4 (d) 2.5

S12.6 The number of simultaneous users that can be supported in a practical CDMA multicell system depends on the performance improvement factor given as
(a) $(G_V \times G_A)/\sigma$ (c) $(G_A \times \sigma)/G_V$
(b) $(G_V \times \sigma)/G_A$ (d) $\sigma/(G_V \times G_A)$

S12.7 Each carrier of the IS-95 standard occupies a of bandwidth.
(a) 25 kHz (c) 200 kHz
(b) 30 kHz (d) 1250 kHz

S12.8 For baseband data rate of 9.6 kbps, the user information data is spread by a factor of to a channel chip rate of 1.2288 Mcps.
(a) 64 (c) 256
(b) 128 (d) 1024

S12.9 The normalised bandwidth efficiency of the IS-95 CDMA system is
(a) 64 chips/s/Hz (c) 0.98 chips/s/Hz
(b) 1.35 chips/s/Hz (d) 0.5 chips/s/Hz

S12.10 The synchronisation channel is assigned the Walsh code
(a) W_0 (c) W_{32}
(b) W_1 (d) W_{63}

S12.11 The channel is used for sending short messages including broadcast messages.
(a) forward traffic (c) sync
(b) paging (d) pilot

S12.12 The pilot channel is transmitted at a power approximately of the total transmit power for all channels from a particular cell-site.
(a) 100 (c) 50
(b) 80 (d) 20

S12.13 The long-code generator is clocked at the chip rate and thus runs times faster than the channel bit rate.
(a) 4 (c) 64
(b) 32 (d) 128

S12.14 orthogonal Walsh codes are available for forward traffic channels.
(a) 64 (c) 7
(b) 55 (d) 4

S12.15 The CDMA reverse channel employs digital modulation technique.
(a) BPSK (c) OQPSK
(b) QPSK (d) OFDM

S12.16 If the received signal power is too low, there is probability of bit errors or frame errors.
(a) reasonably high (c) almost zero
(b) extremely low (d) nearly 100

S12.17 Once a link with the nearest base station is established, the open-loop power setting is adjusted in 1-dB increments after every by commands from the cell-site.
(a) 2 seconds
(b) 1 second
(c) 100 milliseconds
(d) 1.25 millisecond

S12.18 A mobile-assisted power control on the forward channel is implemented to reduce interference.
(a) cochannel (c) intracell
(b) intercell (d) near–far

S12.19 Generally, the soft hand-off procedure involves cell-site(s).
(a) 2 (c) 6
(b) 3 (d) several

S12.20 The value of E_b/N_o is usually which depends on the speed of the mobile user, propagation conditions, the diversity scheme used in CDMA systems.
(a) 6 dB (c) 12 dB
(b) 9 dB (d) 18 dB

Answers to Self-Test Quiz

S12.1 (b); S12.2 (a); S12.3 (d); S12.4 (b); S12.5 (d); S12.6 (a); S12.7 (d); S12.8 (b); S12.9 (c); 12.10 (c); S12.11 (b); S12.12 (d); S12.13 (c); S12.14 (b); S12.15 (c); S12.16 (a); S12.17 (d); S12.18 (b); S12.19 (d); S12.20 (a)

Review Questions

Q12.1 What is direct sequence spread spectrum concept? Explain how it works in the CDMA technology?

Q12.2 Describe the IS-95 architecture and compare it with the GSM architecture. Give at least five functions where CDMA is different from GSM.

Q12.3 Why does power control become one of the main issues for the efficient operation of CDMA?

Q12.4 Which forward channels are involved in IS-95 for power control?

Q12.5 What are the orthogonal Walsh codes?

Q12.6 A cellular system employs CDMA scheme. Is it possible to use TDMA instead of CDMA? If not, why not; and if yes what may be the potential advantages?

Q12.7 Why is the near–far problem present in CDMA, not in FDMA?

Q12.8 One approach of using Walsh code in a CDMA system is to assign a code permanently to each mobile user. List the advantages, disadvantages, or limitations of such an approach.

Q12.9 The number of Walsh codes determine the maximum number of MSs that can be served simultaneously. Typically the range of Walsh code is 28–128 bits only. Then, why are large Walsh codes not used? What are the limitations of using a large Walsh code?

Q12.10 Differentiate between hard hand-off, soft hand-off, and softer hand-off.

Analytical Problems

P12.1 A CDMA mobile measures the received signal strength from its serving cell-site as –100 dBm. What should the mobile transmitter power be set to as a first approximation?

P12.2 Calculate the time for one complete repetition of the Walsh code on the forward channel. Find the Walsh functions for a 32-bit code.

P12.3 Determine the maximum number of mobile users that can be supported in a single cell CDMA system using

(a) omnidirectional cell-site antenna and no voice-activity detection

(b) three-sectors at the cell-site and voice-activity detection factor, $v = 0.75$.

Assume the CDMA system is interference limited. The system uses an RF bandwidth of 1.25 MHz to transmit data @ 9.6 kbps, and a minimum acceptable E_b/N_0 is found to be 10 dB. Use typical values for other factors.

P12.4 The cellular spectral efficiency η in a CDMA system is defined as the total number of bits/s/Hz transmitted by all mobile users in a cell. For a QPSK modulation scheme used in CDMA, assume that the spectral efficiency of a single CDMA user is $2/Q$ bits/s/Hz, where Q is the length of the spreading code. Develop an expression for η that depends on the received I_o/N_o, SINR, and σ, if the receiver requires a specified SINR.

P12.5 Compare the capacity of 2G digital cellular systems: IS-136 TDMA, GSM, and IS-95 CDMA, assuming the allocated spectrum of 1.25 MHz in each case. Assume frequency reuse factor of $K = 4$

in IS-136 TDMA and GSM systems. For the CDMA system, assume an acceptable signal-to-interference ratio of 6 dB, data rate of 9600 bps, voice-duty cycle of 50 per cent, effective antenna separation factor of 2.75 (close to ideal 3-sector antenna), and a neighbouring cell interference factor of 1.67.

P12.6 If the channel bandwidth is 1.25 MHz, data rate is 9600 bps, and a minimum acceptable E_b/N_0 is found to be 10 dB, determine the maximum number of users that can be supported in a single-cell CDMA system using

(a) omnidirectional base station antennas and no voice-activity detection
(b) three sectors at the base station and activity detection factor of 3/8

Assume the system is interference limited.

P12.7 Show that the number of mobile users that can be supported by a CDMA system using an RF bandwidth of 1.25 MHz to transmit data at 9.6 kbps is 33 mobile users per sector. Assume $E_b/N_o = 6$ dB; the interference from neighbouring cells = 60 ; the voice-activity factor = 50 ; the power-control accuracy factor = 0.8.

P12.8 In an omnidirectional (single-cell, single-sector) CDMA cellular system, $E_b/N_o = 20$ dB is required for each user. If 100 users, each with a baseband data rate of 13 kbps, are to be accommodated,

(a) Determine the minimum channel bit rate of the spread spectrum chip sequence. (ignore voice activity considerations)
(b) Repeat Part (a) for the case where voice activity is considered and is equal to 40

P12.9 In a three-sector single-cell CDMA cellular system, $E_b/N_0 = 20$ dB is required for each user. If 100 users, each with a baseband data rate of 13 kbps, are to be accommodated,

(a) determine the minimum channel bit rate of the spread spectrum chip sequence. (ignore voice activity considerations)
(b) Repeat Part (a) for the case where voice activity is considered and is equal to 40

P12.10 For the IS-95 system, a chip rate of 1.2288 Mcps is specified for the data rate of 9.6 kbps. E_b/N_o is taken as 6.8 dB. Estimate the average number of mobile users that can be supported by the system in a sector of the 3-sector cell. Assume the interference from neighbouring cells = 50 ; the voice-activity factor = 60 ; the power-control accuracy factor = 0.85; and the improvement factor from sectorisation = 2.55.

P12.11 A total of 36 equal-power mobile terminals share a frequency band through a CDMA system. Each mobile terminal transmits information at 9.6 kbps with a DSSS BPSK modulated signal which has E_b/N_o of 6.8 dB corresponding to bit error probability of 0.001. Calculate the processing gain and the minimum chip rate of the spreading PN code in order to maintain the specified E_b/N_o value. Assume the interference factor from neighbouring cells = 60 ; the voice activity factor = 50 ; the power-control accuracy factor = 0.8.

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13

2G digital cellular technologies are based on circuit-switched data modems that limit data users to a single circuit-switched voice channel, with a limited provision for data transmissions up to 10 kbps rate. The 2.5G technologys allow existing 2G equipments to support higher data-rate transmissions for web browsing, e-mail traffic, mobile commerce, and location-based mobile services. The evolution from 2G to 2.5G cellular technologies such as GPRS, EDGE, and cdmaOne are discussed in this chapter. In order to develop an international standard, to provide enhanced mobile data services at higher data rate and seamlessly integrating data and voice services, an overview of 3G cellular technologies including UMTS, W-CDMA air interface, and Cdma2000 systems along with future trends in wireless technologies is covered here.

3G Digital Cellular Technology

13.1 2.5G TDMA EVOLUTION PATH

The TDMA evolution path originating from GSM and IS-136 standards include High Speed Circuit Switched Data (HSCSD), General Packet Radio Service (GPRS), and Enhanced Data Rates for GSM Evolution (EDGE). These upgraded 2.5G digital cellular standards provide significant improvements in user data transmission rates from a meagre 9.6 kbps in GSM to the theoretical 57.6 kbps on HSCSD network, 171.2 kbps on GPRS network, and 384 kbps on EDGE networks. Figure 13.1 illustrates the digital cellular upgrade paths from 2G digital cellular GSM and IS-136 to various 2.5G digital cellular technologies.

The HSCSD standard is primarily an extension of the GSM standard. It uses the same air interface, system configuration, and hardware infrastructure. It also uses circuit-switched technique that allows individual mobile data users to utilise consecutive time slots to achieve higher speed data access on the existing GSM network. The modification required in GSM network to function as HSCSD network also is a software upgrade at the base station. The HSCSD modifies the error-control coding algorithms and increases the application data rate to 14.4 kbps. By using up to four consecutive time slots, HSCSD is able to provide a gross transmission data rate of up to 57.6 kbps (4×14.4 kbps) to individual mobile data users. HSCSD is suitable for dedicated streaming Internet access or real-time interactive web. GSM only mobile phones do not work in HSCSD networks. Dual-mode user mobile phones provide 57.6 kbps on HSCSD networks, and 9.6 kbps on GSM networks.

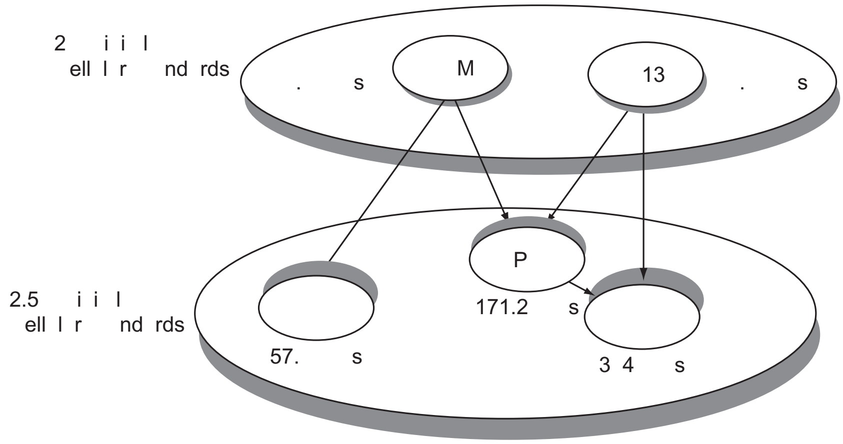

Fig. 13.1 *2.5G TDMA evolution path*

13.2 GPRS TECHNOLOGY

The General Packet Radio Service (GPRS) is an enhancement of the GSM system that makes the use of packet-switched data services possible to the mobile users. GPRS is an overlay on top of the GSM or IS-136 physical layer and network entities. It extends data capabilities of GSM and provides connection to external packet data networks through the GSM infrastructure with short access time. GPRS is well suited for non-real time Internet usage such as the retrieval of email, faxes, and asymmetric web browsing, where the user downloads much more data than it uploads on the Internet.

> **Facts to Know!**
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> GPRS allows negotiation of QoS profiles using the parameters such as throughput, service priority, reliability and delay between the mobile user and the network for each session.

From a high level, GPRS can be thought of as an overlay network onto a second-generation GSM network. This data overlay network provides packet data transport at rates from 9.6 kbps to 171 kbps. Additionally, multiple users can share the same air-interface resources. GPRS attempts to reuse the existing GSM network elements as much as possible, but in order to effectively build a packet-based mobile cellular network, some new network elements, interfaces, and protocols that handle packet traffic are required. Therefore, GPRS requires modifications to numerous network elements.

GPRS is a new non-voice value added service to the mobile users. It supplements Circuit Switched Data and Short Message Service. When all eight time slots of a GSM radio channel are dedicated to GPRS, an individual user is able to achieve as much as 171.2 kbps (8×21.4 kbps of raw uncoded data throughput). The GPRS provides short connection set-up times, virtual connections, and practical data rates up to 115 kbps for each mobile data user. In the core network, beside the transportation of user data, a separate signaling network, based on the Signalling System 7 (SS7) is used to support the user mobility and to access the various databases of the GSM network.

GPRS uses exactly the same physical radio channels as GSM, and only new logical GPRS radio channels are defined. Allocation of these channels is flexible: from one to eight time slots can be allocated per TDMA frame. Allocation to circuit-switched services and packet-switched GPRS services is done dynamically according to a capacity on demand principle. GPRS mobile users are automatically instructed to use dedicated GPRS radio channels and particular time slots for always on access to the network. This means that the capacity allocation for GPRS is based on the actual need for packet transfers.

GPRS allows more efficient packet data transfer compared to GSM data services. The principle is that a mobile user can be constantly connected to the network without occupying any radio resource until a data packet has to be transferred. When a packet is to be transferred, a temporary channel is assigned to the mobile user. The channel is quickly released after completion of data transfer. GPRS allows a single user to use more than one time slot and many mobile users to share the same time slot. It uses an error detection and retransmission scheme to ensure that data packets are correctly delivered to the mobile user.

Facts to Know!

For GPRS, no dial-up modem connection is necessary. It offers fast connection set-up mechanism which gives a perception of being always connected to GPRS users.

13.2.1 Key Features

(a) Packet Switching GPRS involves overlaying a packet-based air interface on the existing circuit-switched GSM network. This gives the mobile user an option to use a packet-based data service. The information data is split into separate but related packets before being transmitted and reassembled at the receiving end.

(b) Mobile Data over GPRS Network The GPRS packet data network uses the air interface and the infrastructure of the GSM network to provide a mobile packet data service that can support data rates of up to 171.2 kbps. GPRS uses the same digital modulation technique and physical packet format as GSM. The data packets are routed to the packet-switched data networks rather than being switched through the PSTN.

(c) Spectrum Efficiency Packet switching simply means that GPRS radio resources are used only when mobile users are actually sending or receiving data. Instead of dedicating a radio channel to a mobile data user for a fixed period of time, the available radio resource can be concurrently shared between several mobile users. This enhances the spectrum efficiency because many GPRS mobile users can potentially share the same bandwidth and be served together.

(d) Capacity Enhancement The actual number of simultaneous mobile data users supported by the system depends on the application and how much data is being transferred. Because of high spectrum efficiency, it may not be required to build idle capacity for use during peak hours, and allow using network resources in a dynamic and flexible way. GPRS reduces SMS centre and signalling channel loading by migrating some traffic using the GPRS/SMS interconnect supported by the GPRS networks.

(e) Reservation Protocol in GPRS A single 200-kHz carrier channel in GSM has eight time slots. Each time slot is capable of carrying data at 9.6 kbps (standard), 14.4 kbps (enhanced), or 21.4 kbps (if forward error correction is not employed). Thus, the raw data rate can be as high as 8×21.4 kbps $= 171.2$ kbps. The same time slots can be reserved for data access using slotted ALOHA reservation protocol medium access technique. In the contention stage, a slotted ALOHA random access technique is used to transmit reservation requests. The base station transmits a notification to the mobile user indicating the allocation of time slots for an uplink or downlink data transmission. Then the mobile user can transfer data on the allocated time slots without contention.

(f) Security Features The security offered by GPRS is similar to that offered by GSM and includes the key security parameters such as anonymity, authentication and confidentiality for user data and signaling.

Facts to Know!

GPRS uses a ciphering algorithm optimised for packet data transmission. Its security functionality is similar to the GSM security.

13.2.2 GPRS Network Architecture

The main objective of the GPRS is to provide an access to standard data networks such as X.25 and TCP/IP protocol networks. These data networks consider GPRS just as a normal sub-network. In IP-based internetworks, routers are used to separate different sub-networks from each other. Implementation of GPRS requires new routers and Internet gateways at the base station with no change in RF hardware, along with new software that redefines the base station air interface standard for GPRS time slots.

In the core network, the existing MSCs are based upon circuit-switched technology, and they cannot handle packet traffic. Thus, GPRS architecture contains new network components to the GSM network. The main new network components are the Serving GPRS Support Node (SGSN) and the Gateway GPRS Support Node (GGSN). Also, a new network component located on the connection to the Inter-PLMN backbone network is the Border Gateway (BG) needed to provide network security. A separate IP network acts as an interface between GSM and packet data networks. GPRS mobile data users generate data traffic on top of GSM speech traffic and thus radio transmission and network capacity is needed throughout the BSS. In addition, some of the GSM network components are upgraded by new software (BTSs, HLR) or hardware (BSC) to meet the requirements set by GPRS technology. Figure 13.2 shows GPRS network architecture.

The serving GPRS support node provides packet routing to and from the SGSN service area for all mobile data users in that service area. The GGSN acts as a gateway between the GPRS network and Public Data Networks such as IP and X.25. GGSNs also connect to other GPRS networks to facilitate GPRS roaming. The GGSN functions as a router and hides the GPRS specific features from the external packet data networks. In other words, mobile data users appear as normal hosts (identified by unique IP addresses) belonging to a sub-network (GPRS sub network) that is served by a router (GGSN).

In addition to adding multiple GPRS nodes and a GPRS backbone, there are certain technical changes that need to be carried out to a GSM network to implement a GPRS service. These include a new air interface

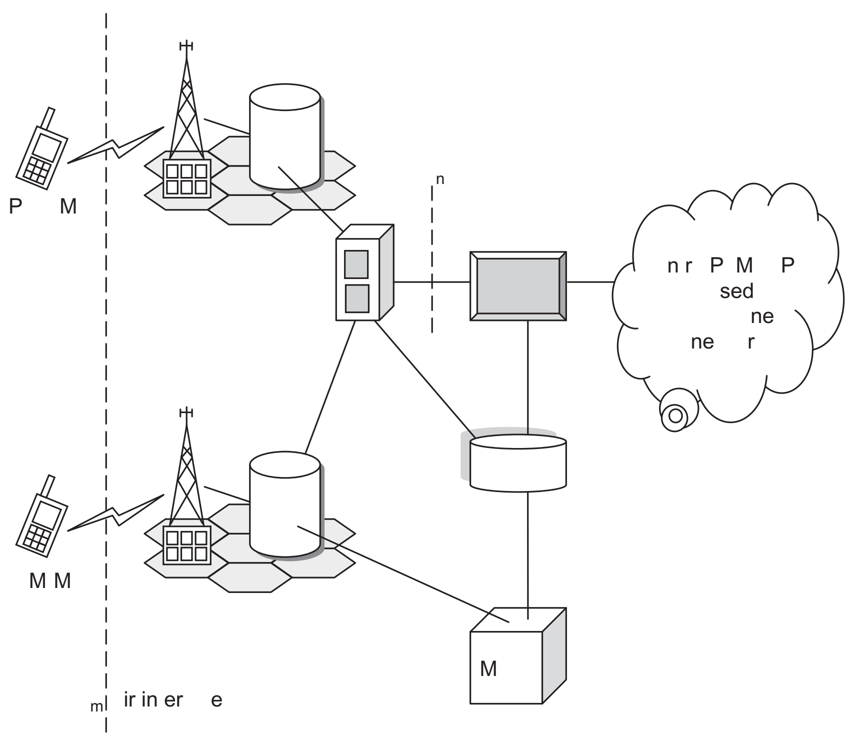

Fig. 13.2 *GPRS network architecture*

for packet traffic, the addition of packet control units at BSC, new GPRS specific signaling, new security features, mobility management to locate the GPRS mobile data users.

GPRS-MS New GPRS-enabled mobile subscriber (GPRS-MS) phones are required because existing GSM phones do not handle the enhanced air interface, nor do they have the ability to packetise traffic directly. A variety of GPRS-MS phones exist, including a high-speed version of current phones to support high-speed data access, a new kind of PDA device with an embedded GSM phone, and PC Cards for laptop computers. All these GPRS-MS phones are backward compatible with GSM systems for making voice calls using GSM network. The GPRS mobile phones are of three types:

(a) *Class A* mobile phones support simultaneously GPRS services as well as other GSM services such as voice and SMS. This support includes simultaneous attach, activation, monitor, and traffic. In the presence of circuit-switched services, GPRS virtual circuits will be placed on hold or on busy rather than being cleared.

(b) *Class B* mobile phone can monitor all GSM and GPRS channels simultaneously, but operate either GPRS or GSM at a time. Therefore, a Class B MS can support simultaneous attach, activation, and monitor, but not simultaneous traffic. As with Class A, the GPRS virtual circuits will not be closed down when circuit-switched traffic is present. Instead, they will be switched to busy or hold mode. Thus, users can make or receive calls on either a switched or a packet call-type sequentially, but not simultaneously.

(c) *Class C* mobile phones operate only GPRS service. It supports only non-simultaneous attach. The user must select which service to connect to. Therefore, a Class C MS can make or receive calls from only default or manually selected service. The service that is not selected is not accessible. Finally, the GPRS specifications state that support of SMS is optional for Class C mobile phones.

Serving GPRS Support Node (SGSN) The SGSN is connected to the BSC and is the service access point to the GPRS network for the GPRS mobile users. SGSN is analogous to MSC used in GSM networks. It can be viewed as a packet-switched MSC. It delivers packets to mobile subscribers (MSs) within its service area. SGSNs send queries to Home Location Registers (HLRs) to obtain profile data of GPRS subscribers. SGSNs detect new GPRS MSs in a given service area, process registration of new mobile subscribers, and keep a record of their location inside a given area. Therefore, the SGSN performs mobility management functions such as mobile subscriber attach/detach and location management.

The SGSN may route its packets over different GGSNs to reach different packet data networks. The main function of SGSN is to handle authentication, data compression and ciphering, and mobility management features. When the authentication is successful, the SGSN handles the registration of a mobile user to the GPRS network and takes care of its mobility management. When the mobile user wants to send (or receive) data to (from) external packet data networks, the SGSN relays the data between the SGSN and relevant GGSN (and vice versa). SGSN collects charging information and forwards it to the Charging Gateway.

Facts to Know!

The SGSN performs authentication and cipher-setting procedures based on keys, algorithms and criteria as used in GSM.

Gateway GPRS Support Node (GGSN) The GGSN is connected to the external packet data networks like the Internet and the X.25. For external packet data networks, the GGSN is just like any other router to a sub-network because the GGSN hides the GPRS infrastructure from the external packet data networks. When the GGSN receives data addressed to a specific mobile data user, it checks if the called address is active. If it is active, then the GGSN forwards the data packets to the serving SGSN. If the called address is inactive, the received data packets are discarded. The mobile originated data packets are routed to the desired network by the GGSN.

> **Facts to Know!**
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> GGSN also performs authentication and charging functions related to data transfer. It stores the current SGSN address and user profile in its location register.

The GGSN tracks the mobile user in association with SGSN. GGSNs are used as interfaces to external IP networks such as the public Internet, GPRS services of other cellular service providers, or enterprise intranets. GGSNs maintain routing information that is necessary to tunnel the Protocol Data Units (PDUs) to the SGSNs that service particular MSs. Other functions include network and subscriber screening and address mapping. One or more GGSNs may be available in GPRS network to support multiple SGSNs.

> **Facts to Know!**
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> In essence, in the GPRS backbone there is an IP/X.25-over-GTP (GPRS tunneling protocol) –over-UDP/TCP-over-IP transport architecture. Ethernet, ISDN, or ATM-based protocols may be used in the physical layer in the IP backbone.

Backbone Networks Backbone networks are IP-based networks which are used for Intra-PLMN and Inter-PLMN communications within or between GPRS networks. There are two types of backbone networks in the GPRS system. The Intra-PLMN backbone network allows the SGSNs and GGSNs of one service provider to communicate with each other via G_n interface, and the Inter-PLMN backbone network allows the SGSNs and GGSNs of different service providers to communicate with one another via G_p interface. The Intra-PLMN backbone network uses a private IP network to ensure the performance and security of the GPRS system. The Inter-PLMN backbone network is an IP network based on the public Internet or a private IP network using leased lines, selected by roaming agreements between service providers.

IP Network All the interconnected GPRS backbone networks comprise one private IP network in which the address allocation is properly coordinated. A private IP network uses private IP addresses that cannot be accessed from external networks. Connections are established directly using the standard protocol (X.25 or IP) addresses of the external networks. The GPRS mobile user can use one to eight time slots over the air-interface depending on the class of the mobile user and the type of GSM traffic at the same time. The GPRS traffic channels are dynamically allocated to GPRS mobile users for transfer of data packets.

The resource allocation in the GPRS network is dynamic and dependent on the resource availability and user demand. Packets can also be sent on idle time between voice calls, which results in more efficient utilisation of the air-interface traffic channels. In the GPRS network, uplink and downlink time slots are reserved separately. This enables to have multislot mobile users with various uplink and downlink capabilities.

13.2.3 GPRS Signalling

In GPRS, the radio resources are reserved only for the duration of the actual data transfer. GPRS can support many more users in a bursty manner with the same resources as provided by HSCSD system. This enables volume-based charging which is more suitable for bursty data applications. GPRS uses a completely redefined air interface in order to handle packet-data access while retaining the original modulation formats specified in the 2G TDMA standards. Four new channel-coding schemes are introduced which are optimised for packet data transfer. The connection establishment time in GPRS is much shorter than in the circuit-switched data, which enables instantaneous access to the Internet. The possibility to use several time slots simultaneously by a mobile data user enables considerably higher bit rates.

A Media Access Control (MAC) utilises the resources of the physical radio interface and provides a service to the GPRS Logical Link Control (LLC) protocol between the MS and the serving GSN (SGSN). LLC is a modification of a High-Level Data Link Control (HDLC)-based Radio Link Protocol (RLP) with variable frame size. The two most important features offered by LLC are the support of point-to-multipoint addressing and the control of data frame retransmission.

GPRS provides a standard interface for the network layer. The GPRS network encapsulates all data network protocols into its own encapsulation protocol, called the GPRS Tunneling Protocol (GTP). This is done to ensure security in the backbone network and to simplify the routing mechanism and the delivery of data over the GPRS network.

Facts to Know!

The location and mobility management involves tracking of the location of the mobile user as it moves. This is necessary to establish the communication link for voice or data transfer as well as enabling the network to route packets to the mobile user accordingly. The SGSN and the GGSN act as visiting and home databases in GPRS respectively.

13.2.4 Location Management in GPRS

Location management depends on three states of the mobile user in which it can operate at any time. In the switched off or IDLE state, the mobile user is not reachable by the network, and the contexts of packet data protocol are deleted. However, appropriate action need to be taken depending on the service requested. For example, short messages may be stored on a server for later delivery to the mobile user. In the STANDBY state, movement across routing areas is updated to the SGSN but not updated across cells. In the READY state, every movement of the mobile user is indicated to the SGSN. Location management includes location updates, paging, and location information dissemination.

Location updates are the messages sent by the MS regarding its changing point of access to the fixed network. In order to deliver incoming messages to the MS, the network will have to page the MS in the possible group of cells. If the MS updates its location quite often, it consumes battery power and wastes the resources. If the MS updates infrequently, a systemwide paging is needed which is also a waste of resources. Location information dissemination refers to the procedures required to store and distribute the location information related to the mobile users serviced by the GPRS network. In the intra-SGSN Routing Area update, the SGSN already has the user profile and packet-data protocol context. The HLR need not be updated. A new temporary mobile subscriber identity is issued.

Facts to Know!

In an inter-SGSN routing area update, the new routing area is serviced by a new SGSN. The new SGSN requests the old SGSN to send the packet-data protocol contexts of the mobile user. The new SGSN informs the home GGSN, the GPRS register, and other GGSNs about the user's new routing context. The location is updated with a routing update procedure.

An intra-SGSN routing area update is also possible when the same SGSN serves the new routing area. The new SGSN updates the HLR. The new SGSN requests the old SGSN to provide the routing context (GGSN address and tunnelling information) of the mobile user. The new SGSN then updates the GGSN of the home network with the new SGSN address and the routing context information.

When a mobile user changes a routing area, it sends a routing area update request containing the cell identity and the identity of previous routing area, to the new SGSN.

13.2.5 Mobility Management in GPRS

The operation of the GPRS is partly independent of the GSM network. However, some procedures share the network elements with current GSM functions to increase efficiency and to make optimum use of free GSM resources such as unallocated time slots. Data is transmitted between a MS and the GPRS network only when the MS is in the active state. In the active state, the SGSN knows the cell location of the MS. However, in the standby state, the location of the MS is known only as to which routing area, consisting of one or more cells within a GSM location area, it is in. When the SGSN sends a packet to a standby MS, the MS must be paged. Because the SGSN knows the routing area in which the MS is located, a packet paging message is sent to that routing area. After receiving the packet paging message, the MS gives its cell location to the SGSN to establish the active state.

Facts to Know!

As a mobile station moves from one cell to another cell, mobility management functions are used to track its location within each designated service area. SGSNs communicate with each other to update the location of MSs.

Packet transmission to an active MS is initiated by packet paging to notify the MS of an incoming data packet. The data transmission proceeds immediately after packet paging through the channel indicated by the paging message. The purpose of the packet paging message is to simplify the process of receiving packets. The MS has to listen to only the packet paging messages, instead of all the data packets in the down-link channels, reducing battery use significantly.

When an MS has a packet to be transmitted, access to the uplink channel is needed. The uplink channel is shared by a number of MSs, and its use is allocated by a BSS. The MS requests use of the channel in a packet random access message. The transmission of the packet random access message follows slotted ALOHA procedures. The BSS allocates an unused channel to the MS and sends a packet-access grant message in reply to the packet random access message. The allocation of the channel having one or multiple time slots is included in the packet-access grant message. The data is transmitted on the reserved channels.

Mobile management deals with hand-off initiation. The mobile user selects the cell based on the information available in BCCH. There is, however, an option available for mobile assisted hand-off procedure also. Power control and security mechanisms are implemented in the similar way as done in GSM. The ciphering algorithm is used for MS-SGSN encryption. Mobility management within GPRS builds on the mechanisms as used in existing GSM networks; as an MS moves from one area to another, its location can be tracked within each mobile network.

The SGSNs communicate with each other and update the user location. The MS profiles are preserved in the Visitor Location Registers (VLRs) that are accessible by the SGSNs via the local GSM MSC.

A logical link is established and maintained between the MS and the SGSN in each mobile network. At the end of transmission or when a MS moves out of the area of a specific SGSN, the logical link is released and the resources associated with it can be reallocated.

Facts to Know!

Web browsing format language is called Wireless Applications Protocol (WAP) that allows standard web pages to be viewed in a compressed format specifically designed for small, portable handheld cellphones.

13.2.6 GPRS Applications and Services

A wide range of corporate and consumer applications are enabled by non-voice or data mobile services such as SMS. Typical GPRS applications include all normal GSM services but in a more efficient and faster way. It also supports all those services available over the Internet like e-mail, and web browsing format language. It also supports a completely new set of user services such as enhanced short messages, wireless imaging with instant pictures, video services, document and information sharing.

The benefits available to service providers include efficient use of scarce radio spectrum resources, fast and flexible implementation with GSM coverage, low investment cost, GPRS evolution to incorporate higher bandwidth and new technology, and IP mainstream bearer characteristics. The benefits offered to mobile data users include wide area coverage, fast access time, and no need for frequent log on/log off, higher speed, high-level security, availability of many different types of mobile phone devices facilitating a broad range of new applications.

13.2.7 Challenges and Issues in GPRS

GPRS is an important new enabling mobile data service that offers a major improvement in spectrum efficiency; capability and functionality compared with other mobile data services. However, there are some limitations with GPRS network, which can be summarized as follows:

(a) Limited Cell Capacity For all mobile users, GPRS does impact a network's existing cell capacity. There are only limited radio resources that can be deployed for different uses—use for one purpose precludes simultaneous use for another purpose. For example, voice and GPRS calls both use the same network resources. However, GPRS manages channel allocation dynamically and allows a reduction in peak time signalling channel loading by sending short messages over GPRS channels. The extent of the impact depends upon the number of time slots, if any, that are reserved for exclusive use of GPRS.

(b) Lower Data Speeds Achieving the theoretical maximum GPRS data transmission data rate of 172.2 kbps would require a single mobile user utilising all eight time slots of a frame without any error protection. The system may not allow all time slots to be used by a single GPRS user. Additionally, the initial GPRS mobile phones are expected to be severely limited—supporting only one, two or at the most three time slots. The bandwidth available to a GPRS user and hence the practical data rates will therefore be severely limited.

(c) Suboptimal Modulation GPRS is based on the GMSK modulation technique. EDGE is based on 8-PSK modulation scheme that allows a much higher bit rate across the air interface. Since 8-PSK is also used for UMTS, there will be a need to incorporate it at some stage to make the transition to 3G systems.

(d) Transit Delays GPRS packets are sent in different directions to reach the called mobile user. This has potential for one or some of the data packets to be lost or corrupted during the data transmission over the radio link. Although GPRS standards incorporate data integrity and retransmission strategies, this results in potential transit delays.

(e) No Store and Forward There is no storage mechanism incorporated into the GPRS standard, apart from the incorporation of interconnection links between SMS and GPRS.

(f) Tariff and Billing GPRS is essentially a packet-switching overlay on a circuit switching network. The GPRS specifications stipulate the minimum charging information that must be collected in the initial service subscription stage. A GPRS network needs to be able to count packets to charging customers for the volume of packets they send and receive.

13.3 EDGE TECHNOLOGY

EDGE, Enhanced Data Rates for GSM Evolution, is a more advanced upgrade for both GSM and IS-136 networks as well as 2.5G GPRS network. It is sometimes referred to as 2.5G+ and standardised as UWC-136 (Universal Wireless Communications Consortium) by IMT-2000. EDGE represents the final evolution of data communications within the GSM and IS-136 standards.

An EDGE system can be deployed in a 600 kHz channel bandwidth also. This is still less than half of that needed by even narrowband CDMA systems. EDGE allows higher data rates than GSM does. This is accomplished by using a new digital modulation format, 8-PSK (octal phase shift keying), which is used in addition to GSM's standard GMSK modulation. EDGE allows for nine different air-interface formats, known as multiple Modulation and Coding Schemes (MCS), with varying degrees of error control protection. Each MCS state may use either GMSK (low data rate) or 8-PSK (high data rate) modulation technique for network access, depending on the instantaneous user demands and the operating conditions. Because 8-PSK is more susceptible to errors than GMSK, EDGE has nine different MCSs (Modulation and Coding Schemes), each designed for a different quality connection. They differ in how much forward error correction is needed and in whether 8–PSK can be used at all for noisy wireless communication links. In that case, it automatically uses GMSK.

Facts to Know!

EDGE system inherits almost all its main features from GSM and GPRS, including the eight-time-slots TDMA frame structure and even the slot period of 0.577 ms. EDGE's channel bandwidth is 200 kHz, same as that of GSM.

Each mobile user connection may adaptively determine the best MCS setting depending upon the user data access requirements and radio propagation conditions. This adaptive capability to select the best air interface is referred to as incremental redundancy. Packets are transmitted first with maximum error protection and maximum throughput. Subsequent packets are transmitted with less error protection (by employing punctured convolutional codes) and reduced throughput, until the communication link has an unacceptable delay or outage.

Facts to Know!

EDGE is a new radio interface technology with enhanced modulation, and increases the GPRS data rates by up to three times. However, the radio coverage area is smaller in EDGE than in GPRS because of the higher data rates and relaxed error-control mechanism.

The type of modulation and amount of FEC necessary for each MCS type is shown in Table 13.1, along with the data capacity available from a single slot and from the multi time slots.

Table 13.1 *Modulation and coding Schemes (MCS) of EDGE*

MCS	*Modulation type*	*Slot data rate*	*EC coding*	*Channel data rate*
MCS-1	GMSK	8.8 kbps	143%	70.4 kbps
MCS-2	GMSK	11.2 kbps	91%	89.6 kbps
MCS-3	GMSK	14.8 kbps	45%	118.4 kbps
MCS-4	GMSK	17.6 kbps	22%	140.8 kbps
MCS-5	8-PSK	22.4 kbps	187%	179.2 kbps
MCS-6	8-PSK	29.6 kbps	117%	236.8 kbps
MCS-7	8-PSK	44.8 kbps	43%	358.4 kbps
MCS-8	8-PSK	54.4 kbps	18%	435.2 kbps
MCS-9	8-PSK	59.2 kbps	8%	473.6 kbps

Facts to Know!

EDGE could be directly implemented in an existing GSM system so as to offer every mobile user a landline-quality voice or data connection, but the system would also like to serve GPRS users along with. For this reason, the high-speed data services deployed over EDGE are sometimes referred to as EGPRS (Extended GPRS). For those mobile users not having an EDGE-equipped mobile phone, it is fully compatible with existing GSM as well as GPRS.

When EDGE uses 8-PSK modulation without any error protection, and all eight times slots of a GSM radio channel are dedicated to a single mobile data user, a raw peak throughput data rate of ($3 \times 8 \times 22.8$ kbps =) 547.2 kbps can be provided. Practically, raw data rates of up to about 384 kbps for a single dedicated mobile user on a single GSM channel can be achieved due to the slotting schemes used in EDGE, network contention issues and error-control coding requirements. This implies that EDGE air interface channel coding offers approximately (384 kbps divided by 8 =) 48 kbps per traffic channel. However, it is possible to provide up to several megabits per second of data throughput to individual mobile users by combining the capacity of multicarrier transmissions.

To support EDGE, the cellular service provider has to upgrade its transceivers with software updation as the modulation scheme changes. EDGE uses 8-PSK modulation at higher data rates and standard GMSK modulation at lower data rates. There is an option to use either ordinary GSM voice or be packetised and carried as data.

13.4 2.5G CDMAONE CELLULAR TECHNOLOGY

Based on CDMA air-interface and multiple access technology, there is a single upgrade path from 2G digital standard to 3G digital standards. The interim upgrade solution from 2G (IS-95) to 3G (Cdma2000) is called IS-95B or 2.5G cdmaOne. Like GPRS, IS-95B cdmaOne standard provides circuit-switched and high-speed packet-switched data access on a common CDMA radio channel.

Each IS-95 CDMA radio channel supports up to 64 different user channels with a throughput of 9.6 kbps (rate set 1) and 14.4 kbps (rate set 2 as specified in IS-95A). The cdmaOne technology allows a dedicated mobile user to combine up to eight different user channels simultaneously and in parallel for an instantaneous throughput of (8×14.4 kbps =) 115.2 kbps per mobile user. However, due to the slotting techniques of the air interface and other factors, practically about 64 kbps throughput is available to a single mobile user in cdmaOne system.

The cdmaOne cellular technology offers numerous benefits to the cellular service providers and their mobile users, such as capacity increases of about 4 to 5 times as compared to that of a GSM digital cellular system; simplified system planning through the use of the same frequency in every sector of every cell; improved radio coverage characteristics allowing for the possibility of fewer cell-sites; availability of bandwidth on demand; improved call quality; enhanced privacy; and increased talk time for handheld mobile phones.

> **Facts to Know!**
>
> cdmaOne requires new software in base station controller, without the need of changing any hardware in IS-95A. New cdmaOne mobile phones are backward compatible and work with cdmaOne system at 64 kbps or 14.4 kbps data rate as well as with IS-95A system at 14.4 kbps data rate.

In IS-95 and IS-95A systems, the mobile users informs the measured link quality parameters to the mobile switching centre through the serving base station on frequent regular basis. This helps the mobile switching centre to initiate a soft-handoff between the mobile user and candidate base stations at appropriate time. However, in IS-95B cdmaOne system, the mobile users are capable of searching different radio channels in the network without getting any commands from the base station and take hand-off decision. The mobile users can rapidly switch over to different base stations to maintain communication link quality.

13.5 NEED OF 3G CELLULAR NETWORKS

The mobile phone has the potential to become a generic platform for, or gateway to, the complete range of communication services which may include voice, data, video, multimedia, and internet browsing. There is a need for more sophisticated services such as Internet on mobile phone, and convergence of fixed telecommunication network, computer network, the digitisation of enterprise media and wireless cellular networks on a single miniaturised user device. Third-generation digital wireless cellular networks and standards have the capability to deliver transmission data rates of up to 384 kbps for wide-area coverage, and 2 Mbps for indoor or fixed applications, thereby maximising the efficiency of available radio spectrum.

There are several areas in which capability enhancements to the present 2G or 2.5G digital cellular networks are needed. These include higher system capacity, high-data rate transfer, improved voice quality, and better standardisation. Therefore, next-generation cellular networks must incorporate the possibility of using much higher data rates when needed. On the other hand, there is no need to use higher data rates for normal telephone-quality voice communication. Using higher data

> **Facts to Know!**
>
> 3G cellular systems should have a variable data rate, flexibility of selecting appropriate data rate necessary for the application. Therefore, a 3G digital cellular technology should have greater capacity for voice communication as well as allowing higher data rates for data transfer.

rates than necessary results in not only wastage of spectrum, but also wastage of battery power in the mobile. Since higher data rates need greater bandwidth and therefore greater transmitter power for an equivalent signal-to-noise ratio at the mobile receiver. In fact, the trend in vocoder design is toward lower bit rates for telephone-quality voice.

The available data rate in 2G/2.5G cellular systems for data transfer is quite low because these systems transmit digitised voice at much lower bit rates. The recent development of PCS mobile phones with Internet browsing capabilities, and of notebook computers and personal digital assistants that can connect to a handheld mobile phone or can have a cellular or PCS phones built in for web browsing, only signify the urgent need for much higher data-rate capability at the system level for data transfer. Mobile users who are used to connecting to the Internet over wired LANs in offices or ADSL telephone-line connections or using cable modems at home still find wireless access speed extremely slow and unsatisfactory. Other high-speed data applications, such as streaming multimedia or video telephony, are simply impossible with present 2G or 2.5G digital cellular systems.

There is a lack of standardisation in the second-generation cellular networks, especially in North America, where three incompatible systems compete. Moreover, backward compatibility is still required with analog AMPS system if a mobile phone is to be used in rural areas. In addition to the various terrestrial microwave radio communication systems, each existing and proposed satellite systems use their own protocols and modulation scheme. It would certainly be useful to have one worldwide standard for the third-generation cellular technology.

13.6 THE IMT-2000 GLOBAL STANDARDS

Third-generation systems adopted by ITU is called IMT-2000, IMT stands for International Mobile Telecommunications, and the number 2000 represents frequencies in the 2000-MHz region (which the ITU wanted to make globally available for the new cellular technology), target user data rates of 2000 kbps (only under optimal conditions), and the year 2000 (when the ITU hoped the system would become available—a hoped year only). In ITU's IMT-2000, three different data rates are 144 kbps, 384 kbps, and 2000 kbps, each corresponding to a different type of carriers' core voice standard ISDN.

(a) 144 kbps as the absolute minimum acceptable transmission data rate. It corresponds to B-rate ISDN line, which is a high-speed fixed Internet access technology using optical fibers. Data rates of 144 kbps will be available for mobile users in high-speed vehicles.
(b) 384 kbps as the ideal achievable data rate. It corresponds to H-rate ISDN channel, often used for videoconferencing with picture quality approaching to that of television. Data rates of 384 kbps will be available for pedestrians and perhaps for slow-moving vehicles.
(c) 2000 kbps as the data rate that should be achievable while operating a mobile phone inside a building for stationary users. It corresponds to a European P-rate ISDN line, which is usually a fiber-optic cable carrying up to 30 separate parallel phone lines. The requirement is to set up small picocells in public areas, such as on trains or in airport departure lounges, giving users access to very high data-rate applications.

Facts to Know!

The need of having different data rates is due to the fact that the pedestrian users encounter less frequent or no hand-offs because of their lower speed than users in moving vehicles. Moreover, stationary users are much less affected by multipath fading.

13.6.1 Compatibility Requirements

The fundamental problem for a single global standard IMT-2000 is that no single standard could upgrade cdmaOne cellular system and handover to GSM cellular system. This means that one TDMA based standard and two very similar CDMA-based IMT-2000 standards are set for deployment. The main reason for this is compatibility with existing systems, which can be briefly described in three ways.

(a) Direct Upgrades Upgrades in cellular technologies typically add packet switching and better modulation techniques, while maintaining existing cell sizes and channel-assignment schemes. This limits the options substantially available with cellular service providers, where a majority of 2G systems are based on TDMA, so direct upgrades need to retain the TDMA structure.

(b) Roaming In principle, a mobile phone can be designed to support any number of different cellular systems, enabling them to be used worldwide. It involves multiple modes of operation, each representing a different 3G system. Some mobile phones may ultimately support all the IMT-2000 systems.

(c) Handover Roaming is inconvenient for most mobile users, as mobile phones have to be reinitialised to use a different network. A 3G system can be configured such that it actually hands over 3G mobile users to a 2G network as they move outside its coverage area. The mobile user should notice no difference, unless a dedicated 3G service such as multimedia is being accessed. This puts some design constraints in the 3G system. So mobile phones should be able to operate in both 2G and 3G networks.

13.6.2 Service Requirements

The data rates of user services are given in Table 13.2, together with their level of asymmetry and switching mode.

Although the first three services require circuit switching, this is likely to be accomplished via virtual circuits rather than actual circuits. The virtual circuits are more efficient than real ones because they send data during the pauses in conversation. Most videophone and videoconference protocols send only the parts of a picture that have changed instead of a complete new image for each frame, allowing significant bandwidth savings. Every bit of information is packetised, including voice, fax, and video, but packets in a virtual circuit are given priority over others. This guarantees capacity to a mobile user who has paid for it and frees it up for others when not in use.

13.6.3 Spectrum Requirements

New services along with predictions of mobile growth in future will certainly need additional spectrum. Assuming that existing 2G networks will eventually be upgraded to 3G, the ITU believes that at least 160 MHz more will be needed in each region. The ITU estimated the additional spectrum needed to accommodate it in each of its three global regions, as shown in Fig. 13.3, together with the spectrum already used by 2G services and allocated to 3G.

There are three main 3G systems: EDGE, W-CDMA, and Cdma2000. These are collectively known as IMT-2000 and offer packet-switched data at rates exceeding 384 kbps. EDGE is a straightforward upgrade to GSM and is also compatible with other TDMA systems, such as IS-136 and PDC. W-CDMA is designed to be backward compatible with GSM, and requires new spectrum. It is also known as UMTS. Cdma2000 is a straightforward upgrade to cdmaOne.

Table 13.2 *Service types available over IMT-2000*

Service classification	*Switch type*	*Upstream data rate*	*Downstream data rate*	*Asymmetry factor*	*Application*
Speech	Circuit	28.8 kbps	28.8 kbps	Symmetric	Telephony
Switched data	Circuit	43.2 kbps	43.2 kbps	Symmetric	Fax
Interactive multimedia	Circuit	256 kbps	256 kbps	Symmetric	Videoconference
Simple messaging	Packet	28.8 kbps	28.8 kbps	Symmetric	Email
Medium multimedia	Packet	19.2 kbps	768 kbps	40	Web Surfing
High multimedia	Packet	20 kbps	2000 kbps	100	TV

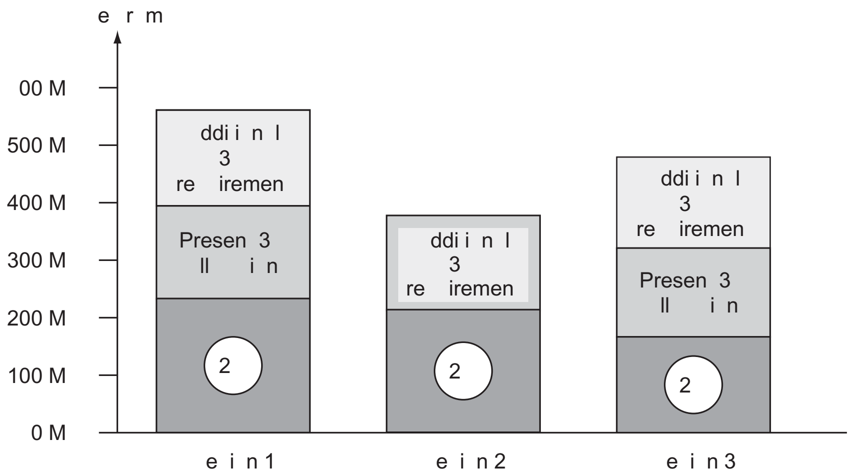

Fig.13.3 *Spectrum required by cellular systems in different ITU regions*

13.7 UMTS TECHNOLOGY

The main driver for mobile communications has been voice telephony. However, the introduction of new high-speed data capabilities, including GPRS and EDGE, and the evolution to UMTS, gives new and present GSM service providers the potential for offering a whole range of mobile multimedia services such as electronic postcards, web surfing, access to corporate LANs and Intranets, and e-mail from a mobile phone. UMTS is the standard for delivering 3G services and is based on the world's most widely deployed cellular mobile technology, GSM. It offers the prospect of a truly global wireless standard for personal multimedia communications.

Third-generation mobile wireless systems are often referred to as Universal Mobile Terrestrial Telecommunication Systems (UMTS). The term UMTS includes all aspects of the system, including the physical layer, network planning and architecture, protocols, services and applications. The objective of UMTS system is to integrate all forms of mobile communications, including terrestrial, satellite, and indoor communications. Consequently, UMTS must support a number of different air interfaces.

UMTS is a totally new radio access technology, which is not anymore based on GSM radio frequency but brings a completely new modulation method. It can provide a high-capacity radio interface to the GSM core network. UMTS will provide access at up to 2 Mbps in the local area and less than 1 Mbps wide-area access with full mobility. ETSI has decided to base the UMTS core network on the core-switching network evolved from GSM. Figure 13.4 depicts initial frequency use by UMTS/W-CDMA systems worldwide.

13.7.1 Objectives of UMTS Technology

1) Use of frequency bands of 1885 MHz – 2025 MHz and 2110 MHz – 2200 MHz
2) High-frequency spectrum efficiency
3) Radio-resource flexibility to multiple networks and traffic types within a frequency band; radio-bearer capabilities of up to 2 Mbps data rates

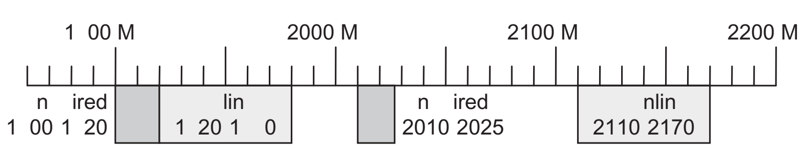

Fig.13.4 *Initial frequency allocation for UMTS/W-CDMA*

4) Integration of residential, office and cellular services into a single system and one user equipment
5) Applicability to different needs; public, private, basic telephony for simple telecommunications; broadband multimedia for advanced telecommunications
6) Seamless and global radio coverage achievable
7) Speech and service quality at least comparable to current fixed networks, including security that cannot be compromised in mobile use
8) Service capability up to multimedia
9) UMTS user number independent of network or service provider
10) Capacity and capability to service the whole population—creation of direct satellite access for a mass user base
11) Separation of service provision and network operation
12) Low cost of services and user devices; flexible personalisation, and ease of use
13) Flexibility for the introduction of new services and technical capabilities

13.7.2 UMTS Standardisation and Releases

UMTS Release R99 specifies the addition of the UMTS Radio Access Network (UTRAN), which is typically added to circuit-switched voice infrastructure and GPRS Internet access. This overlay network effectively increased the bandwidth of the core network to allow high-speed data transfer with an always-on connection. The addition of UMTS R99 effectively adds a new 'front-end' high-speed network to the voice and data networks. In UMTS networks, the modulation scheme is known as WCDMA, which has two basic modes of operation: FDD and TDD, which in turn enable faster connections. The new network elements introduced as part of UMTS R99 are The Radio Network Controller (RNC), and The Node B. R99 supports WCDMA and ATM-based transport mechanism.

Facts to Know!

Almost each year, a new release of the UMTS standard is published. The first UMTS release is R99, finalised in 2000 (not 1999).

The UMTS release 2000 or R00 supports GPRS/EDGE RAN (GERAN) and WCDMA RAN (UTRAN). It is further divided into two standards: release 4 or Rel-4 and release 5 or Rel-5. These mainly specify radio aspects, technical realisation, codecs, end-to-end QoS messaging, and several data-compression mechanisms. Rel-4 introduces new service architecture and quality of service in the fixed network. It also includes TD-SCDMA standard as low chip rate option (1.28 Mcps at 1.6 MHz bandwidth) to UTRA-TDD. Rel-4 also separates transport and control in IP-based CS domain. Rel-5 specifies all-IP-core network to integrate multimedia services, while retaining the same radio interfaces. A high-speed downlink packet access up to 14 Mbps peak data rate is also included along with a wideband 16-kHz AMR codec for better audio quality. Each RNC is assigned to exactly one SGSN and each SGSN serves one or more RNCs.

UMTS Rel-6 standard specifies High-Speed Uplink Packet Access (HSUPA), which is a 3G mobile telephony protocol with uplink speeds of up to 5.76 Mbps. The technical purpose of the enhanced uplink feature is to improve the performance of uplink dedicated transport channels, to increase capacity and throughput and reduce delay. HSUPA uses an uplink enhanced dedicated channel (E-DCH) on which it will employ link adaptation methods similar to those employed by HSDPA. UMTS Rel-6 features include E-DCH for providing significant uplink data capacity and throughput improvements; improved minimum performance specifications for support of advanced receivers that will increase downlink capacity and throughput; and the use of Multiple Input Multiple Output (MIMO) antennas to enable more efficient Multimedia Broadcast and Multicast Services (MBMS), security enhancements, interworking of UMTS and WLAN, IP-based emergency call features, and so on.

Further enhancements such as reduced latency, enhanced uplink performance, IP-based Multimedia Services (IMS) emergency call handling, increase user throughput (about 3 times compared to Rel-6), increased peak data rate (100 Mbps downlink, 50 Mbps uplink), handover to and from UTRAN and GERAN, improvements for real-time services like VoIP are part of standards through 3GPP UMTS Rel-7 and Rel-8. The objective is to move towards all-IP networks, integration of heterogeneous network types, and integration of heterogeneous access technologies. Release 8 is intended for use over any IP network, including WiMAX and Wi-Fi. The proposed E-UTRAN system uses OFDMA for the downlink (tower to handset) and single carrier FDMA for the uplink and employs MIMO with up to four antennas per station. The channel-coding scheme for transport blocks is turbo coding and a contention-free Quadratic Permutation Polynomial (QPP) turbo code internal interleaver.

UMTS Rel-9 specifies High Speed OFDM Packet Access, which is viewed as 4G technology based on OFDMA radio interface to support 50 Mbps uplink and 100 Mbps downlink data rate. Dual-Cell HSUPA is a wireless broadband standard that is defined in 3GPP UMTS Rel-9. The basic idea of the multicarrier feature is to achieve better resource utilisation and spectrum efficiency by means of joint resource allocation and load balancing across the uplink carriers. The specification is to be completed by 2009–10 and details of this technology are in the process of standardisation right now.

13.7.3 UMTS Network Architecture

The UMTS network architecture is shown in Fig. 13.5.

The UMTS network architecture is partly based on existing 2G network components and some new 3G network components. It inherits the basic functional elements from the GSM architecture on the Core Network (CN) side. The MS of GSM is referred as User Equipment (UE) in UMTS. The MSC has quite similar functions both in GSM and UMTS. Instead of circuit-switched services for packet data,a new packet node SGSN is introduced. This SGSN is capable of supporting data rates of up to 2 Mbps. The core-network elements are connected to the radio network via the I_u interface, which is very similar to the A-interface used in GSM. The major changes in the UMTS architecture are in the Radio Access Network (RAN), which is also called UMTS terrestrial RAN (UTRAN). There is a totally new interface called I_{ur}, which connects two neighbouring Radio Network Controllers (RNCs). BSs are connected to the RNC via the I_{ub} interface.

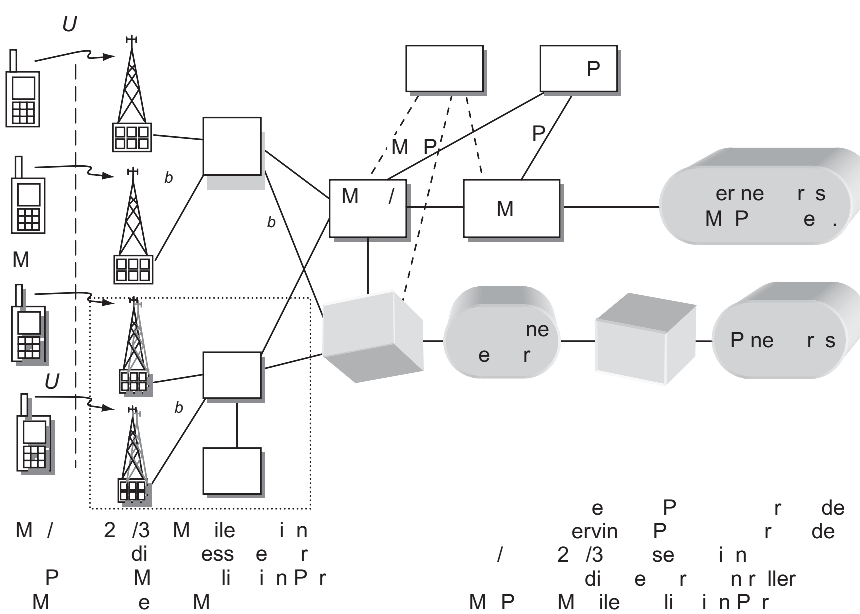

Fig.13.5 UMTS network architecture

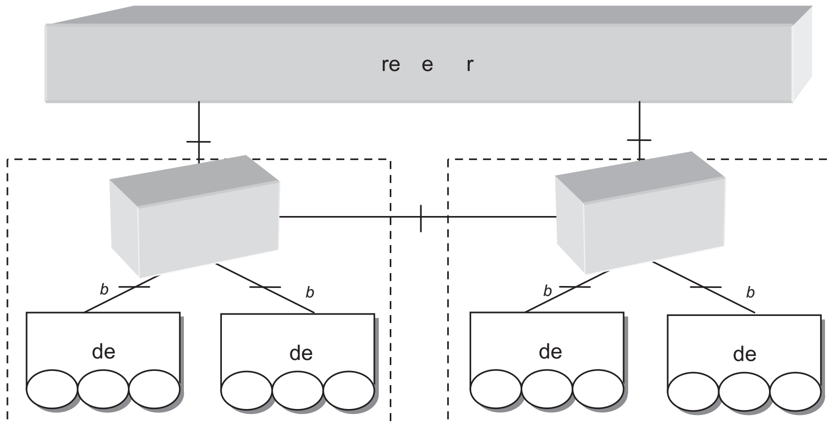

Fig.13.6 *UTRAN architecture*

UTRAN Architecture UTRAN consists of a set of Radio Network Subsystems (RNSs), as shown in Fig. 13.6. The RNS has two main elements: Node B and a radio network controller (RNC.) The RNS is responsible for the radio resources and transmission/reception in a set of cells.

An RNC is responsible for the allocation of all radio resources and use of the serving RNS. The responsibilities of the RNC include

- Radio resource management
- Serving RNS relocation
- Frame synchronisation
- Macro diversity combining
- Intra–UTRAN hand–off
- Splitting of the I_{ub} data streams
- Outer loop power control
- UMTS Radio Link Control (RLC) sublayers function execution

Many new nodes are introduced in the existing networks like NODE-B, Radio Network Controller (RNC) and Core Network (CN) consisting of the SGSN, GGSN and the MSC/VLR. The MSC/VLR would also need to be modified so as to be able to communicate with the 3G Radio Network over ATM. These nodes have to inter-operate with existing 2G or 2.5G nodes. The mobile phones also have to inter-operate with 2G and 3G modes. The two networks will co-exist and inter-operate. Therefore, it is essential to support seamless mobility between the two.

Radio Network Controller (RNC) The RNC is responsible for control of the radio resources in its area. One RNC controls multiple Node Bs. The RNC in UMTS networks provides functions equivalent to the Base Station Controller (BSC) functions in GSM/GPRS networks. The major difference is that RNCs have more intelligence built-in than their GSM/GPRS counterparts. For example, RNCs can autonomously manage handovers without involving MSCs and SGSNs.

Node B The Node B is responsible for air-interface processing and some radio-resource management functions. The Node B in UMTS networks provides functions equivalent to the Base Transceiver Station (BTS) in GSM/GPRS networks. Node Bs are typically physically co-located with the existing GSM Base Transceiver Station (BTS) to reduce the cost of UMTS implementation and minimise planning consent restrictions. This is likely to have a detrimental effect on UMTS quality as the UMTS Node Bs are being placed in non-optimal locations. UMTS operates at higher frequencies than GSM/GPRS and therefore the signal coverage range is less.

13.7.4 UMTS Interfaces

The UMTS interfaces can be categorised as follows:

U_u This is the interface between the user equipment and the network. That is, it is the UMTS air interface. The equivalent interface in GSM/GPRS networks is the U_m interface.

I_u-CS This is the circuit-switched connection for carrying (typically) voice traffic and signaling between the UTRAN and the core voice network. The main signaling protocol used is RANAP. The equivalent interface in GSM/GPRS networks is the A-interface.

I_{ub} This is the interface used by an RNC to control multiple Node B's. The main signaling protocol used is NBAP. The equivalent interface in GSM/GPRS networks is the A-bis interface. The I_{ub} interface is in the main standardised and open, unlike the A-bis interface in GSM/GPRS.

I_u-PS This is the packet-switched connection for carrying (typically) data traffic and signaling between the UTRAN and the core-data GPRS network. The main signaling protocol used is RANAP. The equivalent interface in GSM/GPRS networks is the G_b interface.

I_{ur} The primary purpose of the I_{ur} interface is to support inter-MSC mobility. When a mobile subscriber moves between areas served by different RNCs, the mobile subscriber's data is now transferred to the new RNC via I_{ur}. The original RNC is known as the Serving RNC and the new RNC is known as the drift RNC. The main signaling protocol used is RNSAP. There is no equivalent interface in GSM/GPRS networks.

13.7.5 UMTS Air Interface Specifications

WCDMA is the radio interface technology for UMTS networks, which is totally different from the technology used in TDMA or GSM. The traffic can vary from 8 kbps voice to 2 Mbps data and can be either circuit-switched or packet-switched. The UMTS air-interface specification has certain key features, which are listed in Table 13.3.

The UMTS air interface is based on DS-CDMA (direct sequence CDMA) technology. The user-data sequence is multiplied with a so-called spreading sequence, whose symbol or chip rate is much higher than the user-data rate. This spreads the user-data signal to a wider frequency band. The relation between user-data rate and chip rate is called a *spreading factor*, defined as the ratio of chip rate to data rate. The chip rate in WCDMA

Table 13.3 *UMTS Air-interface specifications*

S. No.	*Parameter*	*Specification*
1.	Frequency spectrum	*Uplink* 1920 MHz – 1980 MHz; *Downlink* 2110 MHz – 2170 MHz
2.	Channel bandwidth	5 MHz
3.	Chip rate	3.84 Mcps
4.	Duplexing technique	FDD and TDD modes
5.	Modulation scheme	Direct sequence CDMA with QPSK
5.	Frame length	10 ms frame with 15 time slots
6.	Coding technique	Orthogonal Variable Spreading Factor (OVSF)
7.	Service type	Multi-rate and multi-service

is 3.84 Mcps, and spreading factors are in the range of 4 to 512; therefore, the user net bit rates supported by one code channel are in the range of 1 kbps to 936 kbps in the downlink. Up to three parallel codes can be used for one user, giving bit rates of up to 2.3 Mbps. In the uplink, data rates are half that of those in downlink because of modulation differences.

The nominal channel bandwidth of the WCDMA signal is 5 MHz. The specification provides the flexibility to define the exact channel centre frequency of 200 kHz raster, so the actual channel separation might be smaller than the nominal 5 MHz, down to the specified minimum of 4.4 MHz.

Facts to Know !

The WCDMA standard includes two modes of operation: WCDMA/FDD and WCDMA/TDD. In WCDMA/FDD, the uplink and downlink signals are at different frequency bands. In WCDMA/TDD, the uplink and downlink signals are at the same frequency but are separated to different time periods.

The WCDMA transmission is split into 10-ms frames, each of which consists of 15 slots of 666 ms time slots. Each time slot consists of 2560 chips. The bit rate and, for example, channel coding can be changed in every 10-ms frame, offering very flexible control of the user-data rate. Every time slot has bits reserved for pilot signal, power control, transport format indication bits, and, if necessary, closed-loop transmit diversity bits. The exact signal format and multiplexing are quite different in uplink and downlink signaling. Also, the dedicated and shared channels have several differences in signal format.

13.7.6 UMTS Channels

There are three types of channels defined in UMTS: physical channels, logical channels, and transport channels. Physical channels are defined differently for FDD and TDD. FDD identifies a physical channel by its carrier frequency, access code, and the relative phase of the signal in the uplink. Similarly, TDD identifies a physical channel by its carrier frequency, access code, relative phase of the signal for the uplink, and the time slot in which it is transmitted. All physical channels follow four-layer structure of superframes, radio frames, sub-frames, and time slots/codes. The configurations of subframes or time slots are different depending on the resource-allocation scheme. The time slots or codes are used as a TDMA component so as to separate different user signals in the time and the code domain. All physical channels need guard symbols in every time slot.

Logical channels are described by the type of information they carry. Two types of logical channels are defined: control/signaling and traffic channels. Control/signaling logical channels consist of control information such as synchronisation and information related to the radio transmission required for process of a call. Traffic logical channels are used to transfer user and/or signaling data. The logical channels used by a UMTS system alongwith their main function and use for uplink or downlink are briefly given in Table 13.4.

Table 13.4 *Logical channels in UMTS*

Channel type	*Channel*	*Mainly used for*	*Direction*
Control Channel (CCH)	Broadcast Control Channel (BCCH)	Broadcasting system control information	Downlink (RAN → UE)
	Paging Control Channel (PCCH)	Transfer of paging information	Downlink (RAN → UE)
	Dedicated Control Channel (DCCH)	Transmission of dedicated control information	Uplink and Downlink (RAN ↔ UE)
	Common Control Channel (CCCH)	Transmission of control information	Uplink and Downlink (RAN ↔ UE)
	Shared Channel Control Channel (SHCCH)	Transmission of control information for TDD only	Uplink and Downlink (RAN ↔ UE)

(*Continued*)

Table 13.4 (*Continued*)

Channel type	*Channel*	*Mainly used for*	*Direction*
	Opportunity Driven Multiple Access (ODMA) Dedicated Channel (ODCCH)	Transmission of dedicated control information between UEs	Uplink and Downlink (RAN ↔ UE)
	ODMA Common Control Channel (OCCCH)	Transmission of control information between UEs	Uplink and Downlink (RAN ↔ UE)
Traffic Channel (TCH)	Dedicated Traffic Channel (DTCH)	Transfer of User information	Uplink and Downlink (RAN ↔ UE)
	ODMA Dedicated Traffic Channel (ODTCH)	Transfer of user information between UEs. An ODTCH exists in relay link.	Uplink and Downlink (RAN ↔ UE)
	Common Traffic Channel (CTCH)	Transfer of dedicated user information for all or a group of specified UEs	Downlink (RAN → UEs)

Transport logical channels signify as how the information is transmitted on the radio interface. Transport logical channels are the services offered by the physical layer to the higher layers. Transport logical channels can be classified into two groups:

Common Transport Logical Channels When there is a need for in-band identification of the UEs when particular UEs are addressed, common transport logical channels are used.

Dedicated Transport Logical Channels When the UEs are identified by the physical channel (frequency, time slot, and code), dedicated transport logical channels are used.

The transport logical channels used by UMTS system along with their main function and use for uplink or downlink are described in Table 13.5.

UMTS assures backward compatibility with the second-generation TDMA-based IS-136, GSM, and PDC technologies, as well as all 2.5G TDMA technologies such as GPRS and EDGE. The network structure and bit level packaging of GSM data is retained by UMTS with additional capacity.

13.7.7 UMTS Security Procedure

UMTS Authentication and Key Agreement (AKA) is a security mechanism used to accomplish the authentication features. The underlying mechanism is based on a challenge/response authentication protocol which is a security measure intended for a mobile subscriber to verify the identity of another mobile subscriber without revealing a secret password shared by the two. The key concept is that each mobile subscriber must prove to the other that it knows the password without actually revealing or transmitting such a password.

Facts to Know!

The authentication and key agreement procedure in UMTS is invoked by a serving SGSN after the mobile subscriber accomplishes registration, service request, location update request, an attach request and a detach request or connection re-establishment request.

The relevant information about the mobile subscriber must be transferred from the mobile subscriber's home network to the serving network in order to complete the process. The home network's HLR/AuC provides the serving network's VLR/SGSN with Authentication Vectors (AVs), each one holding the information fields. The authentication and key agreement procedure is summarised in the following steps:

Table 13.5 *Transport logical channels in UMTS*

Channel type	*Channel*	*Mainly used as/for*	*Direction*
Common Transport Channel (CTCH)	Random access Channel (RACH)	A contention-based channel to transmit small data	Uplink (UE → RAN)
	ODMA Random Access Channel (ORACH)	A contention-based channel in relay link	Uplink and Downlink (RAN ↔ UE)
	Common Packet Channel (CPCH)	A contention-based channel to transmit bursty data traffic	Uplink (UE → RAN)
	Forward Access Channel (FACH)	Small data transmission without closed-loop power control	Downlink (RAN → UE)
	Downlink Shared Channel (DSCH)	Shared channel by several UEs carrying dedicated control/traffic data	Downlink (RAN → UE)
	Uplink Shared Channel (USCH)	Shared channel by several UEs carrying dedicated control/traffic data in TDD mode only	Uplink (UE → RAN)
	Broadcast Channel (BCH)	Broadcasting system information into an entire cell	Downlink (RAN → UE)
	Paging Channel (PCH)	Broadcasting control information into an entire cell	Downlink (RAN → UE)
Dedicated Transport Channel (DTCH)	Dedicated Channel (DCH)	A channel dedicated to one UE	Uplink or Downlink (RAN ↔ UE)
	Fast Uplink Signaling Channel (FAUSCH)	To allocate dedicated channels in conjunction with FACH	Uplink (UE → RAN)
	ODMA Dedicated Channel (ODCH)	A channel dedicated to one UE used in relay link.	Uplink or Downlink (RAN ↔ UE)

Step 1. Visited network's VLR/SGSN requests a set of AVs from the HLR/AuC in the mobile subscriber's home network.

Step 2. HLR/AuC computes an array of AVs. This is done by means of the authentication algorithms and the mobile subscriber's private secret key K, which is stored only in the home network's HLR/AuC and the User Identity Module (USIM) in the mobile subscriber's mobile subscriber.

Step 3. Home network's HLR/AuC responds by sending *n* authentication vectors back to the visited network's VLR/SGSN.

Step 4. Visited network's VLR/SGSN chooses one AV and challenges the mobile subscriber's USIM by sending the RAND and AUTN fields in the vector to it.

Step 5. The mobile subscriber's USIM processes the AUTN. With the aid of the private secret key K, the mobile subscriber is able to verify that the received challenge data could only have been constructed by someone who had access to the same secret key K. The USIM will also verify that the AV has not expired by checking its Sequence Number (SEQ) field. Provided that the network can be authenticated and that the AV is still valid, the USIM proceeds to generate a Confidentiality Key (CK), an Integrity Key (IK) and a Response for the network (RES).

Step 6. The mobile subscriber responds with RES to the visited network.

Step 7. Visited network's VLR/SGSN verifies that response is correct by comparing the Expected Response (XRES) from the current AV with the Response (RES) received from the mobile subscriber's USIM.

13.8 W-CDMA AIR INTERFACE

One of the main air interfaces for 3G system is referred to as wideband CDMA (W-CDMA). W-CDMA is the dominant transmission technology for 3G cellular systems. The W-CDMA standard is a new radio interface standard or radio access technology. WCDMA is only the air interface used with UMTS mobile communication standard which allows communication between the base station and mobile station. WCDMA is also termed as the multiple access technique used in UMTS. In other words, WCDMA is simply the air interface, UMTS is the complete set of standards and protocols, including the specification that UMTS use WCDMA as the air interface.

The 3G W-CDMA air interface standard is designed for always-on packet-based wireless service. W-CDMA can support packet data rates of up to 2 Mbps per user in static condition. The system offers high-quality data, multimedia, streaming audio/video and broadcast services to mobile users.

W-CDMA has been designed to allow handovers to GSM systems, but GSM networks cannot be upgraded to W-CDMA. Some network components such as the GPRS backbone can be reused. W-CDMA is designed to provide backward compatibility and interoperability for all IS-136, GSM, GPRS, EDGE, and PDC systems except the wider air-interface bandwidth necessitates a complete change of the base station RF equipment. The wideband designation in W-CDMA refers to the channel bandwidth of 5 MHz. This is four times that of cdmaOne (1.25 MHz), and 25 times that of GSM (200 kHz). Such a wider bandwidth is chosen to allow higher data rates as low as 8 kbps to as high as 2 Mbps on a single W-CDMA 5 MHz radio channel. Each channel can support between 100 and 350 simultaneous voice calls, depending on antenna sectoring, radio propagation conditions, and vehicle speed of the mobile user.

W-CDMA employs variable or selectable direct sequence spread spectrum chip rates that can exceed 16 Mcps per user. Unlike cdmaOne, which automatically sends every bit of information 64 times, W-CDMA adjusts the gain depending on the received signal strength. Every bit is sent between 4 and 128 times, which means that greater bandwidth is available in areas with a stronger signal. W-CDMA can provide at least six times increase in spectral efficiency over GSM at system level.

The other major difference between W-CDMA and cdmaOne is the need for time synchronisation. W-CDMA has been designed to operate without GPS clock signals. The system needs a slightly different coding technique called Gold codes. Combined with the same QPSK modulation as used in cdmaOne, these give a maximum data rate of around 4 Mbps per channel per cell, exceeding the IMT-2000 requirements. Each channel is reused by every cell along with soft handovers, enhancing spectral efficiency to large extent as compared to TDMA systems.

The W-CDMA uses a chipping rate of 4.096 Mbps with only one carrier per 5-MHz channel and is therefore not backward compatible with IS-95. It also does not rely on the synchronisation of cell-sites by means of the GPS satellites. This provides advantages in the case of indoor cell-sites where the reception of satellite signals is difficult. The salient features of W-CDMA can be summarised as follows:

- Coverage and capacity for speech services in the W-CDMA system is better than GSM, under the same operating conditions.
- The W-CDMA system is capable to handle a mix of real time, variable bit rate, and packet services efficiently and flexibly.
- A data rate of 384 kbps is possible to provide with full coverage (everywhere) and a data rate of 2 Mbps in indoor applications.

Table 13.6 shows some of the key parameters of W-CDMA standard.

Table 13.6 *W-CDMA parameters*

S. No.	*Parameter type*	*Specification*
1.	Channel bandwidth	5 MHz
2.	Forward RF channel structure	Direct sequence spread spectrum
3.	Chip rate	3.84 Mcps
4.	Coherent detection	Pilot symbols
5.	Reverse channel multiplexing	Control and pilot channel time multiplexed. I and Q multiplexing for data and control channels
6.	Spreading modulation	Balanced QPSK (forward channel); Dual channel QPSK (reverse channel)
7.	Spreading factors	4 to 256
8.	Spreading (forward)	Variable-length orthogonal sequences for channel separation. Gold sequences 2^{18} for cell and user separation.
9.	Spreading (reverse)	Same as forward, different time shifts in I and Q channels
10.	Data modulation	QPSK (forward channel); BPSK (reverse channel)
11.	Multirate	Various spreading and multicode
12.	Frame length	10 ms
13.	Number of slots/frame	15
14.	Power control	Open and fast closed loop (1.6 kHz)
15.	Handover	Soft handover

The advantage of multirate is that the system can flexibly support multiple simultaneous applications from a given mobile user and can efficiently use available capacity by only providing the capacity required for each service. Multirate can be achieved with a TDMA scheme within a single CDMA channel, in which a different number of time slots per frame are assigned to achieve different data rates. All the subchannels at a given data rate are protected by error correction and interleaving techniques. An alternative method is to use multiple CDMA codes, with separate coding and interleaving schemes, and map them to separate CDMA channels.

Facts to Know!

The term 'multirate' refers to the provision of multiple fixed-data-rate logical channels to a given user, in which different data rates are provided on different logical channels. Further, the traffic on each logical channel can be switched independently through the wireless and fixed networks to different destinations.

The higher data rates require a wide radio frequency band, which is why a W-CDMA with 5 MHz carrier has been designated. The W-CDMA system aims at efficient use of the available spectrum to provide digital cellular services. The W-CDMA system has larger frequency bandwidths of 5.0 MHz, 10.0 MHz and 15.0 MHz. The system supports Type A hand-offs (equivalent to soft hand-offs in CDMA) and Type B hand-offs (equivalent to hard hand-offs in CDMA). The W-CDMA system uses Walsh functions for bandwidths of 5.0 and 10.0 MHz and Hadamard functions for a bandwidth of 15.0 MHz, to code the digital signals for voice, data and control information, in both forward and reverse channels. Synchronisation can be obtained on the reverse channel at the higher data rates.

13.8.1 Comparison of W-CDMA and IS-95

(a) Bandwidth and Chip Rate W-CDMA increases the chip rate by more than a factor of three, and the channel bandwidth by a factor of four, compared with IS-95. The faster chip rate implies that the W-CDMA receiver can provide greater multipath resolution. And with a RAKE receiver, greater frequency diversity can be obtained. Consequently, performance is expected to be more robust in wireless channels.

(b) Data Rates and Spreading Factor The IS-95 system has a standard data rate that is spread by a fixed spreading code of length 64; lower data rates are provided by repeating the bits. W-CDMA provides a whole range of data rates through its ability to adjust the spreading factor. On the uplink, the spreading factor can be as high as 512. The higher spreading rate permits a lower but more robust data rate. Individual user channels on the uplink are asynchronous and not orthogonal. On the downlink, the spreading factor can range from 4 up to 256. Different mobile users are combined on the downlink via Orthogonal Variable Spreading Factor (OVSF) codes. This combination provides multiples of the basic data rate while minimising the intracell interference on the downlink, thereby permitting bit rates up to 384 kbps on circuit-switched connections and up to 2 Mbps on packet-switched networks. Each mobile phone is capable of transmitting multiple channels on the uplink; the channels are kept orthogonal using the OVSF codes.

(c) Modulation and Synchronisation The IS-95 system uses a pilot channel in the downlink direction to provide synchronisation, channel tracking, and handover functions. In the uplink direction, orthogonal modulation is employed, which permits the more robust noncoherent demodulation to be used. W-CDMA employs coherent detection on both uplink and downlink because a pilot channel is included in both directions. In the downlink direction, each cell-site transmits a pilot channel, as well as multiplexing pilot symbols with the data channels. In the uplink, each mobile user multiplexes pilot symbols with its data.

(d) Forward Error-Correction Codes Both IS-95 and W-CDMA use convolutional codes for performing Forward Error-Correction (FEC) coding except that the IS-95 standard provides a small number of data rates that are implemented by repeating the data symbols or puncturing the code bits by means of a simple pattern. Whereas the W-CDMA standard allows for a wide variety of data rates by allowing variable puncture patterns based on 1/2-rate and 1/3-rate convolutional codes of constraint length 9. The W-CDMA standard includes the option for applying the recently developed and more powerful turbo codes for forward-error correction. The eight-state rate-1/3 version of the turbo codes is employed in this system.

EXAMPLE 13.1 Technical parameters of W-CDMA with IS-95 CDMA system

List the comparison of the main technical parameters of third-generation W-CDMA cellular system and second-generation IS-95 CDMA cellular system.

Solution

Technical Parameter	W-CDMA	IS-95
Channel bandwidth	5 MHz	1.25 MHz
Chip rate	3.84 Mcps	1.2288 Mcps
Data rates	upto 2 Mbps	upto 9.6 kbps
Frame size	10 ms	20 ms
Spreading factor	upto 512	64
Number of channels/terminal	variable	1
Downlink/uplink sharing	FDD/TDD	FDD

Downlink modulation	QPSK	QPSK
Uplink modulation	QPSK	OQPSK/Orthogonal
Downlink FEC	r = 1/2, 1/3 convolutional or turbo	r = 1/2, L=9 convolutional code
Uplink FEC	r = 1/2, 1/3 convolutional or turbo	r = 1/3, L=9 convolutional code
Voice encoding	Adaptive multirate ACELP (4.75 kbps to 12.2 kbps)	CELP at 9.6 kbps and 14.4 kbps
Traffic channels/RF channel	Depends upon data rate	upto 63 in theory

13.8.2 Attributes of W-CDMA System

The major attributes of the W-CDMA system are the following:

(a) Service Flexibility W-CDMA allows each 5 MHz carrier to handle mixed data rate services ranging from 8 kbps up to 2 Mbps. In addition, circuit- and packet-switched services can be combined on the same channel, thereby allowing true multimedia service with multiple packet or circuit connections on a single mobile phone. Services with different quality requirements can be supported with excellent capacity and coverage.

(b) Spectrum Efficiency W-CDMA makes very efficient use of available radio spectrum. Frequency planning is not necessary. A two-or three-layer network can be deployed within the 2 × 15 MHz frequency allocated to service providers since each cell layer requires 2 × 5 MHz spectrum only.

(c) System Capacity and Coverage The W-CDMA system provides improved capacity over the other digital systems. The increased system capacity is due to improved coding gain, voice activity, cell sectorisation and reuse of the same spectrum in every cell. W-CDMA radio transceivers can handle eight times more voice traffic than narrowband transceivers. Each RF carrier can handle approximately 80 simultaneous voice calls, or 50 simultaneous Internet-type data users per carrier. The capacity of W-CDMA is approximately double that of narrowband CDMA in urban and suburban environments. The capacity can be further enhanced by adding support for hierarchical cell structures, adaptive antenna arrays and multi-user detection. The wider bandwidth, use of coherent demodulation, and fast power control in the uplinks and downlinks yield a lower receiver threshold, thereby improving radio coverage.

Facts to Know!

Hierarchical cell structures use a new hand-off method, called mobile-assisted inter-frequency hand-off, between W-CDMA carriers. Adaptive antenna arrays optimise the antenna pattern for each individual mobile user, improving spectrum efficiency and capacity. Multi-user detection minimises interference within a cell and improves capacity.

(d) Improved Voice Capacity Third-generation wireless access is also a very spectrum-efficient mechanism for voice traffic. For instance, operators with a 2 × 15 MHz spectrum allocation will be able to handle at least 192 voice calls per sector or 576 voice calls per cell having 3-sector configuration.

(e) Multiple Services Per Connection W-CDMA accommodates circuit- and packet-switched services with variable bandwidths to be mixed freely and delivered simultaneously while maintaining specified quality levels to the same mobile user. Each mobile user can access multiple services at the same time including voice or a combination of data services.

(f) Fast Service Access A new random-access procedure has been developed that uses fast synchronisation to handle 384 kbit/s packet-data services to support instant access to multimedia services. The procedure requires only a few tenth milliseconds to set up connections between a mobile user and a base station.

(g) Asynchronous Radio Access IS-95 depends on the GPS for synchronisation, which can make implementation for indoor base stations difficult and expensive. W-CDMA has its own internal system for synchronising base stations and it does not depend on external system synchronisation.

(h) Quality of Service The W-CDMA system provides robust operation in fading environment and eliminates the possibility of a dropped call. The system exploits the advantage of multipath fading to enhance communication link and voice quality. By using a RAKE receiver and other advanced signal-processing techniques, each mobile user selects the three strongest multipath signals and coherently combines them to produce an enhanced signal. By using Type A hand-off, a connection is made to the new cell while maintaining the connection with the previous cell.

(i) Seamless Access Dual-mode mobile phones provide seamless handover and roaming access with mapping of services between GSM or IS-136 and IMT-2000 networks. The choice of W-CDMA offers a unique opportunity for creating a harmonised global standard with seamless global roaming for next-generation cellular services.

(j) Economies of Scale The W-CDMA is a cost-effective technology that requires fewer, less expensive cells and no costly frequency reuse planning. The addition of W-CDMA wireless access to a digital cellular network and interworking between the two systems permit the use of existing core network and base-station sites. The average power transmitted by the W-CDMA mobile phone averages about 6–7 mW only, which also means longer battery life.

Facts to Know!

The W-CDMA system offers a significant improvement in system capacity as well as communication and voice quality. The system exploits the multipath fading to the advantage in establishing the best cell and maintaining the call continuity. The system uses Walsh or Hadamard codes at higher bit rates of 4.096 Mcps, 8.192 Mcps and 12.288 Mcps to code the digital signals for voice, data and control information, in both uplink and downlink. The pilot signal synchronisation can be obtained on the downlink also.

(k) Modulation Technique Used in W-CDMA The W-CDMA system uses Direct Sequence Spread Spectrum (DSSS) technique. The spreading of the data is performed by means of a spreading or PN code signal. The code signal is independent of the data and is of a much higher rate than the data signal. At the receiver, de-spreading is accomplished by the cross-correlation of the received spread signal with a synchronised replica of the same signal used to spread the data.

(l) Variable Data Rates The W-CDMA physical layer is required to support variable bit rate transport channels, to offer bandwidth-on-demand services, and to be able to multiplex several services to one connection. With these multiple data streams, forward-error correction is independently applied to each data stream depending upon different service quality requirements. For example, if a voice channel and a video channel are transmitted simultaneously, the voice channel can generally tolerate a higher error rate than the video channel. Consequently, it would use a less powerful or higher rate FEC code. W-CDMA allows for two layers of interleaving—the first over 10 milliseconds, and the second over 20, 40, or 80 milliseconds. These two interleaving layers are present on both the uplink and the downlink.

(m) Multicode Transmission The simultaneous transmission of two or more CDMA channels by the same mobile user is referred to as multicode transmission. This form of transmission tends to increase the peak-to-average ratio of the transmitted waveform, thereby affecting the efficiency of power amplifier of the mobile phone. If only two channels are transmitted, the I channel and the Q channel, then with the use of a special complex scrambling code, the peak-to-average ratio of the W-CDMA signal is kept small, even if the two channels are transmitted at different power levels. When more than two channels are transmitted, the peak-to-average ratio becomes even larger. Due to use of battery with the mobile phone, the peak-to-average ratio is more of a concern in the uplink.

13.8.3 W-CDMA Channels

The forward channels are the pilot channel, the sync channel, the paging channel, and the forward traffic channel. On the reverse link, the channels are the access channel and the reverse traffic channel. The sync channel establishes the synchronisation of the received data at the MS receiver. The paging channel is used to page the MS and send orders to it. The reverse access channel enables the MS to respond to orders received on the paging channel, and originate calls. The traffic channels are used for digitised voice or data. The system permits control information to be multiplexed into the traffic channel stream to send and receive data between the MS and the BS.

Forward W-CDMA Channel The forward W-CDMA channel consists of a pilot channel, a sync channel (optional), up to a maximum of seven paging channels (optional), and several forward traffic channels. Each of these channels is orthogonally spread by the appropriate orthogonal function and is then spread by a quadrature pair of PN sequence. All the channels are added together and sent to the modulator. When a base station supports multiple forward W-CDMA channels, frequency division multiplex is used.

Pilot Channel (Refer Fig. 13.7) A pilot signal is transmitted by a base station to enable the mobile station to help in clock recovery. The pilot signal consists of the all-zeros pattern. It is modulo-2 added to the orthogonal code comprising of Walsh 0 or Walsh 64 function for 5.0 MHz bandwidth at chip rate of 4.096 Mcps, the Walsh 0 or Walsh 128 function for 10.0 MHz bandwidth at chip rate of 8.196 Mcps and the Hadamard 0 function for 15.0 MHz bandwidth at chip rate of 12.288 Mcps. The pilot signal is then sent to the modulator.

Sync Channel (Refer Fig. 13.8) The sync channel is transmitted by a base station to enable the mobile station to obtain frame synchronisation of the W-CDMA signal. The W-CDMA system uses a 16-kbps sync rate for all three bandwidths. The sync signal convolutionally encodes the data with a rate one-half code. The W-CDMA system processes the output of the convolutional encoder as two separate data streams. After encoding, the signal is processed by a block interleaver and a symbol repetition block. The exact order of these two stages does not matter as long as the appropriate bits are in the correct place after the two stages. After repetition, the resultant signal is modulo-2 added with the appropriate orthogonal code, comprising of Walsh 32 or Walsh 96 function for 5.0 MHz bandwidth, the Walsh 64 or Walsh 192 function for 10.0 MHz bandwidth and the Hadamard 128 function for 15.0 MHz bandwidth. The sync signal is then sent to the modulator.

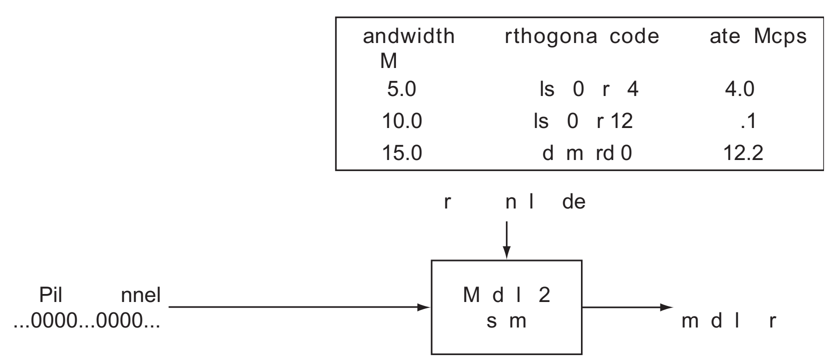

Fig. 13.7 Forward W-CDMA pilot-channel processing

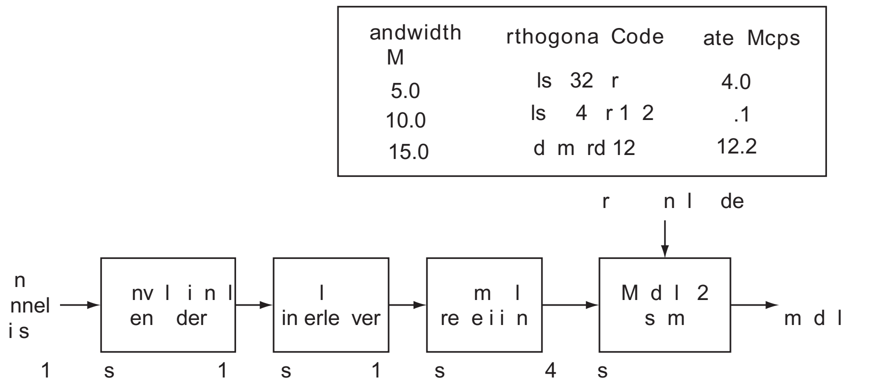

Fig. 13.8 | *Forward W-CDMA sync channel processing*

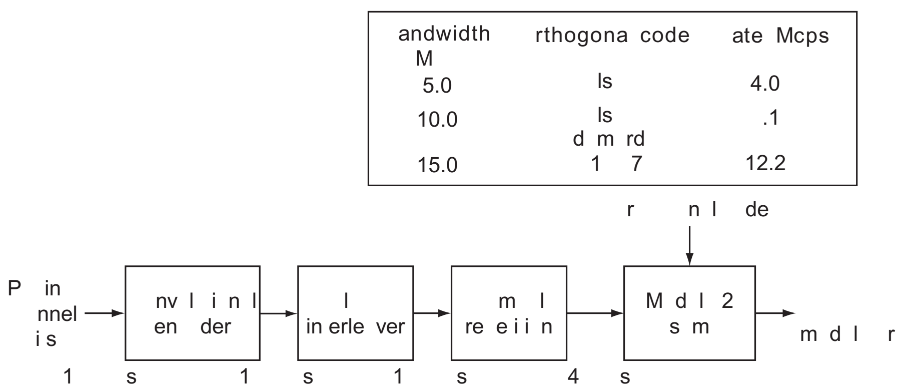

Fig. 13.9 | *Forward W-CDMA paging channel processing*

Forward Paging Channel (Refer Fig. 13.9) The paging channel is transmitted by a base station to enable the mobile station to be paged and to process other orders while the mobile station is powered on and is idle. The W-CDMA system uses a 16-kbps rate for the paging channel.

The paging signal convolutionally encodes the data with a rate one-half code. After encoding, the signal is processed by a block interleaver stage and a symbol repetition stage. After repetition, the resultant signal is modulo-2 added with the appropriate Walsh or Hadamard orthogonal codes. There are from 0 to 7 paging channels using codes 1 to 7, respectively and in sequence. Thus, for a base station with five paging channels, codes 1, 2, 3, 4, and 5 are used. The paging signal is then sent to the modulator.

Forward Traffic Channel (Refer Fig. 13.10) The traffic channel is transmitted by a base station to enable the mobile station to carry voice or data traffic. The W-CDMA system uses 16, 32 or 64 kbps data rate to carry traffic voice/data and 2 or 4 kbps data rate to carry signalling data, depending on the type of voice encoding chosen.

The traffic voice/data channel bits and the signalling channel bits convolutionally encode the data with a rate one-half code separately. After encoding, both the signals are then processed by their respective block interleaver stage and a symbol repetition stage. The power control bits at 2 kbps rate are processed through a symbol repetition stage. The traffic channel data, the signalling channel data and power control bits are then multiplexed. The resultant signal can carry voice or data, power control bits and signalling channel data. Thus, the W-CDMA multiplexes the channel after the symbol repetition and always processes a separate signaling channel.

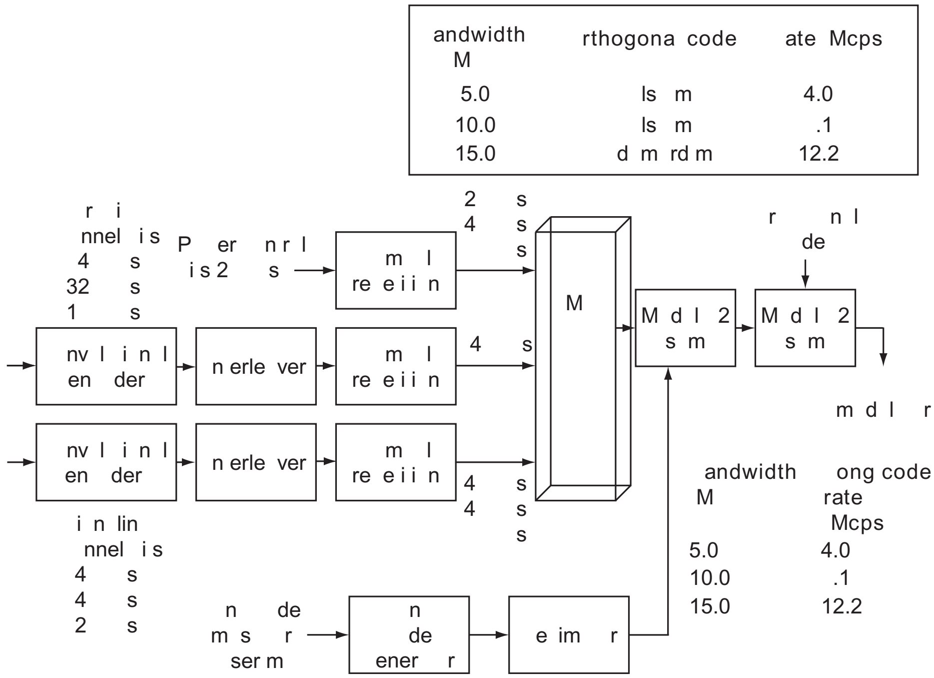

Fig. 13.10 *Forward W-CDMA traffic channel*

The resultant multiplexed signal is then further scrambled by the decimated long code in Modulo 2 Summer. The scrambler is constructed using every 64th bit from a long code generator. The use of 1 out of 64 bits is called decimation. The scrambler prevents long sequences of 0s or 1s from appearing in the data stream. The long-code mask chosen for each channel establishes the voice (or data) privacy for that channel.

Finally, the resultant signal is modulo-2 added to the Walsh (or Hadamard) code for the channel being used.

Forward W-CDMA Channel Modulator (Refer Fig. 13.11) The I and Q signals from each channel (pilot, sync, paging and traffic) are modulo-2 added to an I and 0 pseudorandom noise sequence (also called PN codes). For the W-CDMA system, the I and Q signals are different and are derived from the processed data from the convolutionally encoded data. The same PN sequence is used for both the I and Q channels. The I and Q spread signals are the baseband filtered and the signals from all channels are sent to a linear adder with gain control. The gain control permits the individual channels to have different power levels assigned to them.

The W-CDMA system assigns power levels to different channels depending on the quality of the received signal at a mobile station. The algorithms for determining the power levels are proprietary to each equipment provider. The I and Q baseband signals are then modulated by the I and Q carrier signals, combined together, amplified and sent to the antenna. The resultant signal from the W-CDMA modulator is a quadrature phase shift signal.

The same PN sequence is used on all channels (pilot, sync, paging and traffic) of the W-CDMA forward channel. All base stations in a system are synchronised using the global positioning system satellites. Different base stations use time-shifted versions of the PN sequence to enable mobile stations to select the appropriate base station.

Reverse W-CDMA Channel The reverse path from the mobile station to base station uses a difference frequency band. The reverse W-CDMA channel consists of access channels and reverse traffic channels. All mobile stations accessing a base station over an access channel or a traffic channel share the same W-CDMA frequency

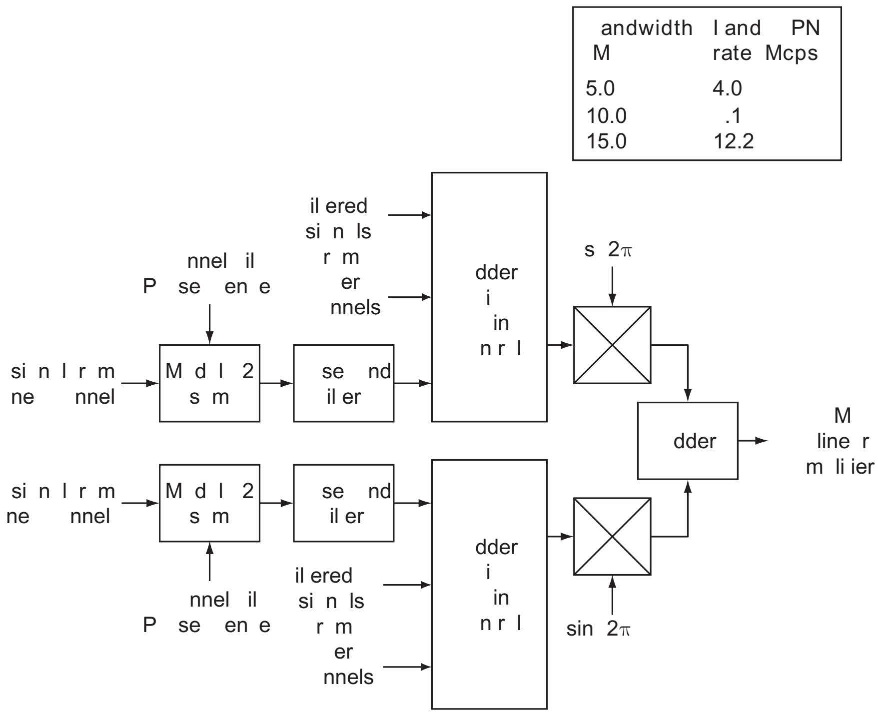

Fig. 13.11 *Forward W-CDMA channel modulator*

assignment using direct sequence W-CDMA techniques. The channel selection is performed using Walsh (or Hadamard) codes, similar to the forward channel. Multiple reverse W-CDMA channels may be used by a base station in a frequency-division multiplexed manner. The W-CDMA system uses Walsh (or Hadamard) functions in both forward and reverse directions because W-CDMA system is designed in such a way so as to recover the pilot signal transmitted by a mobile station.

Reverse Access Channel (Refer Fig. 13.12) The reverse access channel is used by the mobile station to access the W-CDMA system to respond to pages, make call originations and process other messages between the

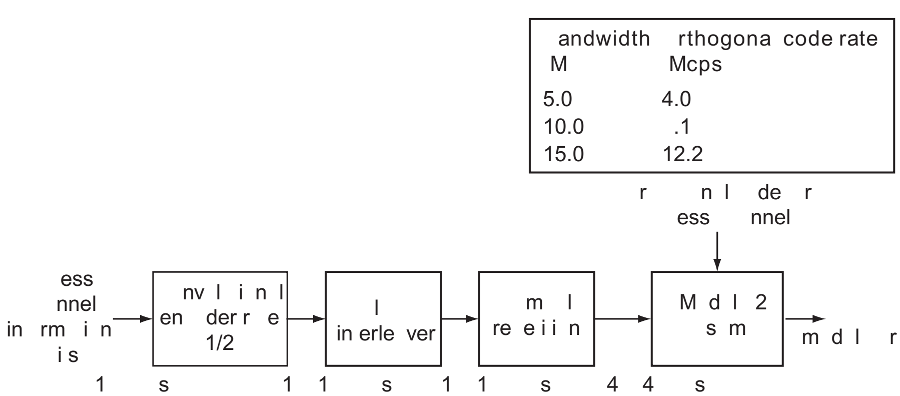

Fig. 13.12 *Reverse W-CDMA access channel*

mobile station and the base stations. The channel operates at 16 kbps. The information bits are convolutionally encoded and processed by block Interleaver and symbol repetition functions. The output of the symbol repetition stage is modulo-2 added to a Walsh (or Hadamard) code for the reverse access channel. The orthogonal properties of the Walsh (or Hadamard) codes are used by the base station to select mobile station transmissions. The orthogonally spread signal is then sent to the modulator.

Reverse Traffic Channel (Refer Figs. 13.13, 13.14) For the W-CDMA system, the reverse traffic channel information bits and signalling information bits are processed separately and channel multiplexing is done in the modulator. The output of the reverse traffic channel (encoded speech at 16, 32 or 64 kbps or data at rates upto 64 kbps) is convolutionally encoded, block interleaved and repeated (if the data rate is less than 64 kbps). The output signal is then modulo-2 summed with the orthogonal code (Walsh or Hadamard) for the reverse traffic channel and sent to the modulator.

For the W-CDMA system, signalling occurs on the reverse traffic channel at 2 or 4 kbps. The output of the signalling channel is convolutionally encoded, block interleaved and repeated to generate a symbol rate of 64 kilosymbols per second (ksps). The output signal is then modulo-2 summed with the orthogonal code (Walsh or Hadamard) for the signalling channel and sent to the modulator.

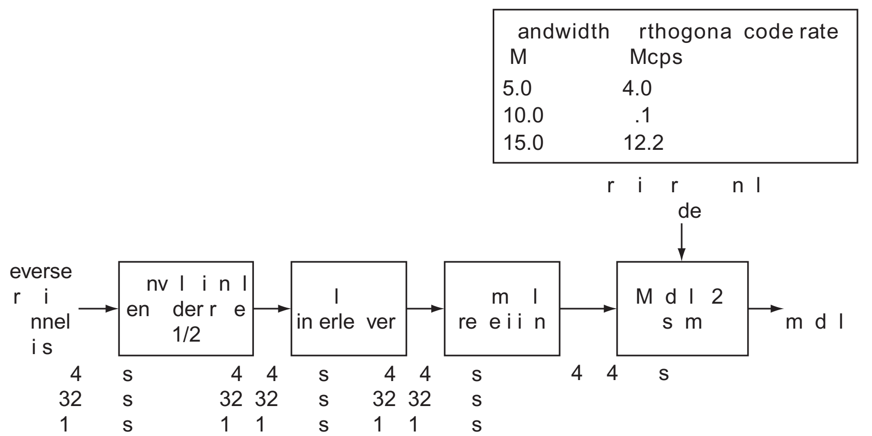

Fig. 13.13 | *Reverse W-CDMA traffic channel*

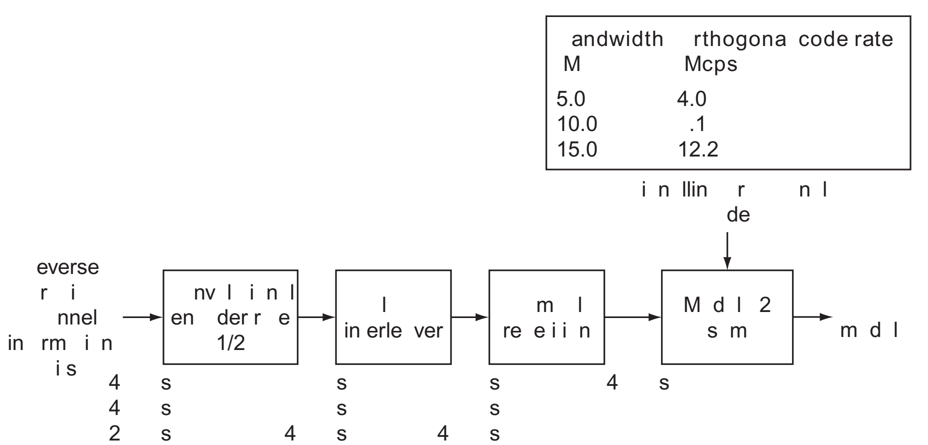

Fig. 13.14 | *Reverse W-CDMA traffic information channel*

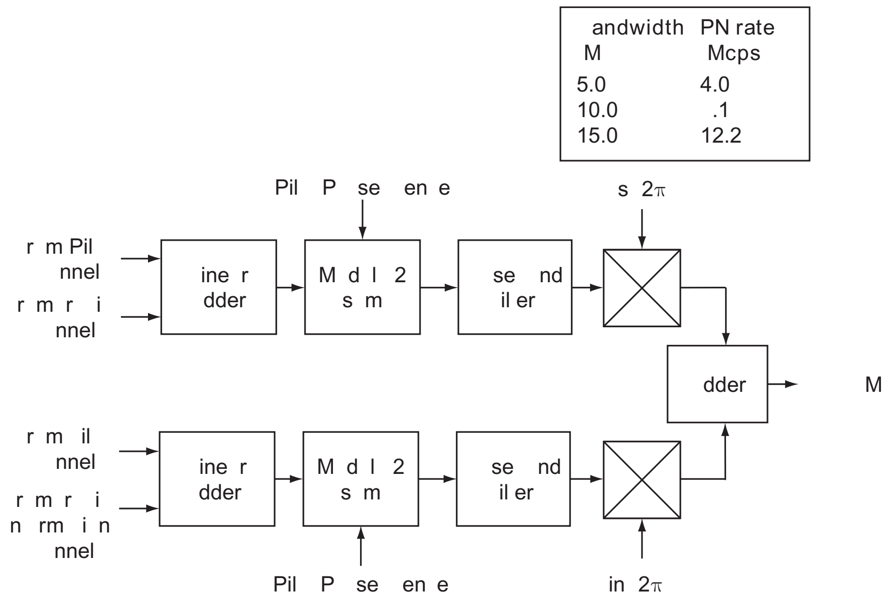

Fig. 13.15 *Reverse W-CDMA modulator*

Reverse W-CDMA Channel Modulator (Refer Fig. 13.15) For the W-CDMA system, the in-phase and quadrature signals are separated after convolutional encoding. The I channel linearity adds the traffic channel data and a reduced level pilot channel (–10 dB below the traffic channel). The Q channel linearity adds the other traffic channel and traffic information channel. Both the I and the Q channels are then modulo-2 summed with the same pilot PN sequence, bandpass filtered, and sent to the modulator. No phase delay is used on the Q channel, so that the resultant modulation is quadrature phase-shift keying.

13.8.4 Modulation Parameters

In order to process the different bit streams on the forward and reverse channels as described in the above section, the various modulation stages involved in W-CDMA system are the following:

(a) Convolutional Encoding With convolutional encoding, the encoded output is a function of the input bit stream and delayed versions of the bit stream. For the W-CDMA system, a rate one-half code is used in both directions, i.e, forward and reverse channels. However, the output of the two bit streams are processed in parallel and carried through to further modulation stages as two parallel bit streams.

(b) Bit Repetition For the W-CDMA system, the nominal data rate is 64 kbps. Therefore, any channel-transmitting data at less than 64 kbps are rate multiplied up to a constant 64-kbps rate.

(c) Block Interleaving The communications over a radio channel are characterised by deep fades that can cause a large number of consecutive errors. Most coding systems perform better on random data errors rather than on blocks of errors. By interleaving the data, no two adjacent bits a re transmitted near to each other, and the data errors are randomised. For the W-CDMA system, the block interleaver span is 5 ms (the traffic channel spans of 10 ms and 20 ms are supported as options). Different interleaver matrices are used for different channels. All channels operate at a constant 64-kbps rate.

(d) Orthogonal Codes For the W-CDMA system, the data streams are modulo-2 added to an orthogonal code in both forward and reverse directions. The orthogonal code is 1 of 256 Walsh functions at 5-MHz bandwidth, 1 of 512 Walsh codes at 10-MHz bandwidth and 1 of 768 Hadamard codes at 15-MHz bandwidth. For the reverse W-CDMA channel, the orthogonal code is the modulo-2 sum of the Walsh or Hadamard codes and the long code.

> **Facts to Know!**
>
> For W-CDMA, a 5-MHz band will use a single 5-MHz W-CDMA system A 15-MHz spectrum will use either one 15-MHz system, or one 10-MHz and one 5-MHz system, or three 5-MHz systems.

(e) Direct PN Spreading The W-CDMA system modulates the processed data stream with a pseudorandom noise spreading sequence at the fundamental rate for the system (4.096, 8.192, or 12.288 Mcps). For the W-CDMA system, the code $g(x) = x^{32} + x^{22} + x^2 + x + 1$ is used for both the I and the Q channels. The PN sequence generator is embedded with data received in messages sent from the base station. The process is used to establish the voice and data privacy on the channel. The same process is used in both forward and reverse directions.

(g) Frequency Tolerance The W-CDMA base-station transmitter must maintain its frequency to within 5 parts per 100 million (100 Hz at 2000 MHz). The mobile station transmit carrier frequency will be 80 MHz 200 Hz below the corresponding base station transmit signal.

(f) Baseband Filtering After PN modulation, the signal is filtered by BB filter.

(h) Power Control in W-CDMA The W-CDMA system defines three different power classes of mobile phones, as shown in Table 13.7 below:

Table 13.7 *Maximum EIRP for W-CDMA mobile phones*

Mobile phone class	*EIRP at maximum output shall exceed*
I	+23 dBm (200 mW)
II	+13 dBm (20 mW)
III	+3 dBm (2 mW)

The mobile user attempts to control the power output based on received signal strength (open-loop power control), and the base station sends power control messages to the mobile user about once every millisecond (closed-loop power control). The net effect is to control the power received at the base station to within 1 dB for all mobile users. The precise power control is necessary for proper operation of the W-CDMA system.

The base station can also vary its transmitted power within 4 dB depending on the error rates reported by the mobile user. The mobile user can report frame-error rates measurement data on the forward traffic channel. The mobile user gates its power on and off depending on the data to be transmitted. This feature is useful for voice-encoded data to improve system performance since the transmitter will be off during pauses in speech and it must reduce its power by 20 dB.

When a mobile user attempts an access on the reverse access channel, it must transmit at a power level of

$$P_{MS}\,(\text{dBm}) = P_{mean} + \text{NOM PWR} + \text{INIT PWR} - \text{P CNST} \tag{13.1}$$

where P_{mean} is the mean input power of the transmitter; NOM PWR is the nominal correction factor for the base station, INIT PWR is the correction factor for the base station from partial path loss decorrelation between transmit and receive frequencies. Its range is $-47 <$ INIT PWR <11 dB; the typical value is 0 dB; P CNST $= -61$.

The values of these parameters for each base station are transmitted on the forward channel in the Access Parameters message. If the access is not successful, the mobile user will increase its power by a predefined PWR STEP. If the maximum power is reached, the mobile user maintains that maximum power.

When the mobile user transmits on the reverse traffic channel, it uses a power level of $P +$ sum of all access probe corrections. After the establishment of communication with the base station, the base station measures the received signal strength and sends a closed-loop power control messages. The sum of all closed-loop power control corrections is added to its previous output power.

The base station transmits separate power control bits for each of the mobile users transmitting on the reverse channel. After receiving a power control bit, the mobile user increases or decrease its power by 1 dB. The mobile user maintains a cumulative sum of all received power control bits to determine the correct power output. The total range of power control is within 24 dB of the open-loop estimated power.

13.9 TD-SCDMA TECHNOLOGY

Time Division–Synchronous Code Division Multiple Access (TD-SCDMA) is one of IMT-2000 3G celllar standards. It uses the Time Division Duplexing (TDD) scheme instead of Frequency Division Duplexing (FDD) scheme as used in WCDMA standard. By dynamically adjusting the number of time slots used for uplink and downlink, the TD-SCDMA system can more easily accommodate asymmetric traffic with different data-rate requirements on downlink and uplink than FDD based WCDMA systems. Since it does not require a paired frequency spectrum for uplink and downlink, the flexibility in spectrum allocation is also increased. By using TDD, different time slots within a single TDMA frame on a single carrier frequency are used to provide both uplink and downlink transmissions. Also, using the same carrier frequency for uplink and downlink means that the wireless channel condition is the same on both directions. The base station can deduce the downlink channel information from uplink channel estimates. This information is useful to the application of switched-beam antenna configuration for the cellular network.

> **Facts to Know!**
>
> Compared to WCDMA which is optimised for symmetric traffic and macro cells operation, TD-CDMA is best used for asymmetric traffic in low mobility applications within micro cells or pico cells.

TD-SCDMA is based on spread spectrum technology. The bandwidth of radio channels in TD-SCDMA is 1.6 MHz and use smart antennas based on switched-beam scheme, spatial filtering, and multiuse detection techniques to achieve several times higher spectrum efficiency than GSM. The system uses DS-CDMA technique at 1.1136 Mcps and has an RF channel bit rate of up to 2.227 Mbps. TD-SCDMA standard also uses TDMA in addition to the CDMA. This reduces the number of users in each time slot, which reduces the implementation complexity of switched-beam antenna and multiuse detection schemes. The mobile data subscribers can have a packet data speed of up to 384 kbps. A 5-ms TDMA frame is subdivided into seven uniform time slots which are dynamiacally allocated to either a single high-data-rate subscriber or several low-data subscribers. But due to non-continuous transmission, higher peak transmitter power is needed to obtain the same radio coverage area. The rate at which the power control is implemented is lower, which limits the mobility, and complicates the radio resource management algorithms.

In TD-SCDMA, uplink signals from mobile subscribers are synchronised at the base station receiver, which is achieved by continuous timing adjustments. This increases the hardware implementation complexity in achieving uplink synchronisation. There is a considerable reduction in the interference between mobile subscribers of the same time slot using different codes by improving the orthogonality between the codes. This results in significant increase in the system capacity. Due to asynchronous traffic data requirement, the downlink will utilise more bandwidth than the uplink in case the mobile subscriber downloads a file from the network. TD-SCDMA is currently adopted as 3G mobile communication standard in China.

13.10 CDMA2000 CELLULAR TECHNOLOGY

Facts to Know!

A key component of Cdma2000 systems is a new Packet Core Network (PCN) that allows for the delivery of packet data services with high speed and better security. The Cdma2000 PCN is one of the major evolutions of Cdma2000 systems to all-IP and multi-media architecture.

Cdma2000 is a technology for the evolution of IS-95/IS-95A to 3G services. The Cdma2000 cellular technology provides a seamless and evolutionary high data rate upgrade path for current users of 2G (IS-95) and 2.5G (IS-95A) CDMA technology.

Figure 13.16 illustrates the digital cellular upgrade paths from 2G digital cellular CDMA to 3G standards based on CDMA technology.

Cdma2000 provides enhanced services to IS-95A (cdmaOne) mobile users. It is designed to be forward and backward compatible in mobile phones. Cdma2000 represents a family of standards. The first 3G CDMA air interface is a system called Cdma2000-1xMC (the MC stands for Multi Carrier) or Cdma2000-IxRTT (Radio Transmission Technology). It implies that a single 1.25 MHz radio channel is used. It requires new hardware in base-station controllers, but no new radio interface. Cdma2000-1xRTT system supports an instantaneous throughput of up to 307 kbps, and typical throughput of up to 144 kbps per mobile user. It can also support up to twice as many voice users as the 2G IS-95 CDMA standards. The system uses rapidly adaptable baseband signaling rates and PN code rates for each mobile user through incremental redundancy and multi-level keying. The improved performance is achieved by adding new channel cards at the base station and by appropriate changes in software.

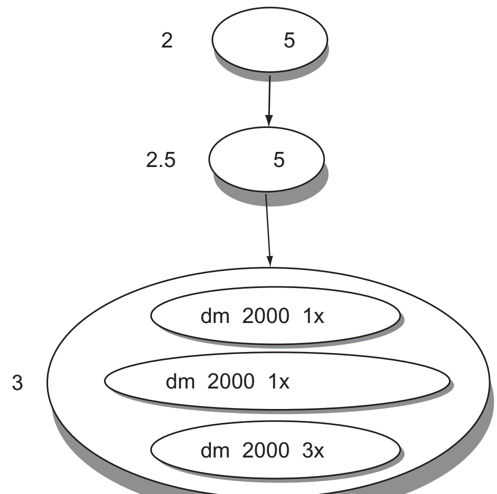

Fig. 13.16 *3G CDMA evolution path*

Multicarrier modulation is applied to 3G voice-oriented cellular networks using CDMA technology. The Cdma2000 employs a multicarrier operation in which a user is allowed to use 1, 3, 6, or 9 of the cdmaOne channels to support a more reliable voice and variety of data channels. The main advantage of using multicarrier modulation is to provide higher data rates for data application, wider bandwidth and consequently better voice quality, and backward compatibility with the existing CDMA systems.

Cdma2000-1xEV is an evolutionary advancement for high-data-rate applications. Using 16-QAM (16-Quadrature Amplitude Modulation) digital modulation technique, the maximum data rate is increased to 2.4 Mbps on the downlink, and 307.2 kbps on the uplink. The system can uses QPSK, 8-PSK or 16-QAM modulation on a single 1.25 MHz channel. The capacity is lowered in case of large interference because 16-QAM requires a better signal quality. As the signal becomes weaker, the modulation level drops down to 8-PSK or QPSK. The uplink is more flexible, using TDMA as well as CDMA to divide a channel into 240 time slots, each of which can be allocated to a different mobile user.

Cdma2000-1xEV has the option of providing the mobile users with dedicated services like only (Cdma2000-1xEV-DO) or data and voice (Cdma2000-1xEV-DV) both. Using Cdma2000 1xEV technology, individual 1.25 MHz channels may be installed in The Cdma2000-1xEV-DO system that dedicates the base station radio channels strictly to mobile data users, and supports greater than 2.4 Mbps of instantaneous high-speed packet throughput per mobile user on a particular CDMA channel. Cdma2000 1xEV-DV supports both

Facts to Know!

Actual user-data rates are typically much lower and are highly dependent upon the user capacity, the propagation conditions, and vehicle speed. Typical mobile users may experience throughputs of the order of several hundred kbps only, which is still sufficient to support web browsing, email access, and m-commerce applications.

voice and data users, and can offer usable data rates of up to 144 kbps with about twice as many voice channels as IS-95B with improved quality of service.

The Cdma2000 3xRTT standard uses three adjacent 1.25-MHz radio channels that are used together to provide instantaneous packet data throughput speeds in excess of 2 Mbps per user, although actual throughput depends upon cell loading, vehicle speed, and propagation conditions. Three non-adjacent radio channels may be operated simultaneously and in parallel as individual 1.25-MHz channels or adjacent channels may be combined into a single 3.75-MHz channel.

EXAMPLE 13.2 Cdma2000 technical specifications

List the important technical parameters of Cdma2000 technology.

Solution

Frequency band	Any existing band
Minimum frequency band required	1X: 2×1.25MHz; 3X: 2×3.75 MHz
Chip rate	1X: 1.2288 Mcps; 3X: 3.6864 Mcps
Maximum user data rate	1X: 144 kbps - 307 kbps; 1X EV-DO: 384 kbps - 2.4 Mbps; 1X EV-DV: 4.8 Mbps
Frame length	5ms, 10ms or 20ms
Power control rate	800 Hz
Spreading factors	4 ... 256 Uplink

The Cdma2000 can use three separate carriers within a single 5-MHz channel, each modulated with a chipping rate of 1.2288 Mbps. This is the same rate as used with IS-95, so this variation essentially uses three IS-95 signals in one larger channel, combining the data from the three to give some combination of greater numbers of voice calls and higher-speed data transmission. In fact, the three carriers do not have to be carried in contiguous spectrum; three 1.25-MHz channels at different points in the spectrum can be used. This is to allow for backward compatibility with the existing IS-95 system. The Cdma2000 system can also use a single carrier operating at a higher chipping rate for new systems where backward compatibility is not required.

Cdma2000 builds on the inherent advantages of CDMA technologies and introduces other enhancements, such as Orthogonal Frequency Division Multiplexing (OFDM), advanced control and signaling mechanisms, improved interference management techniques, end-to-end Quality of Service (QoS), and new antenna techniques such as Multiple Inputs Multiple Outputs (MIMO) and Space Division Multiple Access (SDMA) to increase data throughput rates and quality of service, while significantly improving network capacity and reducing delivery cost.

13.10.1 Forward and Reverse Channels

In W-CDMA, the cell-sites can operate in an asynchronous manner that obviates the need of GPS availability to synchronise base stations. W-CDMA employs what is known as the Orthogonal Variable Spreading Factor (OVSF) codes. OVSF codes allow a variable spreading factor technique that maintains orthogonality between spreading codes of different lengths.

> **Facts to Know!**
>
> QPSK modulation is employed before spreading with the Walsh codes to increase the number of usable Walsh codes.

Cdma2000 employs multiple carriers to provide a higher data rate compared with W-CDMA. It employs N carriers ($N = 1, 3, 6, 9, 12$) for an overall chip rate of N times 1.2288 Mcps which is 3.6864 Mcps for $N = 3$. This mode of operation is suitable for overlaying Cdma2000 over existing IS-95 systems. Walsh codes from 128 chips to 4 chips are employed to provide variable spreading and processing gains. All N carriers use the same single code for scrambling. The cell-sites still need to be synchronised and use the PN-code offsets for differentiation as before. Pilot channels are used for fast acquisition and hand-off as before. In addition to the pilot, synch, and paging channels, auxiliary pilot channels can be used to supply beam-forming information if smart antennas are employed. To support QoS at different rates, a fundamental channel for signaling and a supplemental channel for traffic can be made available. Turbo codes are employed on the forward supplemental channels for high data rates.

Support for variable data rates and operation in a variety of environments once again governs the implementation of the reverse link for 3G systems. In W-CDMA, Gold codes and short codes are used for scrambling on the uplink. The periodicity of the Gold code is 38,400 chips for using a RAKE receiver in the cell-site and that of the short codes is 256 chips for employing multiuser detection.

In cdma2000, the reverse link is made more symmetrical with the forward link in many aspects. For instance, a reverse pilot channel is employed between each mobile user and the cell-site for initial acquisition, time tracking, and power control measurement. More powerful codes are used such as a rate convolutional code with a constraint length of 9. Turbo codes are employed on the reverse supplementary channels. Variable rate spreading is supported to enable better error-correction capability and a variety of data rates.

13.10.2 Handoff and Power Control

Cdma2000 is very similar to IS-95 in terms of hand-off procedures and power control. In W-CDMA, the hand-off procedure is somewhat different. Once again, different sets of pilots are maintained, and the active set corresponds to the pilot channels being used for completing the call. Relative threshold values based on average signal strength levels are employed instead of absolute instantaneous samples.

In W-CDMA, a fast power control scheme is used at 1,500 bps as compared with 800 bps with IS-95 and cdma2000. In WCDMA, closed-loop power control is implemented in a manner similar to IS-95 with the power control bits transmitted 1,500 times a second. This allows a very fast control of power and provides significant capacity gains in W-CDMA, especially at pedestrian speeds. Both inner and outer loop power control mechanisms are employed.

EXAMPLE 13.3 | Key Features of Cdma2000

List the key features of Cdma2000 digital cellular technology.

Solution

(a) *Leading performance* The performance of Cdma2000 in terms of data-speeds, voice capacity and latencies outperform other competing technologies.

(b) *Efficient use of spectrum* Cdma2000 technologies offer the highest voice capacity and data throughput using the minimum frequency spectrum.

(c) *Support for advanced mobile services* Cdma2000-1xEV-DO enables the delivery of a broad range of advanced services, such as high-performance video telephony, multimedia messaging, VoIP, push-to-talk, multicasting and multi-player online gaming with rich 3D graphics.

(d) *All-IP* Cdma2000 technologies are compatible with IP having more flexibility and higher bandwidth efficiencies, and ready to support network convergence.

(e) *Devices selection* Cdma2000 offers the broadest selection of mobile devices and has a significant cost advantage compared to other 3G technologies.
(f) *Seamless evolution path* Cdma2000 has a solid and long-term evolution path that is built on the principle of backward and forward compatibility, in-band migration, and support of hybrid network configurations.
(g) *lexibility* Cdma2000 systems have been designed for urban as well as remote rural areas for fixed wireless, wireless local loop (WLL), limited mobility and full mobility applications in multiple spectrum bands, including 450 MHz, 800 MHz, 1700 MHz, 1900 MHz and 2100 MHz.

EXAMPLE 13.4 Advantages of Cdma2000

Mention major advantages of Cdma2000 digital cellular technology.

Solution

(a) Superior voice clarity
(b) High-speed broadband data connectivity
(c) Low end-to-end latency
(d) Increased voice and data throughput capacity
(e) Time-to-market performance advantage
(f) Long-term, robust and evolutionary migration path with forward and backward compatibility
(g) Differentiated value-added services such as VoIP, PTT, Multicasting, Position location, etc.
(h) Flexible network architecture with connectivity to ANSI-41, GSM-MAP and IP-based networks and flexible backhaul connectivity
(i) Application, user and flow-based Quality-of-Service (QoS)
(j) Flexible spectrum allocations with excellent propagation characteristics
(k) Robust link budget for extended coverage
(l) Increased data throughputs at the boundary of the cell
(m) Multi-mode, multi-band, global roaming features
(n) Improved security and privacy
(o) Lower total cost of ownership

EXAMPLE 13.5 Cdma2000 versus W-CDMA Technologies

Summarise the main differences between two major competing 3G standards—the W-CDMA based on the UMTS and the Cdma2000 that is backward compatible with IS-95.

Solution

The main differences can be summarised as follows.

(a) W-CDMA employs 3.84 Mcps chip rate whereas Cdma2000 has multiples of 1.2288 Mcps chip rates to allow greater compatibility with IS-95 (in particular, 3.6864 Mcps).
(b) W-CDMA employs asynchronous operation to enable deploying picocells within buildings where GPS is not available whereas in IS-95 and Cdma2000, the cell sites operate synchronously by obtaining timing from GPS.
(c) The frame length of W-CDMA is 10 ms to ensure small end-to-end delays, though it is 20 ms in Cdma2000.

Table 13.8 describes the main characteristics of the air-interface parameters of the two 3G standards.

The primary requirements of 3G systems are that they should be able to support a variety of application data rates ranging from 384 kbps circuit-switched connections to 2 Mbps packet-switched connections in indoor-operating environments. This means that there must be support for quality of service and operation from microcells to picocells.

Table 13.8 *Characteristics of the major 3G standards*

Parameters	*W-CDMA*	*Cdma2000*
Multiple access technique	DS-CDMA	*Uplink* DS-CDMA *Downlink* MC-CDMA/DS-CDMA
Chip rate	3.84 Mcps	N × 1.2288 Mcps where $N = 1, 3, 6, 9, 12$
Modulation scheme	*Uplink* Dual channel QPSK; *Downlink* QPSK	Uplink: BPSK; Downlink: QPSK
Frame length	10 ms with 15 slots	5, 10, 20, 40, 80 ms
Pilot structure	Dedicated pilots on the uplink and common or dedicated pilots on the downlink	*Uplink* Code-divided continuous dedicated pilot *Downlink* Code-divided continuous common pilot and dedicated or common auxiliary pilots
Spreading modulation	QPSK both directions	*Uplink* M-ary PSK *Downlink* QPSK
Detection	Coherent pilot aided	Coherent pilot aided
Scrambling codes	*Uplink* Short code (256 chips) or long code (38,400 chips Gold code) *Downlink* Gold code	Long code ($2^{42} - 1$ chips) Short code ($2^{15} - 1$ chips)
Channelisation codes	Orthogonal Variable Spreading Factor (OVSF) codes	*Uplink* Walsh codes *Downlink* Walsh or quasi-orthogonal codes
Inter-cell site operation	Asynchronous Synchronous (optional)	Synchronous

Key Terms

- 1xRTT
- 3GPP
- 8-PSK
- Air Interface
- Cdma2000 IxEVDO
- Cdma2000 IxRTT
- cdmaOne
- Enhanced Data rates for GSM Evolution (EDGE)
- Forward channel
- General Packet Radio Service (GPRS)
- High-speed circuit switched data
- IMT-2000
- Latency
- Orthogonal Frequency Division Multiplexing (OFDM)
- Packet
- Radio frequency spectrum
- Reverse channel
- Second-Generation plus (2.5G)
- TD-SCDMA
- Third Generation (3G)
- Universal Mobile Telecommunications System (UMTS)
- Widebend CDMA (W-CDMA)

Summary

This chapter has presented an overview of current and future third-generation cellular systems that are the evolution of 2G digital cellular standards GSM and CDMA. Various 2.5G cellular standards such as GPRS, EDGE, and cdmaOne are presented here. The recent development of 3G mobile cellular systems (IMT-2000) represents an attempt to create a global cellular standard. Although some countries (including Japan) have already introduced the 3G standard W-CDMA, it is still in the commissioning phase and

may take some time before it is extended throughout the world. The UMTS initially use the existing GSM/GPRS infrastructure and offer only moderate data rates. Several future wireless technologies have also been introduced. Countries use different networks and technologies as backbones for wireless communications. The field of wireless devices and technology is rapidly changing and moving towards integrated services with seamless connectivity among various developed networks. In brief, the future of wireless and mobile communication systems is very promising, and numerous applications will keep on emerging as potential users of this ever-growing technology.

Short-Answer Type Questions with Answers

A13.1 How can GPRS achieve raw data throughput of 171.2 kbps
GSM has a raw uncoded data throughput of 21.4 kbps. When all eight time slots of a GSM radio channel are dedicated to GPRS, an individual user is able to achieve as much as 171.2 kbps (8 × 21.4 kbps).

A13.2 Highlight packet-data transfer capabilities of GPRS in comparison with GSM data services.
GPRS allows more efficient packet-data transfer compared to GSM data services. GPRS uses an error-detection and retransmission scheme to ensure that data packets are correctly delivered to the mobile user. In GPRS, a mobile user can be constantly connected to the network without occupying any radio resource until a data packet has to be transferred on a temporary assigned channel. The channel is quickly released after completion of data transfer. GPRS allows a single user to use more than one time slot and many mobile users to share the same timeslot.

A13.3 Distinguish between Intra-PLMN backbone network and Inter-PLMN backbone network in GPRS.
Intra-PLMN backbone network allows the SGSNs and GGSNs of one service provider to communicate with each other via the G_n interface. It uses a private IP network to ensure the performance and security of the GPRS system. The Inter-PLMN backbone network allows the SGSNs and GGSNs of different service providers to communicate with one another via G_p interface. This is an IP network based on the public Internet or a private IP network using leased lines.

A13.4 How is higher data rates accomplished in EDGE
EDGE uses a new digital modulation format, 8-PSK (octal phase shift keying), in addition to GMSK modulation. EDGE allows for nine different air interface formats, known as multiple Modulation and Coding Schemes (MCS), with varying degrees of error control protection. Each MCS state may use either GMSK (low data rate) or 8-PSK (high data rate) modulation technique for network access, depending on the instantaneous user demands and the operating conditions. This allows EDGE to achieve much higher bit rate across the air interface.

A13.5 What are the major functions of the RNC in TRAN Architecture
The major functions of the RNC include radio resource management, UMTS radio link control sublayers functions execution, serving RNS relocation, Intra-UTRAN hand-off, frame synchronisation, macro diversity combining, and outer-loop power control.

A13.6 What is the significance of Transport logical channels
Transport logical channels are the services offered by the physical layer to the higher layers. Transport logical channels signify as to how the information is transmitted on the radio interface such as in-band identification of the UEs when particular UEs are addressed.

A13.7 Differentiate between cdma ne and W-CDMA technologies.
cdmaOne automatically sends every bit of information 64 times. W-CDMA employs variable or selectable direct sequence spread spectrum chip rates that can exceed 16 Mcps per user. W-CDMA adjusts the

gain depending on the received signal strength. Every bit is sent between 4 and 128 times, which means that greater bandwidth is available in areas with a stronger signal. The other major difference between cdmaOne and W-CDMA is the need for time synchronisation. W-CDMA has been designed to operate without GPS clock signals. Combined with the same QPSK modulation as used in cdmaOne, these give a maximum data rate of around 4 Mbps per channel per cell. The W-CDMA system uses a slightly different coding technique called Gold codes.

A13.8 What is meant by the term multirate' in W-CDMA

The term 'multirate' refers to the provision of multiple fixed-data-rate logical channels to a given user, in which different data rates are provided on different logical channels. With multirate, the system can support multiple simultaneous applications from a given mobile user and can efficiently use available capacity. It can be achieved with a TDMA scheme within a single CDMA channel, in which a different number of time slots per frame are assigned to achieve different data rates.

A13.9 List some techniques which are deployed to improve capacity in the W-CDMA system.

The W-CDMA system provides enhanced capacity due to improved coding gain, voice activity, cell sectorisation, reuse of the same spectrum in every cell, wideband transceivers, support for hierarchical cell structures, adaptive antenna arrays and multi-user detection.

A13.10 What is the main advantage of using multicarrier modulation

The main advantage of using multicarrier modulation in 3G Cdma2000 standard is to provide higher data rates for data application, wider bandwidth and consequently better voice quality, and backward compatibility with 2G CDMA systems.

A13.11 What are the additional enhancements applied in Cdma2000

Orthogonal Frequency Division Multiplexing, advanced control and signaling mechanisms, improved interference management techniques, end-to-end quality of service, and new antenna techniques such as Multiple Inputs Multiple Outputs (MIMO) and Space Division Multiple Access (SDMA) are additional enhancements leading to increase data throughput rates and quality of service, while significantly improving system capacity.

A13.12 How does FDM minimise the impact of frequency selective fading

Frequency selective fading affects some subchannels only and not the complete signal in OFDM. If the data stream is protected by a forward error-correcting code, the impact of frequency selective fading can be minimised. Moreover, OFDM overcomes intersymbol interference in a multipath environment.

A13.13 How is power control mechanism different in IS-95 Cdma2000 and W-CDMA

In W-CDMA, a fast power control scheme is used at 1,500 bps as compared with 800 bps with IS-95 and Cdma2000. In WCDMA, closed-loop power control is implemented in a manner similar to IS-95 with the power control bits transmitted 1,500 times a second. This allows a very fast control of power and provides significant capacity gains in W-CDMA, especially at pedestrian speeds. Both inner and outer-loop power control mechanisms are employed.

Self-Test Quiz

S13.1 GPRS is an overlay on top of the physical layer and network entities.

(a) AMPS
(b) ETACS
(c) GSM
(d) IS-95

S13.2 EDGE is a new radio interface technology with enhanced modulation, and increases the GPRS data rates by up to

(a) three times
(b) four times
(c) six times
(d) eight times

S13.3 When EDGE uses 8-PSK modulation without any error protection, and all eight time slots of a GSM radio channel are dedicated to a single mobile data user, a raw peak throughput data rate of can be provided.
(a) 22.8 kbps (b) 171.2 kbps
(c) 384 kbps (d) 547.2 kbps

S13.4 The frame length in W-CDMA standard is
(a) 10 ms (b) 20 ms
(c) 30 ms (d) 40 ms

S13.5 The data modulation used in reverse channel in W-CDMA system is
(a) BPSK (b) QPSK
(c) dual channel QPSK (d) OQPSK

S13.6 On the W-CDMA uplink, the spreading factor can be up to
(a) 64 (b) 128
(c) 256 (d) 512

S13.7 The channel is the logical control channel specified on the reverse link in W-CDMA system.
(a) sync (b) access
(c) paging (d) pilot

S13.8 The W-CDMA system uses for paging and sync channels.
(a) 4-kbps (b) 8-kbps
(c) 16-kbps (d) 64-kbps

S13.9 The W-CDMA base station transmitter must maintain its frequency to within at 2000 MHz.
(a) 10 Hz (b) 100 Hz
(c) 200 Hz (d) 500 Hz

S13.10 In closed-loop power control, the base station sends power control messages to the mobile user about once every
(a) one millisecond (b) ten milliseconds
(c) hundred milliseconds (d) one second

S13.11 The total range of power control is within of the open-loop estimated power.
(a) 4 dB (b) 12 dB
(c) 20 dB (d) 24 dB

S13.12 Cdma2000-1xRTT system supports a typical throughput of up to per mobile user.
(a) 115 kbps (b) 144 kbps
(c) 384 kbps (d) 2 Mbps

Answers to Self-Test Quiz

S13.1 (c); S13.2 (a); S13.3 (d); S13.4 (a); S13.5 (a); S13.6 (d); S13.7 (b); S13.8 (c); S13.9 (b); S13.10 (a); S13.11 (d); S13.12 (b)

Review Questions

Q13.1 What is the difference between GSM and GPRS? Describe the functions of the network elements in GPRS that are different from GSM.

Q13.2 How is data transfer handled in GPRS architecture? How is data routing done and in what respect is it different from voice routing?

Q13.3 Describe briefly at least four applications and limitations of GPRS network.

Q13.4 List some key characteristics that distinguish third-generation cellular systems from second-generation cellular systems.

Q13.5 How are Walsh codes employed in the cdmaOne forward and reverse channels? Explain the difference.

Q13.6 Why does W-CDMA use Walsh codes in forward and reverse channels for separating users, while cdmaOne uses them only in the forward channel?

Q13.7 List the following parameters for all major 2.5G, and 3G cellular mobile systems standards:
a) RF channel bandwidth
b) Type of modulation
c) Peak data rate
d) Typical data rate
e) Maximum number of concurrent users

Q13.8 Compare and contrast the various 2.5G and 3G cellular technology paths that each of the major 2G cellular standards provide.

Q13.9 There are two different technology paths towards development of 3G cellular standards: one is based on GSM and the other is based on CDMA technology. Which of these two technology paths is the easiest to implement in existing mobile subscriber handsets?

Q13.10 State which among UMTS or Cdma2000 technology would be best suited for real-time network access, and highest internet access speed.

Q13.11 Describe 3G networks based on Cdma 2000 technology. How is a 3G network different from a 2G CDMA network?

Analytical Problems

P13.1 Find the maximum raw instantaneous data rate that can be provided using IS-95B if four user channels are dedicated to a single user.

P13.2 Determine the maximum raw instantaneous data rate that can be provided to a single user in EDGE, assuming that a single time slot on a single GSM channel is available.

P13.3 In an omnidirectional CDMA cellular system, the required E_b/N_o is 10 dB. It is required to accommodate 250 number of users, each with a baseband data rate of 13 kbps. Determine the minimum channel chip rate of the spread spectrum sequence if the voice activity is ignored.

P13.4 Repeat P13.3 if a voice activity factor of 0.4 is considered.

P13.5 Consider a CDMA system which has a bandwidth of 1.25 MHz and transmits baseband data at 9.6 kbps rate. The system uses QPSK modulation and FEC convolution coding. The system requires E_b/N_o of 7 dB. Compute the number of subscribers that can be supported by the system. Assume a 3-sector antenna configuration with an effective gain of 2.6, power-control accuracy factor of 0.9, and an interference factor of 0.5. Ignore voice activity factor.

P13.6 A CDMA system has a bandwidth of 1.25 MHz and transmits baseband data at 9.6 kbps rate. If 40 number of users can simultaneously establish communication links, what is the bandwidth efficiency of the system?

P13.7 A WCDMA transmission has a code-length of 8 per symbol. The chipping rate is 3.84 Mcps. What is the channel data rate?

P13.8 If the code length is 256 chips per symbol, determine the time interval between symbol. Given that WCDMA transmission has a chip rate of 3.84 Mcps.

P13.9 In WCDMA transmission, 2 bits are paired during modulation. If the channel data rate is 480 ksymbols/s, find the effective transmission data rate.

P13.10 Data with FEC is transmitted through a wireless channel at the rate of 9.6 kbps. If redundant bits inserted are three times more than the original data bits, compute the rate at which the receiver actually receives the decoded data.

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There has been rapid evolution of commercially available mobile computing devices and networks in recent times. Users have a variety of wireless networking standards and products to choose from. There are shorter range products that typically provide wireless networking services within an office building, and long-distance products used to send data between buildings several kilometres away. There are several viable but non-interoperable wireless networking technologies, standards and products, each targeting a mobility environment and providing a distinct quality of service. This chapter deals with current and emerging wireless network technologies such as IEEE 802.11 WLAN family of standards, ETSI HIPERLANs, IEEE 802.15 WPAN Standards, IEEE 802.16 WMAN technology. An overview of MANETs and WSNs is also given. This chapter concludes with discussions on Mobile IP and security requirements in wireless networks.

14

Emerging Wireless Network Technologies

14.1 IEEE 802.11 WLAN TECHNOLOGY

The Wireless Local Area Network (WLAN) aims to provide wireless data connectivity within a smaller area such as a building, or an office/college campus. WLANs have gained immense popularity and are now available on most laptops and several high-end PDAs. The low cost, ease of installation, and almost no maintenance requirement have resulted in several businesses looking at the WLAN as a convenient corporate solution. WLAN technology offers many unique features such as license-free 2.4 GHz ISM band operation, omnidirectional robust transmission with low radiations, simplified spontaneous operation without much complicated set-up requirements, plug-and-play basis, simple to use operation, in-built encryption mechanisms, maintaining user privacy, support for location aware applications, and battery-operated low-power devices.

Compared with wired LAN, WLAN operates in a hostile wireless medium for communication with an additional requirement to support mobility and security. The wireless medium has serious bandwidth limitations, frequency regulations, location- and time- dependent multipath fading, and interference from other WLANs or wireless devices operating in its near vicinity. Moreover, WLANs are not restricted to any physical boundaries, and they overlap with each other. The 802.11 standard needs to examine management of connection, link reliability, power, and link security. The IEEE standard 802.11 specifies a family of WLANs which define the detailed specifications of physical and MAC layers to accommodate any connectionless data-oriented networks for wireless wideband local access.

Facts to Know!

The FCC does impose transmission power limits on wireless devices using the unregulated ISM bands, which in effect reduces their coverage. This is one of the main reasons not to use these frequencies in long-range walkie-talkie wireless communication links.

14.1.1 WLAN Advantages and Disadvantages

There are a number of advantages of wireless LAN over wired LAN, which are summarised as below:

Quicker Deployment Adding new subscribers to existing WLAN facility does not involve making physical connections or running cables or patching new jacks.

LAN Extension The WLAN can provide services to inaccessible areas by extending the wired LAN Internet connectivity. Thus, it is economical in situations where the site is not conducive to LAN wiring because of building or budget limitations, such as older buildings, leased space, or temporary locations.

Easy Access To hook up to the existing network, the user just needs to enable the wireless device to access the available WLAN services as per the network configuration. There is no need of connecting any cable or finding the plug.

WLAN Mobility The WLAN mobile users can access the network from anywhere within the radio coverage area of the network at any time. There is no line-of-sight requirement with the wireless access points till the received signal strength is good as the radio waves can penetrate the obstacles. For example, students attending class on a campus access the Internet or exchange information for learning.

Cost Effective After the initial cost of installation and commissioning of the network, there is no extra cost incurred in the infrastructure for providing services to additional WLAN users. Wireless technology saves a lot of expenses on cabling, labour, and maintenance. Most WLAN devices are equipped with plug-and-play feature. This helps to reduce the cost due to vendor technical installation, equipment maintenance and to eliminate equipment redundancy in case of system crash.

Smart Working Senior executives or managers can present their briefings using WLAN, without carrying the data files, charts, and any other storage material. The ad-hoc network can be configured for peer-to-peer communication.

Increased Productivity Wireless technologies have a direct impact on real-time increase in productivity due to minimum set-up requirements with centralalised database access.

There are certain disadvantages of wireless LAN such as the following:

Limited Bandwidth Due to limited bandwidth and data-rate capability, the WLAN technology cannot support video teleconference or real-time applications. The WLAN is not capable to download and upload large data files quickly.

Incompatibility WLAN devices from different manufacturers are often incompatible with each other as well as with other networks due to usage of a number of different standards. The problem has been the lack of interoperability among WLAN products from different manufacturers.

Interference and Loss of Signal The wireless link is not always reliable due to external noise and interference. The received signal strength may vary significantly, resulting in sudden disconnections or heavy errors. Also, there is interference to other wireless networks operating in the near vicinity.

Black or Dead Spots Due to the type of surrounding structure (steel reinforcing materials) or high-power running machinery, there may be some spots where there is zero or very weak radio signal strength, thereby rendering the services inaccessible.

Less Security Due to insecure wireless medium, the network is more vulnerable for malicious attacks and jamming.

Need of Backbone Network Wireless transmission is slower, less efficient and less reliable compared to wired networks. The operation of WLAN is dependent on the reliable wired backbone network.

14.1.2 IEEE 802.11 System Architecture

The IEEE 802.11 WLAN standard could be used to provide communication between a number of subscriber terminals as a client/server wireless configuration in infrastructure mode, or as an ad-hoc network using peer-to-peer mode, or a fairly complicated distributed network. The IEEE 802.11 standard defines two basic modes of system architectures to provide connectivity to wireless terminals:

– An infrastructural mode, where a number of wireless terminals are either configured as a client/server mode via a wireless LAN Access Point (AP), or as a distributed wireless network connected to a wired LAN via a number of access points that acts as gateways between the wireless terminals and the wired network. In this mode of operation, the number of wireless terminals are connected to a backbone network through wireless access points.
– An infrastructure-less ad-hoc mode, where wireless terminals do not require the presence of an access point, and form a network by directly communicating, and co-coordinating, with each other to exchange information through the system. In the ad-hoc system architecture mode, wireless terminals communicate in a peer-to-peer basis.

The key components for all these system architectures are wireless Network Interface Cards (Wireless NIC) installed within wireless terminals and WLAN access points. The WLAN interface cards could be operated in continuous aware mode (radio always on) or power-saving polling mode (radio in sleep state to save more power) in which the AP keeps data in its buffer for the wireless terminals and sends a signal to wake them up.

Each access point has a radio coverage area, that is, a limited range of operation, which is typically 200–500 metres in an open environment. The AP is usually placed high on the side walls on the interiors of a building or at the ceiling of a room/corridor and supports a large number of (usually, 115 to 250) wireless-terminal users transmitting, receiving, and buffering data between the WLAN and the wired network. Wireless terminals operating within an access point's coverage area are capable of receiving signals from that access point.

In the client/server configuration, many wireless terminals such as laptops equipped with wireless interface cards are physically close to each other, typically 20 to 500 metres. They can be linked to a common AP which functions as a central hub that serves as a bridge between them and the existing wired LAN. The wireless access cards installed as an add-on unit with the wireless terminals provide the interface between the PCs and the antenna, while the AP serves as the WLAN hub.

Figure 14.1 illustrates the infrastructure-based WLAN system architecture as client/server wireless configuration.

Figure 14.2 illustrates the infrastructure-based WLAN system architecture as a distributed wireless network configuration. As indicated in the figure, an access point may be implemented as part of a wireless terminal. In fact, the AP may be a logic within a wireless terminal that provides access to the DS by enabling DS services in addition to acting as a wireless terminal. The AP provides access to a Distribution System (DS) through the wireless medium to a number of wireless terminals located within the radio coverage of the AP. It is essential that all participating wireless terminals must execute the same MAC protocol and compete

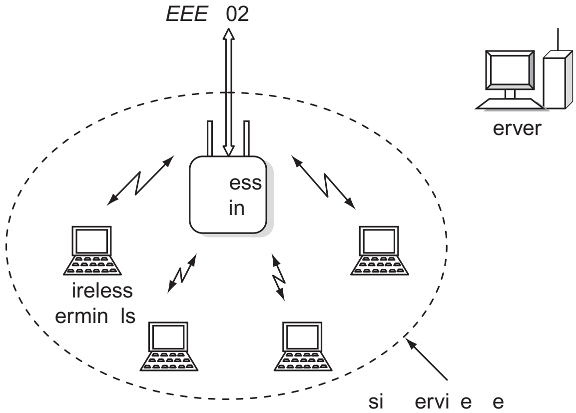

Fig. 14.1 *Infrastructure-based WLAN client/server architecture*

for access to the same shared wireless medium using CSMA/CA protocol. The access-point, together with the wireless terminals associated with it operating within its radio coverage, form a Basic Service Set (BSS). In a BSS, wireless terminals do not communicate directly with one another.

If one wireless terminal in the BSS wants to establish communication with another wireless terminal in the same BSS, the MAC frame is first sent from the calling wireless terminal to the AP, and then from the AP to the called wireless terminal. Thus, the AP can be thought of functioning as a relay point, and a BSS can be referred as a cell. Thus, the two wireless terminals that are communicating within the same BSS get the call routing service from the single AP of that BSS. Distribution services are provided between BSSs; these services may be implemented in an AP or in another special-purpose device attached to the DS.

To integrate the IEEE 802.11 architecture with a traditional wired LAN, a portal is deployed. The portal logic is implemented in a device such as a bridge or router, which is a part of the wired LAN and attached to the Distribution System (DS). A BSS is also connected to a backbone DS through an access point. The DS can be a switch, a bridged IEEE wired LAN, or another wireless network. Typically, the DS is a wired backbone LAN but can be any communication network. A distribution system connects several BSSs via the AP to form a single network and thereby extends the wireless coverage area.

In other words, the collection of BSSs connected by a wired network (also called a distribution system) is known as an Extended Service Set (ESS). The distribution system connects the wireless networks via the APs with a portal, which forms the interworking unit to other LANs. Thus, an ESS consists of two or more basic service sets interconnected by a distribution system. The ESS has its own unique identifier termed as the ESSID. The ESS appears as a single logical LAN to the logical link control layer.

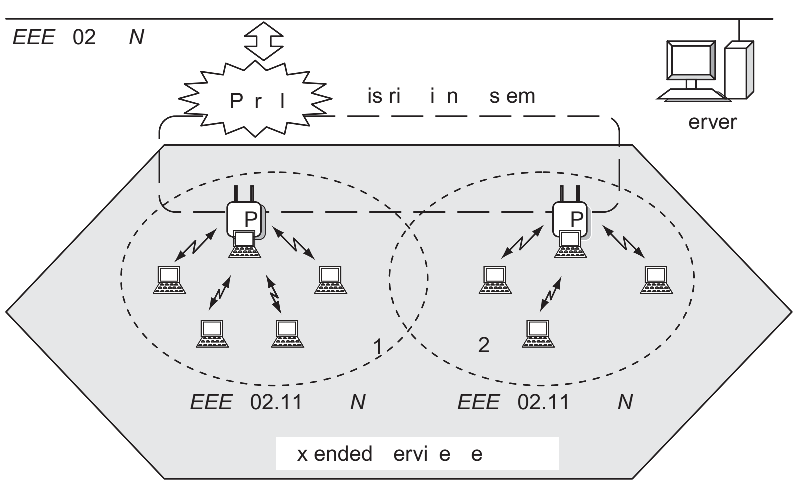

Fig. 14.2 *Infrastructure-based WLAN distributed system architecture*

If a wireless terminal in one BSS wants to establish communication with another wireless terminal located in a different BSS, the MAC frame containing the ESSID is first sent from the calling wireless terminal to the AP of its home BSS, and then relayed by the AP over the DS on its way to the destination BSS and then finally called to remote wireless terminal via its AP. Thus, the AP can be thought of functioning as a bridge as well as a relay point, and an ESS can be referred as a cellular network. The ESSID is used to identify different networks and may be referred as the name of a network itself. Without knowing the ESSID, the wireless terminals cannot participate in the WLAN.

Facts to Know!

It is possible to use a standard PC having a standard NIC (which serves as the wired network interface), as an access point by installing a wireless NIC (which functions as transmitter/receiver), and special software that allows the PC to serve as an AP as well as a security firewall.

EXAMPLE 14.1 Distribution service in WLAN

Suppose a data frame is to be sent from the wireless terminal 3 (WT3) to the wireless terminal 8 (WT8) as shown in Fig.14.3. Describe the step-by-step procedure for routing of the data in WLAN infrastructure architecture.

Solution

The IEEE 802.11 MAC frame format contains all the necessary fields including source and destination WT addresses, ESSID, and a host of control and management bits.

Step 1. *The data frame is sent from the source WT3 to WT1 which is also the AP1 for its home BSS1.*

Step 2. *The AP1 forwards the frame to the DS to which it is connected.*

Step 3. *The DS applies the search algorithm to find the BSS to which the destination WT8 is associated with. As shown in the given figure, the destination WT8 is associated with BSS3.*

Step 4. *The DS directs the frame to AP3 associated with WT6 in the target BSS3.*

Step 5. *The WT6 receives the frame and forwards to the destination WT8.*

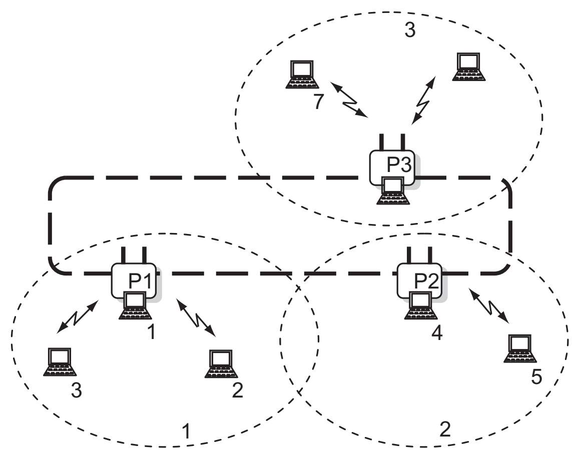

Fig. 14.3 *Illustration of distribution service in WLAN*

Hence, the route for transmission of frames from WT3 to WT8 is
WT3 → (WT1 + AP1) → DS → (AP3 + WT6) → WT8

The data can be transferred between a wireless terminal of a BSS on an IEEE 802.11 WLAN and a host terminal connected to an integrated IEEE 802 wired LAN. Both wireless LAN and wired LAN are logically connected to the same DS and the communication is possible by the integration service provided by DS. The requirement of any address translation and media conversion logic for the exchange of data is taken care of by the integration service.

It is mandatory for each wireless terminal to establish its identity with other wireless terminals. The IEEE 802.11 standard defines mutually acceptable authentication service among wireless terminals as well as deauthentication service to terminate an existing authentication service. The standard also provides for the optional use of encryption with a shared-key RC4 algorithm, to assure privacy of the contents of information from being decoded by unintended wireless terminals.

Multiple access points are typically installed in order to provide seamless, continuous connectivity to wireless terminals as they move from one location to another. Similar to the cellular structure, several APs installed in the building at various strategic locations can serve a larger area. There may be overlapped access areas. The APs track the movement of wireless terminals within a coverage area and decide on whether to allow them to communicate among themselves. For this purpose, the IEEE 802.11 standard provides for a hand-off mechanism, in order to support the transfer of a wireless terminal from one access point to another, as the wireless terminal moves between the respective coverage areas of the two access points. Wireless terminals can select an AP and get associated with it. To deliver a message within a DS, the distribution service needs to know where the destination wireless terminal is located. Specifically, the DS must know the identity of the AP to which the data should be delivered for forwarding it to the destination wireless terminal. To meet this requirement, each wireless terminal must maintain an association with the AP within its serving BSS. The AP can then communicate the association information to all other APs within the ESS to facilitate routing and reliable delivery of addressed frames.

Facts to Know!

It is recommended that the power supplies required for providing power to the APs via Ethernet media should only be purchased from the same manufacturer as the AP itself.

The APs support roaming, and the distribution system handles data transfer between the different APs. APs provide synchronisation within a BSS, support power management, and can control medium access to support time-bounded service. When a wireless terminal moves from one BSS to another BSS, reassociation service enables an established association to be transferred from one AP to another AP. The wireless terminal should give disassociation notification before leaving an ESS or switching off. Disassociation notification can also be given from an AP that an existing association is terminated.

When a wireless terminal moves away from its access point, the Signal-to-Noise (SNR) of the wireless link drops, and will eventually drop below a certain threshold value known as the BSS search threshold. This event triggers the roaming algorithm to start looking for other access points to associate with. In this process, the wireless terminal initiates a sweeping function that performs a series of scans on different frequencies to construct an updated access point list. When the SNR drops below a second threshold known as BSS switching threshold, the roaming algorithm triggers a re-association by selecting another access point from the access point list. Typically, the access point with the strongest signal strength is selected. When the moving wireless terminal selects an access point from the access point list, it sends an association request, requesting to bind to that particular access point. Assuming that the moving wireless terminal is not denied access, the access point responds with an association response that in effect binds the moving wireless terminal to the access point. It is important to emphasise here that roaming is not the only mechanism that triggers hand-offs. More specifically, access points can be configured to use a particular load-balancing algorithm that initiates hand-offs when the load becomes uneven across multiple overlapping BSSs. However, irrespective of what mechanism triggers hand-offs, they typically rely on physical and MAC layer characteristics.

The IEEE 802.11 standard allows wireless terminals to be handed over from one access point to another, when a wireless terminal moves between the radio coverage areas of the two access points. However, before a wireless terminal can be handed over to a new access point, the wireless terminal should be able to discover the new access point. The 802.11 standard defines 13 different frequency channels of 5 MHz each. The standard allows two modes by which a wireless terminal can detect the presence of an access point.

Passive Scanning Mode A wireless terminal sweeps from channel-to-channel to detect the presence of Beacon frames which are periodically (with a typical period of 100 ms) transmitted with synchronisation information by the access-points. The Beacon frames contain all the information corresponding to the BSS such as ESS ID, beacon interval, capabilities and Traffic Indication Map (TIM) that is needed by a wireless terminal to associate itself with the access point. A wireless terminal can establish the presence of an access point on a channel if it is able to detect a Beacon frame on that channel. The advantage of passive scanning is that the wireless terminal saves battery power because it does not have to transmit anything.

Active Scanning Mode A wireless terminal actively seeks out access points by broadcasting probe request frames on every channel. An access point that receives a probe-request frame responds back to the wireless terminal by sending the probe-response frame. The wireless terminal can establish the existence of an access point on a channel if it receives the probe-response frame on that channel.

Once a wireless terminal has discovered access points in an area, it has to choose an access point with which to associate. The IEEE 802.11 mandates that a wireless terminal be associated with only one access point at a given time. This allows the switches in the wired network to forward the messages meant for a wireless terminal only to the access point that the wireless terminal is associated with. Once the scanning process has finished, the wireless terminal has an updated list of APs in its radio coverage. This information is used by the wireless terminal to associate with the AP that provided a higher Signal-to-Noise Ratio (SNR). Before a wireless terminal can be associated with an access point, it has to authenticate itself to the access point. After the access point sends an acknowledgment verifying the wireless terminal's identity, the wireless terminal sends a re-association request to the new access point. The wireless terminal is considered to be associated with the new access point only after it receives a re-association response from the new access point. The total latency in the entire hand-off process is the sum of the delay in the scanning process to detect an access point, and the delay in authenticating and re-associating the wireless terminal with the new access point.

Ad-hoc WLAN System Architecture In addition to infrastructure-based networks, the IEEE 802.11 standard allows the system architecture of ad-hoc networks among wireless terminals, thus forming one or more Independent BSSs (IBSS) as shown in Fig. 14.4.

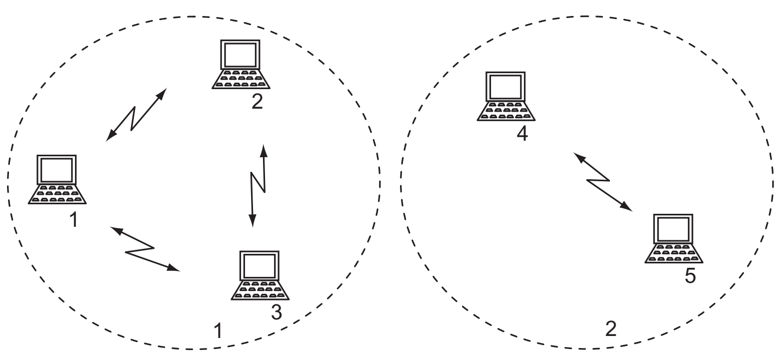

Fig. 14.4 *Infrastructure less ad-hoc-based WLAN system architecture*

In an ad-hoc WLAN system architecture, an IBSS comprises of a group of wireless terminals either using the same carrier transmission frequency without any overlapping of their respective radio coverage areas, or using the different carrier transmission frequencies allowing overlapping of their respective radio-coverage areas. For example, the wireless terminal WT1, WT2, and WT3 are configured within their respective radio-coverage areas forming IBSS1. Similarly, WT4 and WT5 forms IBSS2. This means that WT1 can communicate directly with WT3, but not with WT5 because there is no AP or DS which can connect both IBSSs. As there is no central controller in the ad-hoc architecture, the wireless access cards use the CSMAS/CA protocol to resolve shared access of the wireless medium.

EXAMPLE 14.2 Infrastructure versus ad-hoc WLAN architectures

Compare the salient features, advantages and disadvantages of IEEE 802.11 WLAN standard infrastructure and ad-hoc architectures.

Solution

(1) Many WLANs need infrastructure architecture because these networks provide access to other networks such as standard wired IEEE 802 LAN. In these infrastructure-based WLANs, wireless communication typically takes place between the wireless terminals and the access point only, not directly between the wireless terminals. The access points control medium access, and function as a bridge to other wireless or wired connected networks. The wireless terminals and the access point should be within their radio-coverage areas.

Ad-hoc WLAN architecture, however, does not need any infrastructure to work. Each wireless terminal can communicate directly with other wireless terminals within each other's radio-coverage areas. So for controlling medium access, no access point is necessary. If wireless terminals are not within the same radio range, they simply cannot communicate with each other.

(2) Most of the network functionality in infrastructure-based WLAN architecture lies within the access point. Therefore, the design of the wireless terminals can remain quite simple. Since only the access point controls medium access, no collisions are possible. Different carrier-sense multiple access protocols with or without collision can be used. Collisions may occur if medium access of the wireless terminals and the access point is not coordinated. This architecture may be useful for quality of service guarantees such as minimum bandwidth and specific data rate for certain wireless terminals.

In ad-hoc WLAN architecture, the complexity of each wireless terminal is higher because every terminal has to implement medium access mechanisms including those related to handle hidden or exposed terminal problems, and priority mechanisms to ensure a certain quality of service.

(3) Infrastructure-based WLAN architectures are less flexible in operation, whereas ad-hoc WLAN architecture exhibits the greatest possible flexibility as it can be configured easily in emergency situations without any need of infrastructure.

14.1.3 IEEE 802.11 Protocol Architecture

The IEEE 802.11 standard only covers the physical layer and Medium Access Control (MAC) layer. Figure 14.5 depicts IEEE 802.11 WLAN protocol architecture, showing the physical layer and MAC layer.

IEEE 802.11 Physical Layer The IEEE 802.11 WLAN standard defines three different physical layer specifications depending upon the type of transmission used. The standard allows two types of radio transmissions: Direct Sequence Spread Spectrum (DSSS), Frequency-Hopping Spread Spectrum (FHSS), and one type of infrared transmission.

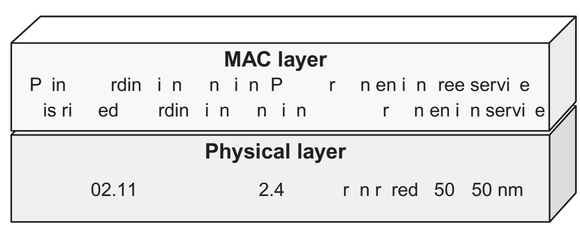

Fig. 14.5 | *IEEE 802.11 WLAN protocol architecture*

- Physical layer based on the radio transmission operating in the ISM band at 2.4 GHz using DSSS baseband modulation technique. The carrier modulation scheme used is Differential Binary Phase Shift Keying (DBPSK) for 1 Mbps transmission data rate, and Differential Quadrature Phase Shift Keying (DQPSK) for 2 Mbps transmission data rate. The symbol rate is 1 Msps, resulting in a chipping rate of 11 Mcps. The maximum transmit power is limited to 1 W in the US, and 100 mW EIRP in Europe. Due to wider bandwidth, DSSS provides a more stable signal and better radio coverage.
- Physical layer based on the radio transmission operating in the ISM band at 2.4 GHz using FHSS baseband modulation technique. The FHSS system defines 79 pseudo-random hopping channels, each with a bandwidth of 1 MHz in the 2.4-GHz ISM band. The carrier modulation scheme used is 2-level GFSK for 1-Mbps transmission data rate and 4-level GFSK for optional 2-Mbps transmission data rate.

This system has a variable packet duration of up to 20 ms. The recommended hop rate of the IEEE 802.11 FHSS system is 2.5 hops per second. The maximum transmit power is limited to 1 W in the US, and 100 mW EIRP in Europe. The FHSS based WLAN system provides a low power simple implementation as compared with DSSS-based WLAN system.

- Physical layer based on the infrared transmission operating near visible light having wavelength at 850 nm–950 nm. The actual transmission uses an intensity modulation scheme, in which the absence of a signal corresponds to binary 0, and the presence of a signal corresponds to binary 1. The IEEE 802.11 infrared physical layer specifies omnidirectional coverage of up to 20-m radius. The standard does not require a line-of-sight between the sender and receiver but should also work with diffuse light. This allows for point-to-multipoint communication. The carrier modulation scheme used is 16-level PPM for 1-Mbps transmission data rate. Each 16-bit string consists of fifteen 0s and one binary 1. The carrier modulation scheme used is 4-level PPM for 1-Mbps transmission data rate. For the 2-Mbps data rate, each group of 2 data bits is mapped into one of four 4-bit sequences, each sequence consisting of three 0s and one binary 1. The peak transmitted optical power is specified at 2 W with an average of 125 or 250 mW.

Table 14.1 summarises the key parameters defined in IEEE 802.11 physical-layer specifications.

Table 14.1 *IEEE 802.11 physical-layer specifications*

Baseband modulation	*Carrier modulation*	*Symbol rate*	*Bits/Symbol*	*Data rate*
DSSS	DBPSK	1 Msps	1	1 Mbps
DSSS	DQPSK	1 Msps	2	2 Mbps
FHSS	2-GFSK	1 Msps	1	1 Mbps
FHSS	4-GFSK	1 Msps	2	2 Mbps
Infrared	16-PPM	4 Msps	0.25	1 Mbps
Infrared	4-PPM	4 Msps	0.5	2 Mbps

EXAMPLE 14.3 Data transfer time in IEEE 802.11 WLAN system

The IEEE 802.11 WLAN system operates at 2 Mbps. Determine the data transfer time of a 20-kbytes file.

Solution

Transmission data rate = 2 Mbps or 2000 kbps (given)

Size of a file to be transferred = 20 kbytes (given)

Using 1 byte = 8 bits, size of the file = 20 × 8 = 160 kb

Therefore, data-transfer time = 160 kb/2000 kbps

Hence, data-transfer time = 80 ms

EXAMPLE 14.4 Size of the file in IEEE 802.11 WLAN system

The IEEE 802.11 WLAN system operates at 2-Mbps data transmission rate. Compute the size of the file transferred in 16 seconds.

Solution

Transmission data rate = 2 Mbps or 2000 kbps (given)

Data transfer time = 16 seconds (given)

Therefore, size of a file transferred = 2 Mbps × 16 seconds

Hence, size of a file transferred = 32 Mb or 4 MB (Mega bytes)

The IEEE 802.11 MAC Layer The basic functions of the MAC layer comprises of medium access control, reliable data delivery, and security. MAC frames can be transmitted between wireless terminals; between wireless terminals and an access point; and between access points over a DS. The IEEE 802.11 WLAN standard defines two types of MAC algorithms for medium access control: Point Coordination Function (PCF) and Distributed Coordination Function (DCF). PCF offers both asynchronous data and an optional time-bounded service types using an infrastructure-based system architecture but needs an access point to control medium access and to avoid contention for time-sensitive or high-priority data exchange. DCF offers only asynchronous service types in an ad-hoc system architecture of peer wireless terminals having bursty data within an IBSS. The asynchronous service supports both broadcast and multi-cast packets based on best-effort delivery mechanism.

The IEEE 802.11 MAC algorithm is also called DFWMAC (Distributed Foundation Wireless MAC) that provides a distributed access control mechanism with an optional centralised control. PCF is a centralised MAC algorithm used to provide contention-free service, and is implemented on top of the DCF, and exploits the features of DCF to assure access to its wireless terminals. The lower sublayer of the MAC layer is DCF which uses a contention algorithm to provide access to all types of traffic data including ordinary asynchronous traffic.

The PCF operation consists of polling in a round-robin fashion to all wireless terminals configured for polling by the centralised polling-point coordinator. The point coordinator can seize the medium and lock out all asynchronous traffic while it issues polls and receives responses. At the beginning of a superframe, the point coordinator may optionally seize control and issues polls for a given period of time. This time interval varies because of the variable frame size issued by responding wireless terminals. The remainder of the superframe is available for contention-based access. At the end of the superframe duration, the point coordinator contends for access to the medium. If the medium is idle, it gains immediate access and a complete superframe period follows. If the medium is busy, it must wait until the medium is idle to gain access.

The DCF sublayer of the IEEE 802.11 MAC layer simply uses the CSMA algorithm. If a wireless terminal has a MAC frame to transmit, it listens to the medium. If the medium is idle, the wireless terminal may transmit. If the medium is busy, the wireless terminal must wait until the current transmission is complete. The DCF does not include CSMA/CD because collision detection is not practical on a wireless network. The dynamic range of the received signals is very large on the wireless medium, and a transmitting terminal cannot effectively distinguish incoming weak signals from noise and the back reception of its own transmissions. DCF includes a set of delays including binary exponential back-off technique to ensure the smooth and fair functioning of the CSMA algorithm even in a heavy traffic situation.

The IEEE 802.11 defines specifically CSMA/CA with binary exponential back-off wait time within a pre-specified contention window technique. To counter errors due to signal fading and interference in a wireless medium, CSMA/CA with ACK or CSMA/CA with RTS/CTS algorithm is usually employed. In CSMA/CA with ACK, the receiving wireless terminal answers directly with an acknowledgement (ACK) of the received MAC frame. If no ACK is received, the sender automatically retransmits the frame. But this may cause further delay and the number of retransmissions is also limited. In CSMA/CA with RTS/CTS algorithm, the standard defines two small control packets: a Request To Send (RTS) packet of 20 bytes, and a Clear To Send (CTS) packet of 14 bytes. If the receiving terminal receives the RTS prior to MAC data frames, it answers with a CTS message. Basically, this mechanism reserves the wireless medium for one sender terminal exclusively for the duration of transmission of its data.

14.1.4 IEEE 802.11 Standards

The IEEE 802.11 is the dominant family of standards for WLAN in the world. The IEEE 802.11 specifies physical layer and MAC layer standards, operating at the 2.4 GHz ISM band with a data rate of 1 or 2 Mbps, protocols, power levels, modulation schemes, and so on. The MAC layer should be able to operate with multiple physical layers, each of which exhibits a different medium sense and transmission characteristic such as infra-red and spread spectrum radio transmission techniques.

IEEE 802.11a WLAN Standards WLAN technology is improving at a fast pace. Future WLAN products could operate at higher frequencies and provide higher bit rates. To meet such demands, the IEEE 802.11 group has added another layer in the 5.2 GHz band, utilising OFDM to provide data rates up to 54 Mbps. This standard, known as the IEEE 802.11a, became the first to use OFDM in packet-based wireless communication. The IEEE 802.11a uses the US 5 GHz UNII (Unlicensed National Information Infrastructure) frequency band. The UNII band is divided into three parts:

(a) The UNII-1 Band 5.15 GHz to 5.25 GHz with maximum power output 50 mW for indoor applications

(b) The UNII-2 Band 5.25 GHz to 5.35 GHz with maximum power output 250 mW either indoor or outdoor applications

(c) The UNII-3 Band 5.725 GHz to 5.825 GHz with maximum power output 1W for outdoor applications

The IEEE 802.11a standard specifies Orthogonal Frequency Division Multiplexing (OFDM), also called multicarrier modulation as baseband modulation scheme. OFDM uses multiple carrier signals at different frequencies, transmitting some of the bits on each channel. All the subchannels are dedicated to a single data source. The IEEE 802.11a specifications supports the use of various coding and carrier modulation schemes. A convolution code at a rate of 1/2, 2/3, or provides forward error correction. The system uses up to 48 subcarriers with frequency spacing of 0.3125 MHz that are modulated using BPSK, QPSK, 16-QAM, or 64-QAM digital modulation schemes.

EXAMPLE 14.5 Achievable data rate in IEEE 802.11a

Consider an IEEE 802.11a WLAN system in which OFDM baseband modulation scheme is used. The OFDM system has 52 subcarriers out of which 4 subcarriers are used as pilot subcarriers and the remaining as data subcarriers. OFDM symbol duration including guard interval for ISI mitigation is 4 μs. If the system uses ¾ FEC code rate and 64-QAM carrier modulation scheme then show that the achievable transmission data rate is 54 Mbps.

Solution

Step 1. *To find number of data subcarriers*

Total number of subcarriers in OFDM system = 52 (given)
Number of subcarriers used as pilot subcarriers = 4 (given)
Hence, number of data subcarriers = 52 − 4 = 48 subcarriers

Step 2. *To find number of data bits transmitted per OFDM symbol*

FEC code rate = (given)
Type of carrier modulation used = 64-QAM (given)
As 64-level or 2^6-QAM technique corresponds to 6 bits per symbol, then
Number of data bits transmitted per OFDM symbol = 6 × × 48
Number of data bits transmitted per OFDM symbol = 216 bits

Step 3. *To compute transmission data rate*

OFDM data symbol duration = 4 μs (given)
Therefore, transmission data rate = 216 bits/4 μs
Hence, transmission data rate = 54 Mbps

Table 14.2 summarises the key parameters defined in IEEE 802.11a physical-layer specifications.

IEEE 802.11b WLAN Standards In order to meet the user demand for higher data rates, the IEEE standard 802.11b (popularly known as Wi-Fi for Wireless Fidelity), is an extension of the IEEE 802.11 DSSS standards

Table 14.2 *IEEE 802.11a physical-layer specifications*

Modulation scheme	*EC coding rate*	*Coding bits per subcarrier*	*Code bits per O DM symbol*	*Data bits per O DM symbol*	*Data rate*
BPSK	1/2	1	48	24	6 Mbps
BPSK	3/4	1	48	36	9 Mbps
QPSK	1/2	2	96	48	12 Mbps
QPSK	3/4	2	96	72	18 Mbps
16-QAM	1/2	4	192	96	24 Mbps
16-QAM	3/4	4	192	144	36 Mbps
64-QAM	2/3	6	288	192	48 Mbps
64-QAM	3/4	6	288	216	54 Mbps

Facts to Know!

Due to use of a single frequency, IEEE 802.11 transmission is half-duplex. Although the maximum transmission rate of 802.11 wireless network is 11 Mbps, the maximum throughput achievable in an 802.11b network is only between 5 and 6 Mbps.

and specifies a physical layer providing a basic rate of 11 Mbps and a fall-back rate of 5.5 Mbps in the 2.4 GHz ISM band. The chipping rate is 11 Mcps and provides the same occupied bandwidth as that of original IEEE 802.11. In order to achieve a higher data rate in the same bandwidth in the same chipping rate, the IEEE 802.11b standard specifies the use of a modulation scheme known as Complementary Code Keying (CCK) is used. The system uses CSMA/CA technique, and the typical coverage is up to 100 metres. The security level is comparatively low.

Table 14.3 summarises the key parameters defined in IEEE 802.11b physical layer specifications.

Table 14.3 *IEEE 802.11b physical-layer specifications*

Baseband modulation	*Carrier modulation*	*Symbol rate*	*Bits/Symbol*	*Data rate*
DSSS	8-CCK	1.375 Msps	4	5.5 Mbps
DSSS	8-CCK	1.375 Msps	8	11 Mbps

EXAMPLE 14.6 IEEE 802.11b CCK modulation scheme

IEEE 802.11b WLAN standard uses the 8-level CCK modulation scheme. Illustrate with the help of a simple functional schematic that the maximum achievable data rate is 11 Mbps.

Solution The data rate up to 11 Mbps is achieved in IEEE 802.11b WLAN standard by using orthogonal coding technique called Complementary Code Keying (CCK). Fig. 14.6 shows a simplified block diagram of the basic principle of the CCK modulation.

t t e K t

- the sequence of input binary data stream is treated in blocks of 256 eight-bit symbols at 1.375 Msps (11 Mbps divided by 8 bits per symbol)
- the encoder maps each eight-bit symbol into 8 four-phase coded symbols
- The 256 orthogonal coded symbols are selected from the $4^8 = 65{,}536$ available symbols

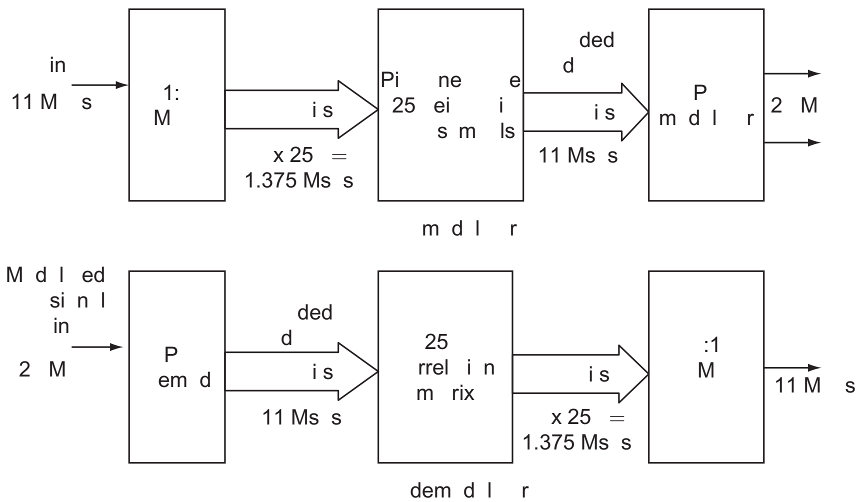

Fig. 14.6 *IEEE 802.11b CCK modulation scheme*

- The I- and Q-elements of the blocks of 8 four-phase coded symbols are transmitted serially using a four-phase QPSK modulator

tt e K e t

- Each eight received complex waveforms from the QPSK demodulator are grouped in a block and sent to the decoder to find the closest eight-bit symbol associated with the demodulated 8 four-phase signals.
- The chip rate and occupied bandwidth is the same as that of the original IEEE 802.11 standard, but the data rate is increased to 11 Mbps.

The IEEE 802.11b standard also specifies an optional alternative to CCK modulation scheme which is known as Packet Binary Convolution Coding (PBCC). Although this scheme requires increased computation at the receiver, yet it achieves more efficient transmission technique with possibility of higher data rates for future enhancements.

IEEE 802.11e WLAN Standards The IEEE 802.11e WLAN standards is concerned with MAC layer enhancements to improve efficiency of polling, channel robustness, quality of service similar to IP telephony and video streaming, and address some security issues. Any wireless terminal implementing 802.11e WLAN standards is referred to as QoS terminal in which the PCF and DCF modules are replaced with a Hybrid Coordination Function (HCF). HCF controlled channel access centrally manages medium access in a more efficient and flexible manner. Enhanced distribution channel access includes priority.

IEEE 802.11f WLAN Standards The IEEE 802.11f WLAN standards addresses interoperability issue among Access Points (APs) manufactured by different companies. This standard facilitates the roaming of a wireless terminal from one AP of a one manufacturer to another AP of a different manufacturer while maintaining the continuity of a communication link.

IEEE 802.11g WLAN Standards The IEEE 802.11g WLAN standard operates at 2.4 GHz ISM band with new modulation schemes, forward error correction and OFDM scheme to provide a wider array of data rates from 1 Mbps to 54 Mbps. IEEE 802.11g is backward compatible with 802.11 and 802.11b standards by specifying the same modulation and framing schemes for 1, 2, 5.5, and 11 Mbps data rates. IEEE 802.11g adopts the 802.11a OFDM scheme for 6, 9, 12, 18, 24, 36, 48, and 54 Mbps. The OFDM scheme is referred to as Extended Rate Physical (ERP) layer OFDM. In addition, ERP Packet Binary Convolutional Coding (ERP-PBCC) scheme is used to provide data rates of 22 and 33 Mbps optionally.

EXAMPLE 14.7 Physical-layer characteristics of IEEE 802.11 standards

Tabulate the physical-layer characteristics of IEEE 802.11 family of standards.

Solution Table 14.4 gives the physical-layer characteristics of the IEEE 802.11 family of standards.

Table 14.4 *Physical-layer characteristics of IEEE 802.11 standards*

Parameter	*IEEE 802.11*	*IEEE 802.11a*	*IEEE 802.11b*	*IEEE 802.11g*
Allocated frequency band	2.4–2.4835 GHz	5.15–5.35 GHz; 5.725–5.825 GHz	2.4–2.4835 GHz	2.4–2.4835 GHz
Available bandwidth	83.5 MHz	300 MHz	83.5 MHz	83.5 MHz
Number of channels	3	12	3	3
Baseband modulation	DSSS, FHSS with QPSK	OFDM with PSK and QAM	DSSS with QPSK and CCK	DSSS, OFDM
Channel data rate	1, 2 Mbps	6, 9, 12, 18, 24, 36, 48, 54 Mbps	1, 2, 5.5, 11 Mbps	1, 2, 5.5, 6, 9, 11, 12, 18, 24, 36, 48, 54 Mbps
Compatibility	IEEE 802.11	Wi-Fi5	Wi-Fi	Wi-Fi up to 11 Mbps

IEEE 802.11h WLAN Standards The IEEE 802.11h WLAN standards addresses the issues of spectrum allocation and power management. In the IEEE 802.11a standard, the 5-GHz band used in US and Europe for WLAN applications is slightly different. In order to comply with European regulatory requirements for spectrum and maximum allowable transmit power, the standard defines a dynamic channel selection and transmit power control mechanisms.

IEEE 802.11i WLAN Standards The IEEE 802.11i WLAN standards defines a new set of security mechanism, known as Wi-Fi Protected Access (WPA), which addresses three main security areas: the authentication, key management, and data transfer privacy at the MAC layer. It uses stronger encryption schemes such as AES with 128-bit keys or 104-bit RC4, instead of Wired Equivalent Privacy (WEP) mechanism provided in MAC layer of IEEE 802.11. To improve authentication, IEEE 802.11i requires the use of an authentication server which distributes the key and defines a more robust authentication protocol.

IEEE 802.11k WLAN Standards The IEEE 802.11k WLAN standards defines enhancements in radio resource measurement which enables AP to improve roaming decisions, to regulate access to a given channel, to query wireless terminals to collect statistics about transmission, reception and retries of packets in order to have a complete view of network performance. The standard also extends the transmit power control procedures defined in 802.11b to other frequency bands, to provide range control as well as to reduce power consumption and interference.

IEEE 802.11n WLAN Standards The IEEE 802.11n WLAN standards provides a series of enhancement techniques to both the physical layer and MAC layers leading throughput of up to 100 Mbps. The standards include Multiple-Input Multiple-Output (MIMO) and 40-MHz operation to the physical layer. The standard includes the use of multiple antennas, smart antennas, change of signal encoding schemes and MAC access protocols. It addresses other performance-related requirements such as more uniform radio coverage, improved range at existing throughputs, and increased resistance to interference.

EXAMPLE 14.8 IEEE 802.11 family

Summarise the scope of work and specification parameters of the IEEE 802.11 family of WLAN standards in a tabular form.

Solution

Table 14.5 summarises the IEEE 802.11 family of WLAN standards.

Table 14.5 *IEEE 802.11 family of WLAN standards*

S. No.	*Standard*	*Scope of work*	*Specifications/Parameters*
1.	IEEE 802.11	3 physical layers and 1 MAC layer	IR – 850-950 nm; ISM 2.4 GHz - DSSS/FHSS; 1, 2 Mbps
2.	IEEE 802.11a	Physical layer	5 GHz-OFDM; up to 54 Mbps
3.	IEEE 802.11b	Physical layer	ISM 2.4 GHz - DSSS; 5.5, 11 Mbps
4.	IEEE 802.11c	MAC layer	Support of IEEE 802.11 frame
5.	IEEE 802.11d	Physical layer	Extends operation
6.	IEEE 802.11e	MAC layer	QoS and security enhancement
7.	IEEE 802.11f	Inter access point protocol	Interoperability
8.	IEEE 802.11g	Physical layer	Extends 802.11b up to 54 Mbps
9.	IEEE 802.11h	Physical layer and MAC layer	Enhances 802.11a channel selection and power control
10.	IEEE 802.11i	MAC Layer	Security enhancements
11.	IEEE 802.11j	Physical layer	Extends 802.11a to Japan
12.	IEEE 802.11k	Higher layers	Enhances radio resource measurement
13.	IEEE 802.11m	-------	Enhances 802.11 technical and editorial corrections
14.	IEEE 802.11n	Physical layer and MAC layer	Enhancements for higher throughput (>100 Mbps)
15.	IEEE 802.11p	Vehicular wireless access	Looking at issues relating to using Wi-Fi radios in cars to access stationary wireless APs
16.	IEEE 802.11r	Fast BSS transition	Addressing hand-off delay due to authentication
17.	IEEE 802.11s	Mesh networking	Developing support for multi-hop wireless networking to improve coverage and reduce installation costs

14.1.5 WLAN Applications

Wireless LANs have clear-cut edge over traditional wired LANs because WLANs satisfy the additional requirements for mobility, ad-hoc networking, relocation, and coverage of locations which are difficult to wire. Due to advancements in transmission technologies, modulation and coding techniques, and many new protocols and algorithms, WLANs have been able to achieve acceptable data rates, reasonable security, and occupational safety concerns in an unlicensed worldwide frequency band.

One of the major application areas for WLAN lies in its capability either to provide similar services as an alternative to a wired LAN where laying of cables is not feasible, or as an extension to existing wired LAN to new operational areas. Thus, a wireless LAN offers an effective solution to set up either a peer-to-peer ad-hoc network with no centralised server or an infrastructure network to link with wired LAN and servers.

The WLAN must meet certain requirements such as connectivity among hundreds of wireless terminals across multiple BSSs as well as to fixed terminals on a wired backbone LAN, broadcast capability, hand-off/roaming capability, transmission robustness, data security, collocated WLANs operation, dynamic configuration, low power consumption, and so on.

Wireless LANs allow for the design of small, independent and battery-operated devices such as small PDAs and notepads. They can survive natural disasters, and wireless communication can still be established if the wireless devices are safe. After providing wireless access to the infrastructure via an access point for the first wireless terminal, addition of more wireless terminals in WLANs does not require any change in fixed infrastructure and hence are cost effective.

There are still few challenges and issues which need to be addressed for enhancement of application areas. For example, WLANs typically offer limited data rate with lower quality of service than their wired LANs. This is mainly because of lower bandwidth availability in radio transmissions. WLANs can offer only 1–10 Mbps data rate instead of 100–1,000 Mbps for wired LANs, higher error rates (10^{-4} instead of 10^{-12} for fiber optics) due to interference, eavesdropping, unwanted radiations, and longer delay due to usage of extensive error correction and detection mechanisms in WLANs. Moreover, WLANs are limited to operation in license-free frequency bands, which are not the same worldwide.

14.2 ETSI HIPERLAN TECHNOLOGY

High PErformance Radio LAN (HIPERLAN) is a pan-European alternative for the IEEE 802.11 WLAN standards, and is defined by the European Telecommunications Standards Institute (ETSI). In ETSI, the HIPERLAN standards are defined by the Broadband Radio Access Networks (BRAN) project. The HIPERLAN standards provide features and capabilities similar to those of the IEEE 802.11 WLAN standards, used in the US and other adopting countries.

The HIPERLAN standard family has four different versions: HiperLAN/1, HiperLAN/2, HIPERACCESS, and HIPERLINK. The high-speed HiperLAN/1 standard supports mobility at data rates above 20 Mbps in the 5-GHz RF band up to 100-m radio range. It specifies both infrastructure and multi-point to multi-point ad-hoc system architecture. HiperLAN/2 operates at up to 54 Mbps in the same RF band up to 25 Mbps data rates in an ad-hoc system architecture, and has the potential for sending and receiving data, images, and voice communications, and intends to accommodate ATM as well as IP-type access with QoS support. HIPERACCESS technology is used for remote access up to 5 km, up to 25 Mbps data rates, having provision for more than 155 Mbps with QoS. HIPERLINK is designed to interconnect different access points and switches with a high-speed link in the backbone of up to 155 Mbps data-rate capability.

The specific requirements for HIPERLAN can be listed as

– Multi-hop and ad hoc networking capability
– Radio coverage up to 100 metres
– Support of both asynchronous and synchronous traffic
– Data rates up to 23.5 Mbps in the 5.2 GHz unlicensed band
– Support of time-bounded services and power-saving mode

The maximum data rate for the user depends on the distance of the communicating nodes. With short distances up to 50 m and asynchronous transmission, a data rate of 20 Mbps is achieved; with up to 800-m distance, a data rate of 1 Mbps are provided. For connection-oriented services such as video-telephony, data rates of at least 64 kbps are offered.

14.2.1 HIPERLAN/1 System Architecture

HIPERLAN/1 is mainly designed to work without the need of any infrastructure. Two nodes may exchange data directly, without any interaction from a wired (or radio-based) infrastructure. Thus, the simplest HIPERLAN/1 consists of two nodes. Further, if two HIPERLAN/1 nodes are not in radio contact with each other, they may use a third node which must forward messages between the two communicating nodes. Figure 14.7 shows the overall system architecture of an ad-hoc HIPERLAN/1.

A multihub topology is considered that also allows overlay of two HIPERLANs to extend the communication beyond the radio range of a single node. There are two overlapping HIPERLANs, A and B, and the node 4 acts as a bridge between the two. Each node is designated either as a Forwarder (F) node or a Non-Forwarder (NF) node. In the above figure, nodes 1, 4, and 6 are forwarder nodes and these have forwarding connections. A forwarder node retransmits the received packet to other nodes in its neighbourhood, if the packet is not meant for it. Nodes 2, 3, and 5 are non-forwarder nodes, which simply accept the packet that is meant for it. Each non-forwarder node should select at least one of its neighbour nodes as a forwarder node. Inter-HIPERLAN forwarding needs mutual agrrement and cooperation, and should exchange regular update messages to support proper routing and maintenance.

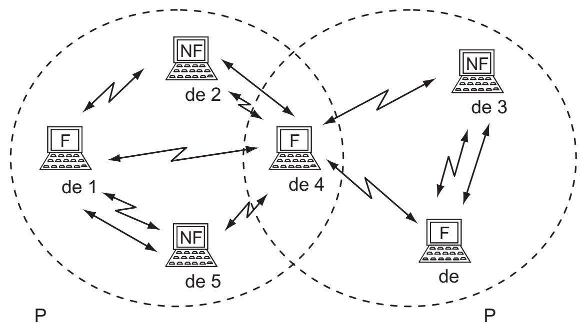

Fig. 14.7 *HIPERLAN/1 ad-hoc system architecture*

14.2.2 HIPERLAN/1 Protocol Specifications

The HIPERLAN/1 standard covers the physical layer and the Media Access Control (MAC) part of the data link layer like IEEE 802.11. The physical-layer specifications are summarised as follows:

Frequency band	5.15 GHz–5.35 GHz (Europe and US); 5.725 GHz–5.825 GHz (US only)
Channel spacing	40 MHz (Europe); 33 MHz (US)
Number of channels	5 channels (Europe); 6 channels in 5.15 GHz − 5.35 GHz (US) + 3 channels in 5.725 GHz − 5.825 GHz (US)
Multiple access technique	Non-Preemptive Multiple Access (NPMA)
Modulation technique	Single carrier GMSK
Data rate	Up to 23.5 Mbps
Equalisation technique	Decision Feedback Equaliser (DFE)
MAC address	6 bytes
Transmitted power	Up to 1 W (+30 dBm)

On the physical layer, a single carrier GMSK modulation is used in HiperLAN/1. The physical layer of the HIPERLAN/1 standard encodes each 416 data bits into 496 coded bits with a maximum of 47 code words per packet and 450 coded bits per packet for training the equaliser. The NPMA protocol is a listen-before-talk protocol, similar to CSMA/CA used in IEEE 802.11 WLAN standards, except that it is active rather than passive and contention resolution and ACK is mandatory. The NPMA protocol supports both asynchronous and isochronous (voice-oriented) transmissions.

The HIPERLAN/1 MAC layer defines a priority scheme and a lifetime for each packet to facilitate QoS control, in addition to the routing, encryption and power saving. There is a new sublayer called Channel Access and Control (CAC) sublayer. This sublayer deals with the access requests to the channels. The accomplishing of the request is dependent on the usage of the channel and the priority of the request. The CAC layer provides hierarchical independence with Elimination-Yield Non-Preemptive Multiple Access mechanism (EY-NPMA). EY-NPMA encodes priority choices and other functions into one variable length radio pulse preceding the packet data. EY-NPMA enables the network to function with few collisions even though there would be a large number of users. Multimedia applications work in HIPERLAN/1 because of EY-NPMA priority mechanism. The MAC layer defines protocols for routing, security and power saving, and provides naturally data transfer to the upper layers. The MAC address uses six bytes to support IEEE 802.2 LLC layer. Each packet has six address fields that identify source, destination, and the immediate neighbour for multihop implementations.

If a node senses the medium to be free for at least 1700 bit durations, it immediately transmits. If the channel is busy, the node has three phases when the channel becomes available for data transmission. These three phases are the following:

Prioritisation Phase During this phase, the highest priority node among the competing nodes having five available priority levels will transmit the data. At the end of the prioritisation phase, all the nodes listen to the asserted highest priority. The combing algorithm is more structured and active which provides for a more robust process.

Contention Phase It has two periods, elimination and yield. During the elimination period, each node runs a random number generator to select one of 12 available slots in which it sends a continuous burst of 256 bits. Thereafter, the node listens to the channel for 256-bit duration. If necessary, it repeats the burst transmission. The contention process is more complicated and has active as well as passive parts. In the yield phase, the remaining nodes have an exponentially distributed random yield period to listen to the channel.

Transmission Phase If a node survives for the entire yield period, it will start transmitting the data that automatically eliminates other nodes that are listening to he channel.

14.2.3 HIPERLAN/2 Standards

HIPERLAN/2 is designed as a fast wireless connection for many kinds of networks such as the UMTS backbone network, ATM and IP networks. HIPERLAN/2 allows interconnection into almost any type of fixed network technology. This makes it suitable, for example, to connect mobiles, portables and laptops to a fixed access point. The basic services provided in HIPERLAN/2 are QoS in voice, data, and video transmission. HIPERLAN/2 also provides unicast, multicast and broadcast transmissions.

It uses the unlicensed 5 GHz U-NII band and supports up to 54 Mbps data rate at the physical layer and about 35 Mbps at the network layer. Modulation techniques such as BPSK, QPSK, 16QAM or 64QAM are used. The physical layer of HIPERLAN/2 is very similar to IEEE 802.11a wireless LANs. However, the multiple access technique is dynamic TDMA in contrast to CSMA/CA used in IEEE 802.11a. The HIPERLAN/2 standard covers physical, data link control and convergence layers. Convergence layer takes care of service-dependent functionality between the data link control and network layer. Convergence sublayers can also be used on the physical layer to connect the IP, ATM or UMTS networks. This feature makes HIPERLAN/2 suitable for the wireless connection of various networks. Good security measures are

Facts to Know!

Two unregulated bands are the Industrial, Scientific and Medical (ISM) band and the Unlicensed National Information Infrastructure (U-NII). The total bandwidth allocated to ISM and U-NII bands are 234.5 MHz in 2.4-GHz band and 300 MHz in 5-GHz band respectively.

offered by HiperLAN/2. The data is secured with DES or Triple DES algorithms. The wireless access point and the wireless nodes can authenticate each other. The most popular worldwide HIPERLAN/2 manufacturers are Motorola (USA), Alvarion (Israel), and SICE (Italy).

14.3 IEEE 802.15 WPAN TECHNOLOGY

The IEEE 802.15 standards is a family of protocols to address the needs of Wireless Personal Area Network (WPAN) at different data rates in 2.4 GHz ISM band, same as defined in IEEE 802.11 WLAN standards. A WPAN is a wireless communications network among a number of portable and mobile devices on the network such as cellphones, pagers, PCs, laptops, peripherals, PDAs, and consumer electronic devices within a small service area of up to about 10 metres, which enables the use of low power, low cost, and extremely small-sized devices.

> **Facts to Know!**
>
> The IEEE project P802.15 covers all of the different working Groups for WPANs. The last digit in the standard identifies specific working groups such as ".1" for Bluetooth and ".3" for high rate WPANs.

14.3.1 IEEE 802.15.1 System Architecture

The IEEE 802.15.1 WPAN standards support applications which require medium data rate (typically up to 1 Mbps). Bluetooth technology has been adopted as the IEEE 802.15.1 WPAN standards which are commercially available in numerous devices ranging from cellphones, PDAs, laptops to wireless mouses and cameras.

Bluetooth devices can communicate with other Bluetooth devices in several ways. The basic unit of the WPAN system architecture is a piconet. It consists of a master device and from a minimum of one to a maximum of seven slave devices. All slave devices must be within 10-metre radio range of the master device. The master device determines the assignment of frequency-hopping sequence and timing offset for slave devices to transmit. A slave device may only communicate with the master device after getting permission to do so. There can be 255 parked slave devices in the single piconet but at any time, a maximum of seven are communicating.

> **Facts to Know!**
>
> In a piconet, the master and slave devices transmit alternatively. The master device starts its transmission in even-numbered time slots only, and the slave device starts its transmission in odd-numbered time-slots only.

Figure 14.8 depicts a simple configuration of a piconet architecture.

Each of the slave devices has an assigned 3-bit active device address. Many other inactive slave devices can remain synchronised to the master device, referred to as parked devices. The master device regulates the channel access and other operations for all active devices as well as parked devices. Piconet supports both point-to-point and point-to-multipoint connections.

A device in one piconet may also exist as part of another piconet and may function as either a master or slave in each piconet. This form of system architecture configured from overlapping piconets

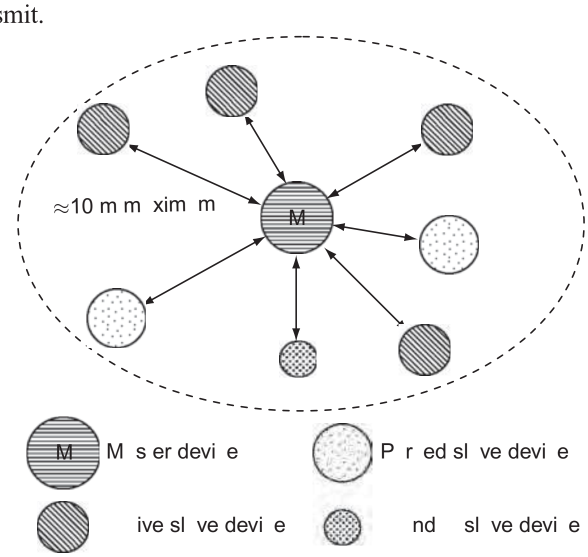

Fig. 14.8 *A simple piconet architecture*

is called a scatternet. Thus, a scatternet can be referred to a group of independent and non-synchronised piconets that share at least one common device. Current implementations of piconet or scatternet depend primarily on simple point-to-point data links between devices within direct radio range of each other. However, the IEEE 802.15 specification defines not only a point-to-point wireless link but also more complex networking topologies. Two piconets can be connected through a common bridge or gateway to form a scatternet. Figure 14.9 depicts a simple configuration of scatternet architecture.

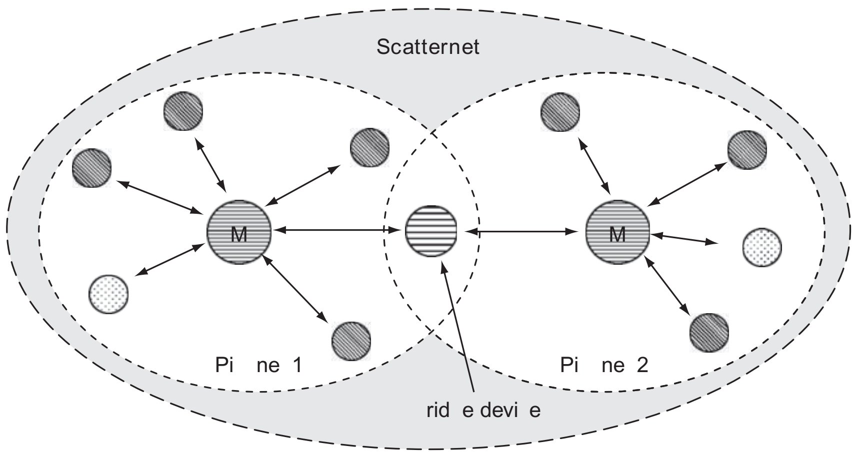

Fig. 14.9 *A simple scatternet architecture*

The objective to form scatternets is to provide effective and efficient communication over multiple hops with acceptable response times and power consumption so that end-to-end wireless communication links can be established. The devices must have point-to-multipoint capability to engage in scatternet communication. These interconnected piconets within the scatternet form a backbone for the Mobile Ad hoc Network (MANET), and can enable devices which are not directly communicating with each other, or which are out of radio coverage area of another device, or to exchange data through several hops in the scatternet. There may be a maximum of 10 fully loaded piconets in a scatternet. Adding more piconets leads to a graceful degradation in the performance of a single piconet because more and more collisions may occur. A collision occurs if two or more piconets use the same carrier frequency at the same time. This could probably happen as the hopping sequences are not centrally coordinated.

A master or slave device in one piconet can function as a slave in another piconet by being paged by the master of this other piconet. This automatically means that any device can create a new piconet by paging a device that is already a member of a piconet. Any device participating in one piconet can page the master or slave device in another piconet. This could lead to a switch of roles between the master or slave device in this scenario. Inter-piconet communications are established over the shared device. Time multiplexing must be used for that device to switch between piconets.

Facts to Know!

The Bluetooth hopping sequence is considerably faster than that of most residential cordless telephones operating in the same 2.4-GHz ISM band and that usually hop frequencies about 100 times per second. The interference caused by cordless phones to a Bluetooth transmission can result in data errors or break-ups in a voice stream.

The WPAN uses about 79 different channel frequencies in a total spectrum bandwidth of 80 MHz, each channel having a bandwidth of 1 MHz. The piconet/scatternet architecture allows many devices to share the same service area and make efficient use of the available bandwidth. A frequency-hopping scheme is used with a carrier spacing of 1 MHz. With frequency hopping, a logical channel is defined by the frequency-hopping sequence. At any given time, the bandwidth available is 1 MHz only, with a maximum of eight devices (one master and seven slave) sharing the bandwidth. However, different hopping sequences or logical channels can simultaneously share the same 80-MHz bandwidth. A piconet shares the logical channel and data transfer, whereas a scatternet shares the service area and available bandwidth. In other words, all piconets can share the total of 80-MHz bandwidth available. The system uses FH-CDMA technique for separation of piconets.

When devices in different piconets, operating on different frequency sequences, happen to use the same hop frequency at the same time, collisions will occur which will degrade the performance in terms of throughput. If the size of the scatternet is quite large, that is, as the number of piconets in a scatternet increases to cover a large area, the performance will degrade significantly due to increase in collisions.

14.3.2 IEEE 802.15.1 Protocol Architecture

The IEEE 802.15 WPAN standard protocol architecture defines the physical layer specifications and MAC layer requirements. The IEEE 802.15.1 standards, also popularly known as Bluetooth specifications, defines the physical layer comprising of a Frequency-Hopping Spread Spectrum (FHSS) device that uses the worldwide unlicensed 2.4-GHz ISM frequency band. Table 14.6 summarises the radio and baseband parameters of Bluetooth devices.

Table 14.6 *IEEE 802.15.1 Bluetooth physical-layer specifications*

S. No.	*Parameter*	*Specification*
1.	RF spectrum allocation	2.4 GHz (unlicensed ISM band)
2.	Number of RF channels	79
3.	Channel bandwidth	1 MHz
4.	Multiple access scheme	FHSS-TDD-TDMA in piconet; FHSS-CDMA in scatternet
5.	Frequency hop rate	1600 hops per second (hop slot of 625 μs)
6.	Modulation scheme	GFSK
7.	Transmit Power	
	Class I:	1 mW − 100 mW with power control
	Class II:	0.25 mW − 2.4 mW with optional power control
	Class III:	1 mW nominal
8.	Symbol transmission rate	1 Msps
9.	Piconet topology	Up to 7 simultaneous links in a logical star
10.	Scatternet topology	Up to 10 piconets per coverage area

In most countries including US, there are 79 different frequency channels available in the available 2.4 GHz–2.4835 GHz ISM band. The nominal bandwidth for each channel is 1 MHz. A Bluetooth transceiver uses all 79 channels, and hops (changes frequencies) pseudo randomly across all channels at a rate of 1600 hops per second for data transmissions. When connected to other Bluetooth devices, a Bluetooth device hops in a pseudorandom sequence, with each physical

Facts to Know!

There are three types of error-correction techniques used in the Bluetooth protocols: 1/3 rate Forward Error Correction (FEC), 2/3 rate FEC and the Automatic Re-transmission Request (ARQ).

channel occupied for 625 μs. Each 625 μs time period is referred to one time slot, and these are numbered sequentially. When in inquiry or page mode, it hops at 3200 hops per second with a hold time of 312.5 μs. If the hop frequency encounters the deep fade, the packet is lost. It is retransmitted in the next hop at a randomly selected frequency. Frequency hopping in Bluetooth provides resistance to interference and multipath effects. It is a simple system with low power consumption and reliable transmission. It also provides a form of multiple access among co-located devices in different piconets.

The Bluetooth specification uses Time Division Duplexing (TDD) and Time Division Multiple Access (TDMA) for device communication. TDD is a link transmission technique in which data are transmitted in one direction at a time, with transmission alternating between uplink and downlink directions. Since more than two devices share the piconet medium, the access technique can be characterised as FH-TDD-TDMA. The FH sequence is determined by the master in a piconet and is a function of the master device's address. Different piconets use different hop sequences because they will have different master devices in the same area. With the use of FEC and error detection/ARQ techniques between devices on different piconets in the same scatternet, FH-CDMA technique is used.

A single time slot is 625 μs in length, representing the length of a single-slot packet. At the baseband layer, a packet consists of an access code, a header, and the payload. The access code contains the piconet address (to filter out messages from other piconets) and is usually 72 bits in length. The header contains link control data, encoded with a Forward Error-correcting Code (FEC) with a 1/3 rate for high reliability. Such a code is a repetition code and thus every bit in the header is transmitted three times. The header is usually 18 bits in length, and includes the active member address for a currently active slave.

Bluetooth uses GFSK modulation scheme in which a binary 0 represents a negative frequency deviation and a binary 1 represents a positive frequency deviation from the centre frequency. The minimum frequency deviation is 115 kHz. Power control is used to keep the Bluetooth devices from radiating any more RF power than necessary. The power-control algorithm is implemented using the link-management protocol between a master device and the slave devices in a piconet.

It has a range of approximately 10 metres, although ranges of up to 100 metres can be achieved with the addition of RF power amplifiers. Because the transceiver has an extremely small footprint, it is easily embedded into physical devices, making it a truly ubiquitous radio link.

All communication between devices takes place between a master and a slave, using time-division duplex, with no direct slave-to-slave communication. The master will poll each active slave to determine if it has data to transmit. The slave may only transmit data when it has been polled. Also, it must send its data in the time slot immediately following the one in which it was polled. The master transmits only in even-numbered time slots, while the slaves transmit only in odd-numbered time slots. In each time slot, a different frequency channel is used (a hop in the hopping sequence).

A piconet is established with a potential master device identifying other devices in its radio coverage area that wish to participate in the piconet. It begins with an INQUIRY message by the potential master device if the address is unknown, followed by a subsequent PAGE message. If the address is already known, a connection is made by sending a PAGE message only. Once the master has located all other devices within its range, it is able to establish a connection to each device, setting up a piconet.

A power-saving mode can be used for devices in a piconet if there are no data to be transmitted. The master device can put the active slave devices in HOLD mode, where only an internal timer is running. In the PARK mode, a device is still synchronised to the piconet but does not participate in the communication. In the SNIFF mode, a slave device just listens to the piconet at a programmable reduced rate and duty cycle.

Communication between different piconets takes place by devices jumping back and forth between these nets. If this is done periodically, for instance, isochronous data streams can be forwarded from one piconet to another. A master device can also leave its piconet and act as a slave device in another piconet. It is clearly not possible for a master device of one piconet to act as the master device of another piconet as this would lead to identical hopping sequence. As soon as a master device leaves a piconet, all traffic within this piconet is suspended until the master device returns. However, scatternets are not yet supported by all devices.

14.3.3 IEEE 802.15.3 High-Rate WPAN

The IEEE 802.15.3 standard, also called WiMedia, physical layer operates in the unlicensed 2.402–2.480 GHz frequency band. It is designed to achieve higher data rates of the order of 11–55 Mbps, which are needed in high-fidelity audio and high-definition video applications. WiMedia Ultra Wideband (UWB) technology is optimised for Wireless Personal-Area Networks (WPANs), delivering high-speed, low-power multimedia capabilities for the PC, mobile and automobile market segments. This technology goes one step further related to spread spectrum used in WLANs as it transmits digital data over a wide spectrum of frequency bands with very low power. Typically, the occupied spectrum is quite wide, say, at least 25 per cent of the centre frequency that is, 500 MHz for a 2-GHz system.

The IEEE 802.15.3 standard defines the specifications for high-rate WPANs supporting speeds of 11 Mbps, 22 Mbps, 33 Mbps, 44 Mbps, and up to 55 Mbps in the 2.4 GHz ISM band. It supports peer-to-peer or ad hoc networks. HR-WPANs piconets support child and neighbour piconets. It supports two different channel plans—a coexistence channel plan, and a high-density channel plan. Channels are limited to 15-MHz bandwidth. It uses Trellis Code Modulation (TCM) and Forward Error Correction (FEC) technique which encodes the digital signal so that single bit errors can be detected and corrected.

UWB transmits low-power, short-range signals. UWB broadcasts a very short digital pulse (less than 1 ns) that is timed very precisely. The sender and receiver must be synchronised with very high accuracy. If the sender knows exactly when a pulse should arrive, multi-path propagation is no longer an issue because only the strongest signal will be detected within a very short time slot. UWB technology can be used for WLANs transmitting very high data rates over short distances.

Facts to Know!

UWB does not use a conventional radio signal carrier transmitting signals in the regulated frequency spectrum. Instead, UWB uses low-power, precisely timed pulses of energy that operate in the same frequency pattern as low-end noise such as that emitted by TV monitors and computer chips.

UWB can send data at speeds of up to 2 Gbps. Digital signals need to be spread over a wide band using techniques such as FHSS or DSSS. UWB uses short analog pulses for signaling, called impulse modulation. Orthogonal Frequency Division Multiplexing (OFDM), commonly referred to as MB-OFDM, in which a frequency band is divided into five groups containing a total of 14 frequency bands is also used in the physical layer of UWB.

The IEEE 802.15.3 system employs a 11 Msps symbol rate, and uses Trellis-coded QPSK at 11 Mbps, uncoded QPSK modulation at 22 Mbps, and 16/32/64-QAM at 33, 44, 55 Mbps respectively. The RF and baseband processors used in the physical layer are optimised for short-range transmission limited to 10 metres. The physical layer also requires low current drain (less than 80 mA) while actively transmitting or receiving data in the power-saving mode. The MAC layer specifies a superframe structure to provide provisions for supporting multimedia QoS, ad-hoc PAN topology similar to Bluetooth with master and slave devices, and power management. This standard is not compatible with either Bluetooth or the IEEE 802.11 WLAN family of protocols.

Facts to Know!

Only some 802.15.3 devices must be capable of being a piconet coordinator (PNC). Portable devices that are battery powered and need to be switched off periodically should not assume the role of PNC.

IEEE 802.15.3a is the proposed enhancement to 802.15.3. It uses UWB technology to support higher data rates for multimedia and imaging applications. It is based on the WiMedia specifications, transmits at 480 Mbps at a distance of up to 2 metres. Security for IEEE 802.15.3 HR WPANs is based on the Advanced Encryption Standard (AES) which defines how any two devices can establish a secure communications session. It also supports message integrity verification at the MAC layer. 802.15.5 mesh networking standard extends the capabilities of 802.15.3 networks.

14.3.4 IEEE 802.15.4 Low-Rate WPAN

The IEEE 802.15.4 standards define low-rate, low-power WPANs with low bandwidth requirements. The physical layer specifies the operation at 2.4-GHz ISM band worldwide to offer a transmission data rate of 250 kbps. It supports 16 channels between 2.4 GHz and 2.4835 GHz with a 5-MHz channel spacing. It employs a 16-ary quasi-orthogonal modulation technique based on DSSS binary data which offers better performance than differential BPSK. The applications have few or no QoS requirements with provision for retransmission of data in case of errors due to interference. Moreover, transmissions will be infrequent, with the devices operating in a passive mode most of the time. Higher throughput, lower duty cycle, and lower latencies are the features of these specifications.

Facts to Know!

The IEEE 802.15.4 standard defines star and peer-to-peer topologies only, since a cluster tree network essentially consists of multiple-star topology networks.

As an alternative to already congested ISM band, it defines the operation at 868 MHz band for Europe to offer a data rate of 20 kbps, and at 915 MHz band for US to offer a data rate of 40 kbps. At such low data rates, larger coverage area and either receiver sensitivity (–95 dBm) can be achieved. The 868 MHz or 915 MHz physical channel supports a single channel between 868.0 and 868.6 MHz and 10 channels between 902.0 and 928.0 MHz. It uses a simple DSSS in which each bit is represented by a 15-chip m-sequence. Each device should be able to transmit at least 1 mW in order to ensure a nominal range of 10–20 metres.

The MAC layer is common to both types of physical layers. Channel equalisation is not needed for either physical layer because of small coverage area and the relatively low chip rates. Typical packet sizes for monitoring and control purpose is of the order of 30–60 bytes only.

Table 14.7 gives a comparison of major parameters of different WPAN technologies.

Table 14.7 *Comparison of IEEE 802.15 WPAN standards*

Parameter	*IEEE 802.15.1*	*IEEE 802.15.3*	*IEEE 802.15.4*
Technology	Bluetooth	High-rate WPAN	Low-Rate WPAN
RF spectrum	2.4 GHz ISM band	2.402–2.480 GHz ISM band	2.4 GHz ISM band; 868 MHz band; 915 MHz band
Channel access scheme	TDD with master-slave polling	CSMA/CA with guaranteed time slot in a superframe structure	CSMA/CA with guaranteed time slot in a superframe structure
Modulation scheme	FHSS @ 1600 hops/s	Trellis-coded or uncoded QPSK or 16/32/64-QAM	DSSS with BPSK or MSK
Maximum data rate	Up to 1 Mbps	11 Mbps−55 Mbps	250 kbps at 2.4 GHz; 20 kbps at 868 MHz; 40 kbps at 915 MHz
Power consumption	1 mA − 60 mA	< 80 mA	20−50 μA
Coverage range	Up to 10 meters	Up to 10 meters	Up to 20 meters

14.3.5 WPAN Applications

Each microchip in a Bluetooth device is built with necessary identification coding and software controls. This ensures that the device can operate only and communicate with those units having custom features such as CVSD modulation with voice coding to withstand high bit-error rates, adaptive output power and fast frequency hopping to minimise interference, short data packets to maximise capacity, flexible packet types to support a wide range of applications, synchronous and asynchronous data services, easy integration of TCP/IP for networking fast acknowledgments allowing low coding overhead, transmission/reception interface designed to minimise power consumption, non line-of-sight transmission through walls, omnidirectional coverage, and built-in security. Various applications of WPAN technology are given below.

Facts to Know!

WPAN devices are designed to be small and consume very little power, thereby have limited storage and processing capabilities. This makes it difficult to implement sophisticated security mechanisms.

(a) A Wireless Personal Area Network (WPAN), popularly known as Bluetooth technology, is an extremely short-range wireless network, formed around the personal operating space of a wireless terminal with built-in Bluetooth device.
(b) Typically, WPANs are used to replace cables between a computer and its peripheral devices.
(c) WPANs can be used for transmitting images, digitised music, and other data.
(d) Bluetooth is the only WPAN technology which is commercially available and is an essential component in a series of devices ranging from laptops to wireless mouses to cameras and cellphones.
(e) Bluetooth technology enables short-range wireless communication networks between user devices incorporating a Bluetooth interface, and greatly improves the way users access services and data wirelessly.
(f) With the help of Bluetooth technology, ad-hoc wireless piconets can be formed, which are local area networks with a very limited coverage (about 10 metres) and without the need for an infrastructure, offering asynchronous data and synchronous voice services at data rate of 1 Mbps.
(g) Mobile phones could have a built-in Bluetooth chip in place of WLAN adapters. The mobile phone can then act as a bridge between the local piconet and the cellular network. In this way a mobile phone can be connected to a PDA or laptop in a simple way using wireless piconets.
(h) Bluetooth standard utilises a short-range radio link to exchange information, enabling effortless wireless connectivity between mobile phones, mobile laptops, handheld computers and other peripherals.
(i) Bluetooth technology aims to replace the IrDA specifications of Infrared in mobile phones and computing devices.
(j) Bluetooth can connect cellular mobile phones, a laptop, notebooks, desktop PCs, PDAs, handsfree headsets, LCD projectors, printers, modems, wireless LAN devices, FAX machines, keyboards, joysticks, and virtually any other digital device to one another via Bluetooth short-range radio modules installed in each of these devices, replacing the cable used to connect them.
(k) Bluetooth also provides a universal bridge to existing data networks and a mechanism to form small private Mobile Ad-hoc Networks (MANETs).
(l) WPANs help in the interworking of wireless technologies to create heterogeneous wireless networks. For instance, WPANs and WLANs will enable an extension of devices without direct cellular access to 3G cellular systems.
(m) Devices interconnected in a WPAN should be able to utilise a combination of both 3G access and WLAN by selecting the access mechanism that is best suited at a given time. In such networks, 3G, WLAN, and WPAN technologies do not compete against each other but enable the user to select the best connectivity for intended purposes.

(n) The Bluetooth specification defines two transmit power levels—a low transmit-power level that covers a small personal area within a room, and a high transmit-power level that can cover a medium range, such as an area within a home or office.

14.4 IEEE 802.16 WMAN TECHNOLOGY

The IEEE 802.16 standards address the needs of Wireless Metropolitan Area Network (WMAN) that can provide data communication network in an entire city. The access to the data network is provided by the WMAN to buildings through exterior antennas, communicating with base stations. It can also extend the capabilities of existing cabled access networks such as coaxial systems using cable modems, DSL links, and fiber optic links. The IEEE 802.16 standards have been evolved as a set of air interfaces at 10 GHz–66 GHz band on a common MAC protocol layer. The IEEE 802.16a, also known as WiMAX, extends the air interface support to lower frequencies in the licensed as well as unlicensed 2 GHz–11 GHz band. The WiMAX system can serve more wireless terminal users at relatively lower data rates.

14.4.1 Need of Wireless MAN

In general, wireless networks allow users to be connected as they move about, freeing them from cumbersome wires/cables and phone lines. WLANs and WPANs have restricted both connections and mobility, allowing users to roam around a few hundreds of metres from the source of the RF signal. These users have also been restricted mostly to stay within line-of-sight antennas. Therefore, user mobility has remained largely confined to offices, homes, and hotspots (such as airports and some public places in larger cities) except for voice communications and low-speed data over cellular networks.

To prevent interference in unlicensed bands, user mobility has also been restricted in Wi-Fi networks due to use of low transmitter power. In remote areas as well as areas with low user density, it may not make it economically viable to implement mobile access and hotspots. Moreover, the high cost of installing wired high-speed communication channels over long distances prevent high-speed Internet access at all.

WMANs are a group of technologies that provide wireless connectivity across a substantial geographical area such as a large metropolitan city. The network provides the access of wired networks beyond a single location without the expense of high-speed cable-based connections as well as extends user mobility throughout a metropolitan area. WMAN can provide high-speed connections, including Internet, to areas not serviced by any other method of connectivity.

14.4.2 IEEE 802.16 Protocol Architecture

The IEEE 802.16 standard specifies the physical layer in the 10 GHz–66 GHz band. The channel bandwidths are typically 20 MHz and 25 MHz in US, and 28 MHz in Europe. The point-to-point wireless communication is enabled through a TDM scheme in which a base station transmits the data sequentially to each wireless terminal in its allocated time slot in the downlink direction. The channel access in the uplink direction is by TDMA technique. For the deployment of single-carrier transmission the line-of-sight conditions must exist. The burst design selected allows coexistence of both FDD (full-duplex as well as half-duplex) and TDD forms of communication.

Both FDD and TDD support adaptive burst data transmissions. By using this adaptive burst scheme, each link may adjust the transmission parameters such as modulation and coding schemes individually by frame-by-frame basis. The data bits are randomised to minimise the possibility of transmission of an unmodulated carrier signal and to ensure adequate numbers of bit transitions to support clock recovery. The data is FEC coded with RS coding technique which allows variable block size and has appropriate error-correction capabilities. Frame control and initial access data is additionally coded with robust block convolutional coding technique. The FEC encoded data is mapped to a QPSK, 16-QAM or 64-QAM to form burst transmissions with varying efficiency and robustness.

A frame of typical 0.5, 1, or 2 ms size is divided into time slots for bandwidth allocation and identification of physical-layer transmissions. The frame size in uplink and downlink subframes has to be kept same. A time slot has 4 QAM symbols. Different framing schemes are defined for the FDD and TDD variant. Because the bandwidth requirements may vary from time to time, the mixture and duration of burst data and the presence or absence of the TDMA portion may also vary from frame to frame.

EXAMPLE 14.9 IEEE 802.16 WMAN physical-layer specifications

Summarise IEEE 802.16 WMAN Physical Layer specifications.

Solution

1. Frequency band of operation	10 GHz–66 GHz
2. Channel bandwidth	20/25/28 MHz
3. Multiple access technique	*Uplink* TDMA; *Downlink* TDD and FDD
4. Modulation scheme	QPSK, 16-QAM, 64-QAM
5. Nominal data rate	120 Mbps for 25 MHz; 134.4 Mbps for 28 MHz
6. Security features	Defines an additional privacy sublayer for authentication

The IEEE 802.16 MAC layer protocol supports point-to-multipoint broadband wireless access. It accommodates both continuous and bursty type of data traffic. It allows very high data rate transmissions in both the uplink and downlink directions. The system is capable of supporting hundreds of terminals per channels at the same time that may further be potentially shared by multiple end-users. The types of services required by wireless terminals include TDM voice and data, IP connectivity, and packetised voice over IP (VoIP). The MAC layer specifications provide a wide range of service types with assigned QoS and guaranteed frame rate requirements.

The MAC layer makes use of bandwidth-efficient burst data transmissions under favourable link conditions. Modulation and coding schemes are specified in a burst data format that may be adjusted to each wireless terminal adaptively. The MAC layer also includes a privacy sublayer which provides authentication of network access as well as data encryption. The IEEE 802.16 MAC layer protocol is connection-oriented, with a provision to map inherent connectionless services. This enables to provide a mechanism for requesting desired bandwidth, associating QoS and traffic parameters, and routing data statistics. The MAC reserves some connections such as contention-based initial access, broadcast transmissions in the downlink, signaling broadcast polling of user bandwidth demands, among others.

Facts to Know!

Each wireless terminal has a standard 48-bit MAC address but connections are referenced with 16-bit connection identifiers and may require bandwidth on demand or continuous availability of bandwidth.

The MAC protocol supports an automatic initialising procedure. When a wireless terminal is switched on, it begins scanning its frequency list to find an active channel. It may be programmed to register with one specific base station, each base station having its unique programmable ID. This particular feature is useful when a large number of base stations are deployed close to one another. In such a situation, the wireless terminal might receive unwanted signals from other base stations due to frequency selective fading problem or from side lobes of a nearby base station antenna.

14.4.3 IEEE 802.16a WiMAX

WiMAX is an acronym that stands for Worldwide Interoperability for Microwave Access. WiMAX, also known as Wireless Metropolitan Area Networks (WMANs), provides broadband wireless connectivity across a substantial geographical area such as a large metropolitan city. It is based on the IEEE 802.16a standard and has been designed to evolve as a set of air interfaces based on a common MAC protocol but with physical layer specifications having an air interface support in the 2–11 GHz band, including both licensed and license-exempt spectra.

WiMAX can use radio channel bandwidths that can vary from 1.25 MHz to 28 MHz in steps of 1.75 MHz in 2 GHz–11 GHz band, and uses multicarrier OFDMA scheme to achieve transmission data rates as high as 155 Mbps. WiMAX can provide multiple types of services to the same user with different Quality of Service (QoS) levels. For example, it is possible to install a single WiMAX transceiver in an office building and provide real-time telephone services and best effort Internet browsing services on the same WiMAX connection. This is made possible because WiMAX has been designed to mix contention-based (competitive access) and contention-free (polled access) to provide services which have different QoS levels. The mesh configuration can be deployed to link different base stations without the need to install or lease interconnecting communication lines. WiMAX technology can provide services such as provisions of leased line, residential broadband, commercial broadband and digital television (IPTV) services.

Figure 14.10 demonstrates some of the key applications areas provided by WiMAX systems which include wireless broadband Internet access, telephone access services, television service access and

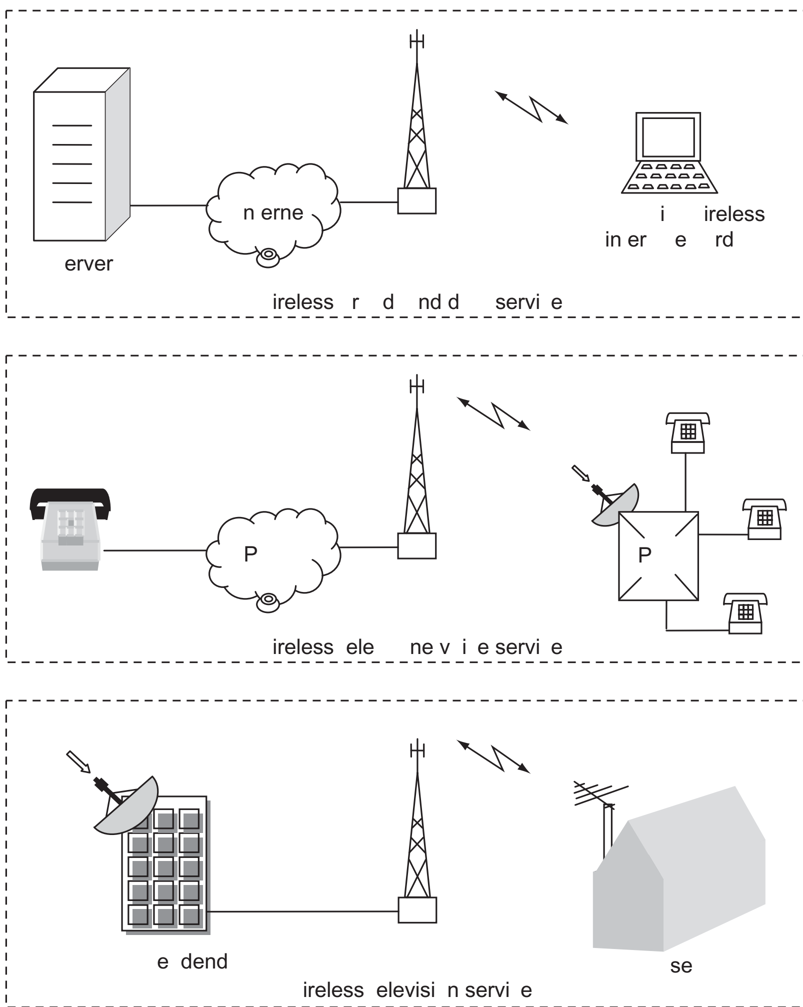

Fig. 14.10 *WiMAX applications*

mobile telephone services. Table 14.8 presents a comparison between the key parameters of original fixed WiMAX standard and the Mobile WiMAX standard that can be used for fixed, portable, and mobile wireless terminals.

Table 14.8 *IEEE 802.16a WiMAX parameters*

Parameter	*ixed WiMA*	*Mobile WiMA*
Frequency band	2.5 GHz, 3.4–3.6 GHz, 5.8 GHz	2.3–2.4 GHz, 2.5–2.7 GHz, 3.3–3.4 GHz, 3.4–3.8 GHz,
Duplexing technique	TDD, FDD	TDD, FDD (optional)
Modulation	OFDM	OFDMA
Wireless terminal types	Fixed	Fixed, portable and mobile
Hand-off	No	Yes

Facts to Know!

WiMAX is a more robust standard for high-speed broadband wireless delivery to laptops and desktops have been launched. It is a standards-based technology enabling the delivery of last km wireless broadband access as an alternative to cable and DSL. WiMAX provides fixed, nomadic, portable and, eventually, mobile wireless broadband connectivity without the need for direct line-of-sight with a base station.

IEEE 802.16a supports enhancements and extensions to the MAC protocols. The Base Station (BS) can communicate with another BS and also directly with Subscriber Stations (SS). The IEEE 802.16a air interface standard is truly a state-of-the-art specification for fixed broadband wireless access systems employing a point-to-multipoint architecture. Compared to the higher frequencies, such spectra offer a less expensive opportunity to reach many more customers, although at generally lower data rates. Although IEEE Standard 802.16a specifies three alternative physical layers, it is the W-OFDM based physical layer that is intended for mainstream applications. The physical layer supports multiple frequency bands and several modulation techniques. The WiMAX MAC layer is connection oriented and includes service-specific convergence sublayers that interface to the upper OSI layers. WiMAX offers multiple simultaneous services through the same link and compatible with Asynchronous Transfer Mode (ATM), IPv4, IPv6, Ethernet, and VLAN.

W-OFDM is a variation of OFDM that further improves its characteristics. The signal reception is corrected for distortions, allowing greater transmission speeds, and the signal is processed to maximise the range and multipath resistance. W-OFDM allows lower-power, multipoint Radio Frequency (RF) networks to be implemented, minimising interference with adjacent networks. W-OFDM effectively permits several independent channels to operate within the same frequency band, allowing multipoint networks and point-to-point backbone systems to be overlaid on one another. Less disruption of adjacent users and insensitivity to external noise means that high-speed multipoint data networks can be simply and rapidly deployed. These systems are tolerant of changes in the RF environment, limiting maintenance requirements, and the systems can be easily and economically expanded to meet the service provider's growing business.

The maximum distances achievable in a WiMAX network depend on the frequency band used. Higher frequencies are used for metropolitan area line-of-sight, point-to-point, or multipoint applications at very high data rates. Lower licensed frequencies will be used for private, line-of-sight network connections of up to 16 kilometers and long-distance links of up to 56 kilometres. Frequencies below 11 GHz will be used for non-line-of-sight networks with a maximum range of up to 8 kilometres. In a typical cell radius deployment of 3-10 kilometres, WiMAX systems can be expected to deliver capacity of up to 40 Mbps per channel, for fixed and portable access applications. This is enough bandwidth to simultaneously support hundreds of businesses with high-speed connectivity and thousands of residences with DSL speed connectivity. Mobile network deployments are expected to provide up to 15 Mbps of capacity within a typical cell radius deployment of up to three kilometres.

WiMAX equipment can operate in different FDD or TDD configurations, have different channel bandwidths or operate in different spectrum bands of 5.8 GHz, 3.5 GHz and 2.5 GHz. The WiMAX will facilitate the deployment of broadband wireless networks based on the IEEE 802.16 standard by helping to ensure the compatibility and inter-operability of broadband wireless access equipment. Under the current conditions, 802.16a could emulate 802.11a and is considered the next step beyond Wi-Fi because it is optimised for broadband operation, fixed and later mobile, in the wide area network. WiMAX technology already includes numerous advances that are slated for introduction into the Wi-Fi standard such as quality of service, enhanced security, higher data rates, and mesh and smart antenna technology allowing better utilisation of the spectrum.

IEEE 802.16a WiMAX standard can transmit at speeds up to 70 Mbps in the 2 to 11 GHz bands; It can also achieve 120 Mbps at short distances in the 10 to 66 GHz bands. It brings full support of mobile devices to WiMAX technology. WiMAX is considered a migration path to 4G, but is more likely to be used by owners of broadband wireless access spectrum rather than cellular mobile carriers. IEEE 802.16a is also expected to play a role in outdoor and private networks, the extension of hotspots, and backhaul applications that lack line-of-sight. Applications include backhaul applications for business, last mile delivery applications, support for simultaneous voice, video, and data transmission, Voice-over-IP (VoIP) connections, broadband access to rural and remote areas, etc.

14.4.4 WiMAX and LTE/3GPP – Comparison

The two prominent mobile broadband technologies are 3G/3G+ cellular and WiMAX technologies. 3G cellular mobile technologies, as defined by IMT2000, are Wideband Code Division Multiple Access (WCDMA), also known as Universal Mobile Telecommunications System (UMTS) in Europe, and Code Division Multiple Access 2000 (Cdma2000). WCDMA is a technology with backward compatibility with GSM, while Cdma2000 is an advanced technology of CdmaOne. WCDMA uses Direct Sequence Spread Spectrum (DSSS) to spread the baseband signal over a 5 MHz spectrum. It is based on the Third Generation Partnership Project (3GPP) Release 99 and provides data rates of 384 kbps for wide area coverage and up to 2 Mbps for hotspot areas. The improvements in WCDMA for data capabilities came in the form of High Speed Packet Access (HSPA) technologies which improved the data speeds of up to 14.4 Mbps for downlinks and 5.76 Mbps for uplinks. Cdma2000 evolutions for data handling capabilities have came in the form of Cdma2000 1x, 1x-EV-DV (Evolutionary Data and Voice), 1x EV-DO (Evolutionary Data Only) and Cdma2000 3x (also called IMT-2000 CDMA MC). It can provide the speed of around 2–4 Mbps and use Orthogonal Frequency Division Multiple Access (OFDMA) technology. The IMT-2000 family of standards support four different multiple access technologies, including FDMA, TDMA, CDMA, and OFDMA (includes WiMAX).

Technological advancements in the form of 3.5G, Long Term Evolution (LTE) or Super 3G and Ultra Mobile Broadband (UMB) are also under consideration in order to meet the demands of increasing data rate applications and value-added services such as interactive video. The prominent features of LTE include

- High data rates of up to 100 Mbps for downlinks and 50 Mbps for uplinks
- Data-centric networks instead of voice-centric networks
- Use of advanced OFDMA technology instead of CDMA
- Horizontally oriented structure instead of vertically oriented
- Flexibility for operators to deploy in different-sized bands according to availability of spectrum
- Higher spectral efficiencies by using advanced antenna systems like MIMO (Multi-Input Multi-Output)

LTE offers several important benefits for users as well as service providers. LTE supports flexible carrier bandwidths, from below 5 MHz up to 20 MHz. LTE also supports both FDD and TDD. LTE radio network products have a number of features such as plug-and-play, self-configuration and self-optimisation that simplify the deployment and management of next-generation networks. LTE will be deployed in parallel with simplified, IP-based core and transport networks that are easier to instal, maintain and introduce services on.

Facts to Know!

One of the requirements on LTE is to provide downlink peak rates of at least 100 Mbps. The technology allows for speeds of over 200 Mbps. Furthermore, RAN (Radio Access Network) round-trip times is less than 10 ms. In fact, LTE is more than any other technology, already meeting key 4G requirements.

LTE defines new radio connections for mobile networks, and will utilise Orthogonal Frequency Division Multiplexing (OFDM), a widely used modulation technique that is the basis for Wi-Fi, WiMAX, and the DVB and DAB digital broadcasting technologies as well.

Facts to Know!

In addition to mobile phones, many computer and consumer electronic devices, such as notebooks, ultra-portables, gaming devices and cameras, will incorporate LTE embedded modules. Since LTE supports hand-over and roaming to existing mobile networks, all these devices can have ubiquitous mobile broadband coverage.

IEEE 802.16d and IEEE 802.16e standards, referred to as WiMAX networks, uses OFDM and supports fixed and nomadic access in line-of sight and Non-line-of-sight environments. IEEE 802.16e further enhanced the ability of WMAN with mobility support. It provides support for hand-offs and roaming. It uses Scalable Orthogonal Frequency Division Multiplexing Access (SOFDMA)—a multi-carrier modulation technique. S-OFDMA allows for an increase in range of channel bandwidths from 1.25 MHz up to 20 MHz. IEEE 802.16e standards provides service providers the ability to offer a wide range of new and revolutionary high-speed mobile applications and services. By using an IEEE 802.16e/WiMAX-based backbone network to connect Wi-Fi hotspots to the Internet, broadband services can be delivered quickly at relatively low costs. The development of Mobile WiMAX has been on the basis of IP which allows seamless compatibility with existing Internet applications. After the inclusion of OFDM-based technologies in the IMT2000 standard by the ITU Mobile WiMAX, it will become more competitive with 3G Cellular for using existing as well as extended bands.

The key features of WiMAX technology include

– Advanced performance (high per-user throughput and low latency)
– Wide variety of devices (laptop add-in cards and modules, game consoles)
– IP-based, optimised for packet-based data applications
– Support for IMS (Internet Multimedia Subsystems)
– Next-generation multiplexing technique
– Support for advanced antenna techniques/systems like MIMO (Multi-Input Multi-Output) and beam forming
– Multiple hand-off mechanisms (supports a variety of hand-off mechanisms)
– Worldwide availability (operates in three spectrum bands: 2.3–2.4 GHz, 2.496–2.69 GHz and 3.4–3.6 GHz)
– Dynamic bandwidth allocation (enabling flexible management of spectrum resources and a more efficient use of spectrum)
– Easy integration with technologies like 2G, 3G and Wi-Fi
– Tolerance to multi-path and self-Interference
– Global roaming (allows subscribers to access different networks using the same device and a single, familiar interface)

– Equipment based on open standards, an attractive Intellectual Property Rights (IPR) structure and high base-station capacity

In order to compare WiMAX and LTE/3GPP technologies, it is assumed that there is an increasing demand for high bandwidth mobile applications and services. Both technologies will deliver the functionalities and speed as claimed, and will be implemented in the market in time. The comparison between these two technologies can be presented in various aspects and phases given below.

Comparison with Existing 3G Technologies WiMAX has an obvious advantage of speed over its competing 3G technologies like HSPA and Cdma2000 3x. WiMAX is ideal for data applications as it is based on IP based packet-switching technology as compared with circuit/packet-switched 3G intended for mix of voice and data applications. On the other hand, 3G is ideally suited for voice services as the quality of service deteriorates in WiMAX under heavy traffic conditions. Moreover, cost of service also goes up exponentially. 3G has also advantage in the field of mobility due to its inherited mobile capabilities while in mobile WiMAX, mobility has been added as an additional feature.

Comparison Based on Spectrum Utilisation The availability and efficient utilisation of spectrum is the most important aspect of wireless systems due to scarcity of this expensive resource. The advantage to 3G technologies is availability of spectrum which can be a great problem for mobile WiMAX. Regulatory restrictions can also pose a problem for WiMAX.

Comparison with LTE/3GPP Market The main advantage to LTE is commercial lead of 3G in Europe and in many other countries. However, there is a concern about reuse of existing 3G infrastructures for LTE due to its completely different air interface. The main advantage to WiMAX is its projected time-to-market which is almost three years ahead of LTE. Mobile WiMAX is a standard-based technology (IEEE 802.16e) while LTE is still not standardised. Standardised technology also may provide easier upgrade paths to future technologies. Another advantage to WiMAX is already presence of interfaces for this technology in many user devices.

Facts to Know!

The success of WiMAX in the market will not only depend on the technology to deliver up to the expectations but will also be affected from the behaviour of regulators and policy makers towards this technology and availability of spectrum.

The comparison of the two technologies shows pros and cons of both technologies. The majority of wireless mobile broadband market will be served by both 3G+ Cellular and WiMAX technologies and LTE, UMB will also be available by that time. WiMAX is ideally suited for providing access to rural and remote areas in both developed and developing countries and is getting popularity in developing countries where operators still have not invested in legacy 3G infrastructure. The hurdles in WiMAX deployment includes unavailability of spectrum, regulatory biasing, lower PC penetration and lower GDP per capita in these developing countries. For improving mobility capabilities, Mobile WiMAX can take the advantage by using the work of proposed IEEE 802.21 hand-off group which will work on the common standard to specify a common hand-off framework applicable to all 802 standards.

The rapid increase in high bandwidth applications and services are increasing pressures on cellular mobile networks. So traditional cellular mobile operators will have to shift the data traffic from their network to ease the congestion on their networks. The better choice for this may be some data-centric standardised network and obviously, the option available in hand to these operators is only the Mobile WiMAX. Another advantage to mobile WiMAX is the fact that by the time LTE will come in the market, there is a possibility that some upgraded version of Mobile WiMAX will also come in the market, hence giving it an edge over its competing technologies for speed and other functionalities. In short, Mobile WiMAX has a great potential to become the mainstream technology and it has become both a threat and opportunity for mobile cellular operators.

Table 14.9 presents a comparative study of some of the key characteristics of the three most popular broadband wireless technologies, that is, WiMAX, Wi-Fi and 3G Cellular Cdma2000 technology.

Table 14.9 *Comparison of WiMAX with Wi-Fi and Cdma2000 technologies*

S. No.	*Parameter*	*Mobile WiMA*	*Wi- i*	*Cdma2000 (1xEV-DO)*
1.	Standard	IEEE 802.16e	IEEE 802.11a/g/n	3GPP2
2.	Allocated Spectrum Band	2.3/2.5/3.5 GHz	2.4/5 GHz	800/900/1800/1900 MHz
3.	RF bandwidth	3.5/5/7/8.75/10 MHz	200 MHz for 802.11a/g; 20/40 MHz for 802.11n	1.25 MHz
4.	Multiple access	TDM/OFDMA	CSMA	CDMA
5.	Duplexing	TDD initially	TDD	FDD
6.	Modulation type	QPSK, 16-QAM, 64-QAM	BPSK, QPSK, 16-QAM, 64-QAM	QPSK, 8-PSK, 16-QAM
7.	Downlink peak data rate	32–46 Mbps	802.11a/g : 54 Mbps; 802.11n : >100 Mbps	3.1–4.9 Mbps
8.	Uplink peak data rate	4–7 Mbps	Same as Downlink Peak Data Rate	1.8 Mbps
9.	Mobility	Moderate	Low	High

14.5 MOBILE AD-HOC NETWORKS (MANETs)

Wirelessly connected devices are creating a revolution in the way networked resources can interact with each other. The concept of ad-hoc networking is one of the advanced mechanisms used for wireless networking. Ad-hoc networks consist of a collection of wireless nodes. These nodes are connected with each other to dynamically establish an ad-hoc or on-the-fly network without the support of any centralised infrastructure. Such a network supports anytime and anywhere mobile computing, allowing the spontaneous formation of mobile networks for the period of usage. In such a network, each mobile host acts as a router which enables peer-to-peer as well as peer-to-remote wireless communications.

14.5.1 MANET Topology

Mobile ad-hoc networks (MANETs) are collections of mobile nodes dynamically establishing short-lived networks in the absence of fixed infrastructure. Each mobile node is equipped with a wireless transmitter and a receiver with an appropriate antenna. These mobile nodes are connected by wireless links and act as routers for all other mobile nodes in the network. Nodes in mobile ad-hoc networks are free to move and organise themselves in an arbitrary manner. These features make MANETs very practical and easy to deploy in places where existing infrastructure is not capable enough to allow communication, for instance, in disaster zones, or infeasible to deploy locations.

MANETs are the short term temporary spontaneously wireless networks of mobile nodes communicating with each other without the intervention of any fixed infrastructure or central control. It is an autonomous system of mobile nodes, mobile terminals, or mobile stations serving as routers interconnected by wireless links. Depending on the locations, antenna coverage patterns, transmit power levels, and cochannel interference levels; a wireless connectivity exists among participating mobile nodes at a given time, either in the form of random multihop transmissions or ad-hoc network. Network communications and management tasks are typically performed in a distributed manner. As the nodes move or adjust their transmission and reception parameters, MANET topology may change from time to time.

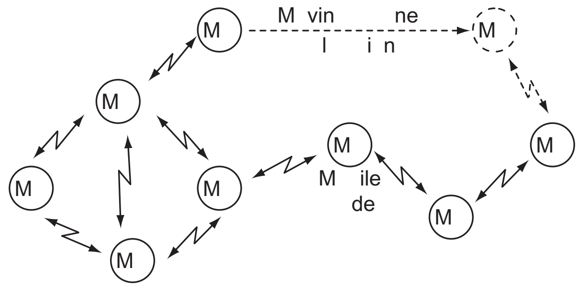

Fig. 14.11 Formation of MANET

An ad-hoc mobile wireless network is a network without any base stations, that is, an infrastructure-less network. Figure 14.11 depicts the formation and operation of a MANET. Data packets are transmitted in a store-and-forward method from the source node to the destination node in peer-to-peer multihop intermediate nodes acting as routers. The network may either operate as standalone or as an extension of an infrastructure network with the help of few selected routers.

With comparison to other wireless networks, mobile ad-hoc networks have unique operational characteristics such as the following:

Dynamic Topology Nodes are free to move in an arbitrary manner as per the user requirements, thereby resulting in the dynamic network topology. The incorrect topology information can considerably increase the end-to-end-delay and routing control overhead, increase the possibility of node failure, and reduce the capacity. Therefore, network topology management plays a key role in maintaining dynamic configuration of the network and the performance of a routing protocol in MANETs.

EXAMPLE 14.10 Communication Links in MANETs

A given MANET consists of 100 mobile nodes. The mobility of the nodes is such that two existing wireless links are broken, while two new wireless links are established every one second. Assume that each mobile node is connected to exactly four adjacent mobile nodes. Compute the total number of wireless links in the network.

Solution

Number of mobile nodes = 100 (given)

Since each mobile node is connected to exactly four adjacent mobile nodes, therefore,

Maximum possible number of wireless links = $4 \times 100 = 400$

Due to mobility of the nodes, there is no change in the maximum possible number of wireless links at any time because it is specified that two existing wireless links are broken, while two new wireless links are established every one second, thereby maintaining the number of wireless links.

As each wireless link involves two mobile nodes connected to each other,

Total number of wireless links in the network = $400/2 = 200$

Unpredictable Link Properties Signal propagation in wireless media is fairly unpredictable due to problems caused by signal fading, interference and multipath cancellation. Packet collision is intrinsic to wireless network.

Limited Bandwidth Generally, wireless networks are bandwidth limited.

Limited Power Mobile devices and nodes are usually battery operated. This demands the use of extremely low-power components in the device. The limited availability of battery power affects transceiver input/output power, and CPU/signal processing.

Limited Security The physical wireless medium of communication is inherently insecure and vulnerable to attack. Without provisions of adequate security, unauthorised access and usage of ad-hoc networks may violate QoS and network performances.

14.5.2 Ad-hoc Routing Protocols

An ad-hoc routing protocol is a standard that controls how mobile nodes decide which way to route packets between computing devices in a mobile ad-hoc network. In MANETs, mobile nodes are not familiar with the topology of their networks in the beginning; instead, they have to discover it. The basic idea is that a new mobile node may announce its presence and should listen for announcements broadcast by its neighbours. Each node learns about nodes nearby and how to reach them, and may announce that it too can reach them. Ad-hoc network routing protocols can be classified as the following:

Proactive (Table-Driven) Routing Protocols This type of protocols maintain fresh lists of destinations and their routes by periodically distributing routing tables throughout the network. The main disadvantages of such algorithms include respective amount of data for maintenance, and slow reaction on restructuring and failures. An example of proactive routing protocol is highly dynamic Destination-Sequenced Distance Vector (DSDV) routing protocol.

Reactive (On-Demand) Routing Protocols This type of protocol finds a route on demand by flooding the network with route request packets. The main disadvantages of such algorithms are high latency time in route finding and network clogging due to excessive flooding. Examples of reactive routing protocol is Dynamic Source Routing (DSR) protocol and Ad Hoc On-Demand Distance Vector (AODV) routing protocol.

Hybrid (Both Proactive and Reactive) Routing Protocols This type of protocols combine the advantages of proactive and reactive routing. The routing is initially established with some proactively prospected routes and then serves the demand from additionally activated nodes through reactive flooding. The choice for one or the other method requires predetermination for typical cases. The advantage depends on the amount of nodes activated and reaction to traffic on gradient of traffic volume. An example of hybrid routing protocol is the Zone Routing Protocol.

Hierarchical Routing Protocols With this type of protocols the choice of proactive and of reactive routing depends on the hierarchic level where a mobile node resides. The choice for one or the other method requires proper attributation for respective levels. The advantage depends on depth of nesting and addressing scheme and reaction to traffic demand on meshing parameters. An example of hierarchical routing protocol is the Cluster Based Routing Protocol (CBRP).

Adaptive (Situation-Aware) Routing Protocols This type of protocols also combines the advantages of proactive and reactive routing. The routing is initially established with some proactively prospected routes and then serves the demand from additionally activated nodes through reactive flooding. Some metrics must support the choice of reaction. The advantage depends on the amount of nodes activated and reaction to traffic demand on gradient of traffic volume. An example of adaptive routing protocols is Temporally-Ordered Routing Algorithm (TORA) routing protocol.

Flow-oriented Routing Protocols This type of protocols find a route on demand by following present flows. One option is to unicast consecutively when forwarding data while promoting a new link. The main disadvantages of such protocols are that it takes long time when exploring new routes without apriori knowledge, and may refer to entitative existing traffic to compensate for missing knowledge on routes. Examples of flow-oriented protocols are Preferred link based routing protocol, QoS aware source initiated ad-hoc routing protocol, and Multipath On-demand Routing Protocol.

14.5.3 Challenges and Issues

The absence of fixed infrastructure in MANETs poses a number of different challenges and issues such as mobility, security, bandwidth constraints, hidden and exposed node problems, and routing mechanisms.

Mobility in ad-hoc networks causes frequent link failure, which in turn causes packet losses. Transmission control protocol treats loss or delay of a package acknowledgement as traffic congestion. It is considered as a complex task in mobile ad-hoc networks. Some of the most challenging issues are

– Limited wireless transmission range
– Broadcast nature of the wireless medium
– MAC related issues such as hidden terminal and exposed terminal problems
– Routing problem due to change in route because of node mobility
– Packet losses due to transmission errors and mobility
– Battery constraints
– Security issues leading to ease of snooping

The most challenging issue is the design of the MAC protocols which define how the wireless medium is shared by all nodes. Due to the nature of the network, a distributed random access MAC is preferred over a centralised MAC. CSMA is one of the earliest mechanisms adopted for ad hoc networks. In CSMA, a transmitter will first sense the wireless channel in the vicinity and refrain itself from transmission if the channel is already in use. Various methods such as ALOHA and *n*-persistent CSMA algorithms can be used to determine how long the deferred node should wait before the next attempt. However, distributed random access protocols such as CSMA suffer from hidden and exposed nodes problems. It is assumed that each node can communicate with another node only if there is a link between them. In a typical exposed node problem, a node within the range of the transmitter may be unnecessarily prohibited from accessing the medium and thus degrades the network throughput.

Hidden-Node Problem Consider a network scenario comprising of three mobile nodes S1, R1, and S2, as shown in Fig. 14.12. It is assumed that all the nodes transmit at the same power. The circles around these nodes depict the radio-coverage area. As shown in the figure, the node R1 is located in the middle of nodes S1 and S2, In other words, the node R1 lies in the radio-coverage areas of the node S1 as well as the node S2.

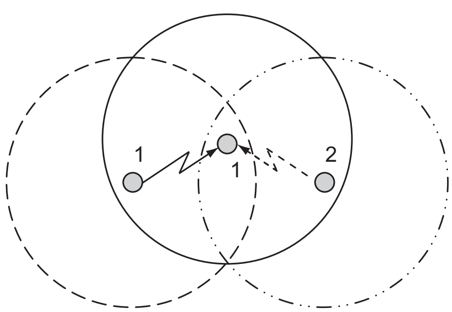

Fig. 14.12 *Hidden-node problem in MANET*

Node R1 can receive transmissions from both nodes S1 and S2; but S1 and S2 cannot receive transmissions from each other because S1 happens to be out of range of S2 and similarly, S2 is out of range of S1. When the node S1 transmits to the node R1, the node S2 cannot detect this transmission using the carrier sense mechanism. If S2 also transmits to R1 then this transmission will collide with that of from S1 for R1. Both transmissions will be corrupted at the node R1. So the node R1 can be a said to be a hidden nodes for nodes S1 and S2. This will increase data packet collisions and hence reduce throughput. CSMA/CA with RTS (request-to-Send)/CTS (Clear-to-Send) mechanism helps to solve the hidden-node problem.

Exposed-Node Problem CSMA/CA with RTS/CTS mechanism resolves the hidden-node problem only if the nodes are synchronised. When a node receives an RTS data packet from a neighboring node, but not the corresponding CTS data packet, that node can deduce that it is an exposed node and is permitted to transmit to other neighboring nodes. If the nodes are not synchronised, the problem may occur that the sender will not receive the CTS data packet or the ACK packet during the transmission of data of the second node.

In wireless networks, the exposed-node problem occurs when a node is prevented from sending packets to other nodes due to a neighbouring transmitter. Consider an example of 4 nodes labeled R1, S1, S2, and R2, as shown in Fig. 14.13, where the two nodes are out of range of each other (Node R1 is out of range of Node R2 and vice versa), yet the two nodes in the middle (Node S1 and Node S2) are in range of each other.

If a transmission between nodes S1 and R1 is taking place, the node S2 is prevented from transmitting to the node R2 as it concludes, based on carrier sense, that it will interfere with the transmission by its neighbouring node S1. However, the node R2 could still receive the transmission of the node S2 without interference because it is out of range from the node S1.

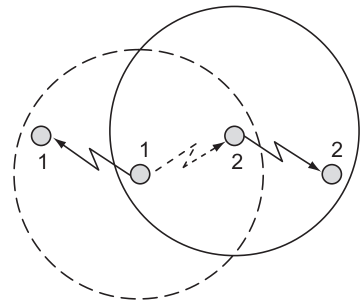

Fig. 14.13 *Exposed-node problem in MANET*

When the node S1 transmits to the node R1, the node S2 detects this transmission using carrier sense mechanism. Node S2 refrains from transmitting to the node R2, hence the node S2 is exposed to S1's transmission. This situation reduces bandwidth utilisation and hence reduces throughput. The possible solution is the use of directional antennas, and separate channels for control and data. Modified CSMA/CA with RTS/CTS mechanism also helps to resolve the exposed-node problem. The duration of data transfer is included in RTS and CTS control packets itself, which instructs other nodes not to transmit for this duration. If a RTS/CTS packet collides, the nodes wait for a random time which is calculated using binary exponential back-off algorithm. This scheme is also called Multiple Access Collision Avoidance (MACA). The only drawback is that it cannot avoid RTS/CTS control packet collisions.

To provide sufficient resources for many applications such as videoconferencing, voice over IP, streaming audio/video in wireless networks, a QoS routing protocol has to be used to carefully choose routing paths with sufficient resources. This becomes more critical in ad-hoc networks due to the need to minimise the use of the shared radio resources over multiple wireless hops and dynamic routing topology.

In many situations, a mobile node wishing to find a route with certain QoS parameters within available bandwidth or given maximum latency may not be able to find a route with sufficient quality. As an example, in the QoS routing scheme, the QoS Guided Route Discovery (QoS-GRD) allows a node to specify QoS metrics that must be satisfied by a discovered path. When a mobile node has a pre-existing route, it may either use QoS-GRD or it may try to establish a new flow along the pre-existing route.

Developing some enhancements for routing protocols such as On Demand Multicast Routing Protocol could deliver stable route selection, efficient route acquisition, optimised transmission of control packets, reliable hop-by-hop transmission, routes reconstruction according to topology changes, and improved node movement pattern. To achieve these objectives, some techniques used are mobility and connectivity prediction, route refresh interval to mobility patterns and speeds, periodically measuring nodes transmission power samples from packets, selecting the most stable path instead of using minimum delay path, and establishing and refreshing multicast routes.

14.5.4 Typical Applications of MANETs

There are numerous application areas in which MANETs can be effectively used. Some of the key applications are

– Personal area networking formed with cellphones, laptops, notebooks, PDAs
– Education sector such as virtual classrooms, conferences, seminars
– Sensor networks for homes, environmental applications, wearable computing
– Civilian environments like meeting rooms, sports stadiums, hospitals
– Military environments in battlefields by soldiers in tanks/planes
– Emergency operations such as search-and-rescue in case of natural disasters

MANETs have the capability of providing advanced features such as data rates comparable with multimedia applications (entertainment), global roaming (through satellite networks), and coordination with other wired/wireless networks. This will create many new applications in future.

14.6 MOBILE IP AND MOBILITY MANAGEMENT

The Mobile Internet Protocol (Mobile IP) is the underlying technology for support of various mobile data and wireless networking applications. With the advent of packet based mobile data applications and the increase of wireless computing, there is a need for the seamless communication between the mobile node device and the packet data network such as the Internet. The Internet infrastructure is built on top of the TCP/IP protocol suite. IP requires the location of any host connected to the Internet to be uniquely identified by an assigned IP address. This raises one of the most important issues in mobility, because when a host moves to another physical location, it has to change its IP address. However, the higher level protocols require the IP address of a host to be fixed for identifying connections.

Facts to Know!

The term 'mobile' in Mobile IP signifies that while a user is connected to applications across the Internet, the user's point of attachment changes dynamically. All wireless communication links are maintained despite the change in underlying network properties.

Mobile IP is an extension to the Internet Protocol proposed by the Internet Engineering Task Force (IETF) that addresses the mobility issue. It enables mobile nodes to stay connected to the Internet regardless of their location and without changing their IP address. More precisely, Mobile IP is a standard protocol that builds on the Internet protocol by making mobility transparent to applications and higher level protocols like TCP. Thus, the objective of Mobile IP is to support end-system mobility while maintaining compatibility with lower layers and transparency to higher layers of TCP/IP standards, large scalability of mobile nodes, optimum bandwidth efficiency, and security including authentication.

14.6.1 Mobile IP Terminology

Mobile IP is a protocol that keeps track of the mobile node's current location and delivers Internet messages there. The operation of Mobile IP uses many new functional entities and terms related to them, which are summarised below.

Mobile Node (MN) A mobile node is a wireless host or a router capable of performing network roaming using Mobile IP, and is an end-system entity that changes its point of attachment to the Internet.

Home Address It is a permanent fixed IP address that is assigned for an extended period of time to the mobile node. It resides in its home network, and is used by TCP and higher level layers.

Home Network It is a network which is identified by the home address of the mobile node. No mobile IP support is needed for a mobile node within the home network.

Home Agent (HA) It is a router on a mobile node's home network which serves as a point for communications with the mobile node when it is away from home, and maintains current location information for the mobile node. The HA can intercept packets destined to the mobile node's home address and tunnel them to the mobile node's current location. It is used with one or more foreign agents.

Care-of-Address (COA) The care-of address of a mobile node is the IP address when operating in a foreign network.

Collocated Care-of-Address It is an externally obtained local IP address temporarily assigned to an interface of the mobile node.

Foreign Agent Care-of-Address The IP address of a foreign agent, which has an interface on the foreign network being visited by the mobile node. It can be shared by many mobile nodes simultaneously.

Foreign Network It is a network other than the mobile node's home network to which the mobile node is currently connected.

Foreign Agent (FA) It is a default router that stores information about mobile nodes visiting its network. It functions as the mobile node's point of attachment when it moves to the foreign network, to provide services to the mobile node. Foreign agents also advertise care-of-addresses which are used by Mobile IP to inform its home agent.

Node A node is a host or a router.

Correspondent Node (CN) The CN can be a fixed or mobile peer host with which the mobile node communicates. For example, a web server.

Tunnel Tunnel is the path followed by the datagram when it is encapsulated.

Binding The association of the home address with a care-of address is called a binding.

Binding Entry It is an entry in the home agent's routing table. Mobile IP maps the mobile node's home address into its current care-of-address.

Link It is a facility or medium over which mobile nodes can communicate at link layer.

Mobile Node's Home Link It is the link which has been assigned the same network-prefix as the network-prefix of the mobile node's home address.

Mobile Node's Foreign Link It is the link that the mobile node is visiting, which has been assigned the same network-prefix as the network-prefix of the mobile node's care-of-address.

Link-layer Address It is an address that identifies the interface's MAC address.

Agent Advertisement It is the process in which the foreign agent advertises their presence by using a special message.

Agent Solicitation It is the message sent by a mobile node to request agent advertisement.

14.6.2 Operation of Mobile IP

Internet messages, destined for the mobile node, are always sent to the mobile node's permanent home address in the mobile node's home network. The Mobile IP is designed to deliver Internet messages from the mobile node's home network to the mobile node in its current location in a seamless manner. To accomplish the identification and routing of messages, Mobile IP allows each mobile node to have two IP addresses and by maintaining the binding between the two addresses transparently. A permanent home address for identification and a temporary care-of-address that represents the current location of the mobile node for routing are used. Mobile IP uses an agent concept. The mobile node has a Home Agent (HA) and a Foreign Agent (FA). The HA maintains a database in which the mobile node's home address resides. When it moves to a foreign network, it establishes an association with its foreign agent which, in turn, establishes an association with the mobile node's home agent. Figure 14.14 shows the functional relationships among different entities and the operation of Mobile IP.

A mobile node wanting to communicate with another mobile node uses the permanent home address of the mobile node as the destination address to send packets. Because the home address logically belongs to the network associated with the home agent, normal IP routing mechanisms forward these packets to the home agent. Instead of forwarding these packets to a destination that is physically in the same network as the home agent, the home agent redirects these packets towards the foreign agent through an IP tunnel by encapsulating the datagram with a new IP header using the care-of address of the mobile node.

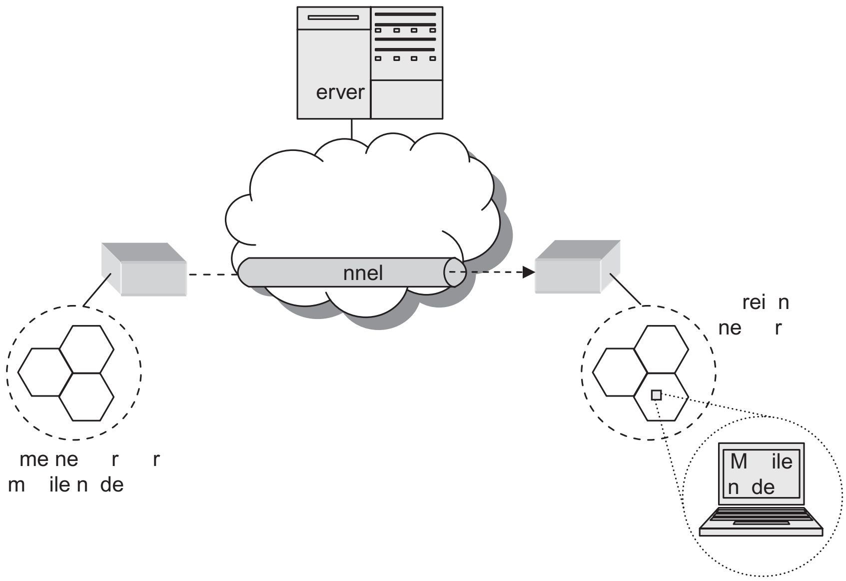

Fig. 14.14 Mobile IP operation

The Mobile IP protocol defines an authenticated registration procedure by which a mobile node informs its home agent(s) of its care-of-address(es); an extension to ICMP Router Discovery, which allows mobile nodes to discover prospective home agents and foreign agents; and the rules for routing packets to and from mobile nodes, including the specification of one mandatory tunneling mechanism and several optional tunneling mechanisms. These form the basis for location and mobility management in Mobile IP.

14.6.3 Location Management in Mobile IP

Location Management in Mobile IP is achieved via agent discovery and registration process. Mobile agents (home agents and foreign agents) advertise their presence by periodically broadcasting agent advertisement messages. An agent advertisement message lists one or more care-of addresses and a flag indicating whether it is a home agent or a foreign agent. The same agent may act as both an HA and an FA mobility extension to ICMP messages which are used for agent advertisement. The mobile node receiving the agent advertisement message observes whether the message is from its own home agent and determines whether it is on the home network or a foreign network. The packet format of Mobile IP agent advertisement message is shown in Fig. 14.15.

If a mobile node does not wish to wait for the periodic advertisement, it can send out agent solicitation messages using ICMP that will be responded by a mobile agent. Figure 14.16 shows the agent discovery procedure in a simple flow diagram.

If a mobile node discovers that it is on the home network, it operates without any mobility services. If the mobile node is on a new network, it registers with the foreign agent by sending a registration request message which includes the permanent IP address of the mobile host and the IP address of its home agent. The foreign agent in turn performs the registration process on behalf of the mobile host by sending a registration request containing the permanent IP address of the mobile node and the IP address of the foreign agent to the home agent. When the home agent receives the registration request, it updates the mobility binding by associating the care-of address of the mobile node with its home address. The home agent then sends an acknowledgement to the foreign agent. A registration reply message indicates whether the registration is successful or not. If an MN does not know the HA address, it will send a broadcast registration request to its

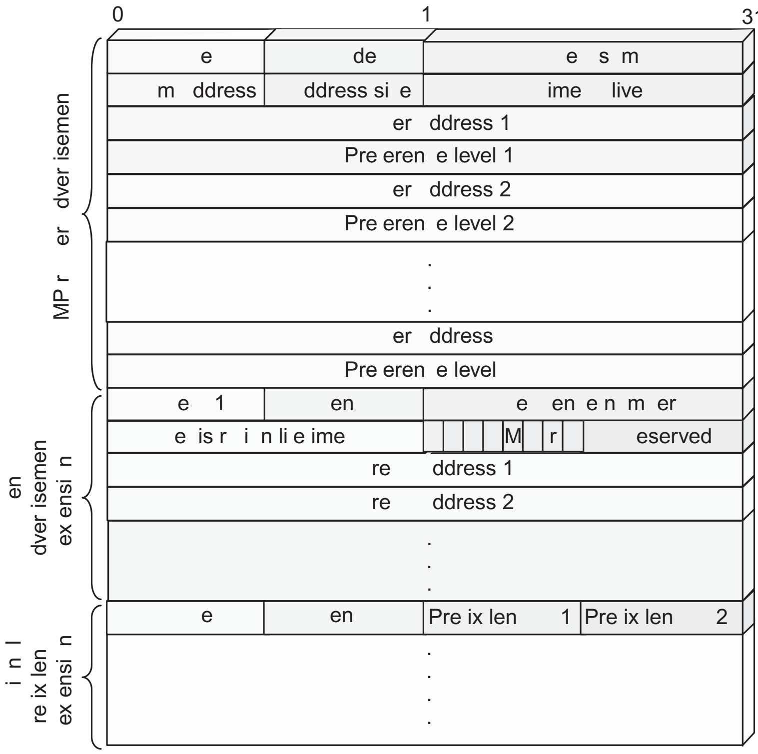

Fig. 14.15 | *Mobile IP agent advertisement message*

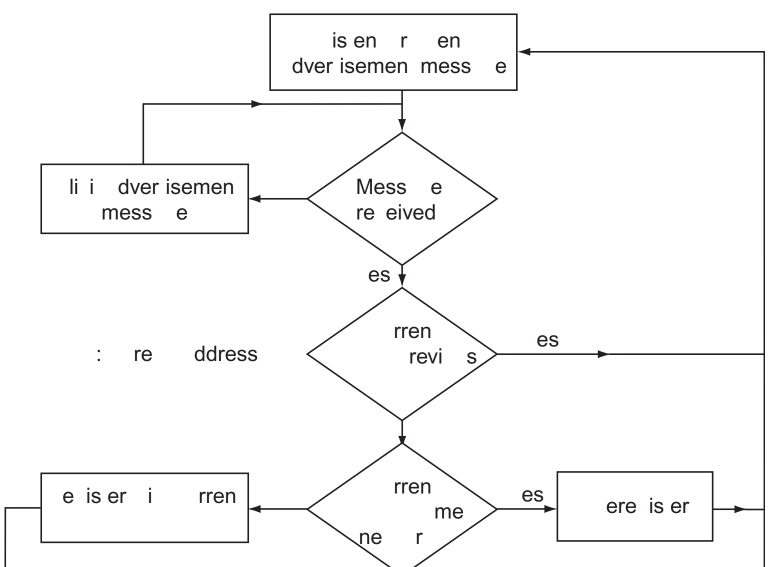

Fig. 14.16 | *Agent discovery procedure*

home network called a *directed broadcast*. The foreign agent in turn updates its visitor list by inserting the entry for the mobile node and relays the reply to the mobile node.

14.6.4 Mobility Management in Mobile IP

When a correspondent node wants to communicate with the mobile node, it sends an IP packet addressed to the permanent IP address of the mobile node. The home agent intercepts this packet and consults the mobility binding table to find out if the mobile node is currently visiting any other network. The home agent finds out the mobile node's care-of address and constructs a new IP header that contains the mobile node's care-of address as the destination IP address. The original IP packet is put into the payload of this IP packet. It then sends the packet. This process of encapsulating one IP packet into the payload of another is known as IP-within-IP encapsulation, or tunneling. When the encapsulated packet reaches the mobile node's current network, the foreign agent decapsulates the packet and finds out the mobile node's home address. It then consults the visitor list to see if it has an entry for that mobile node. If there is an entry for the mobile node on the visitor list, the foreign agent retrieves the corresponding media address and relays it to the mobile node. When the mobile node wants to send a message to a correspondent node, it forwards the packet to the foreign agent, which in turn relays the packet to the correspondent node using normal IP routing. The foreign agent continues serving the mobile node until the granted lifetime expires. If the mobile node wants to continue the service, it has to reissue the registration request.

When acting as source, a mobile node sends packets directly to the other correspondent node through the foreign agent, without sending the packets through the home agent, using its permanent home address as the source address for the IP packets. This is known as triangle routing. Figure 14.17 illustrates triangle routing.

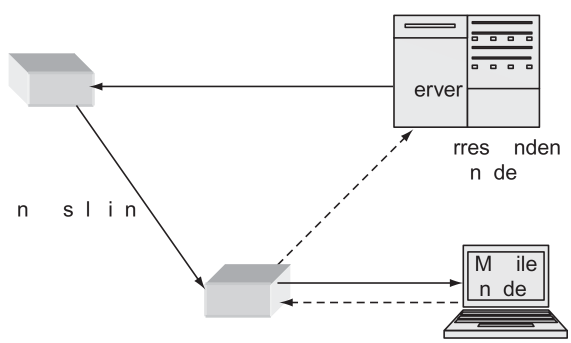

Fig. 14.17 *Triangle routing*

If needed, the foreign agent could employ reverse tunneling by tunneling the mobile node's packets to the home agent, which in turn forwards them to the correspondent node. This is needed in networks whose gateway routers have ingress filtering enabled and hence the source IP address of the mobile host would need to belong to the subnet of the foreign network or else the packets will be discarded by the router.

The triangle routing method may not be efficient in many cases. Consider the case when the correspondent node and the mobile node are in the same network, but not in the home network of the mobile node. In this case, the messages will experience unnecessary delay since they have to be first routed to the home agent that resides in the home network. One way to improve this is route optimisation.

Route optimisation is an extension proposed to the basic Mobile IP protocol. Here, messages from the correspondent node are routed directly to the mobile node's care-of address without having to go through the home agent. Route optimisation provides four main operations such as updating binding caches, managing smooth hand-offs between foreign agents, acquiring registration keys for smooth hand-offs, and using special tunnels. Binding caches are maintained by correspondent nodes for associating the home address of a mobile node with its care-of address. A binding cache entry also has an associated lifetime after which the entry has to be deleted from the cache. If the correspondent node has no binding cache entry for a mobile node, it sends the message addressed to the mobile node's home address. When the home agent intercepts this message, it encapsulates it and sends it to the mobile node's care-of address. It then sends a binding update message to the correspondent node informing it of the current mobility binding.

When a mobile node registers with a new foreign agent, the basic Mobile IP does not specify a method to inform the previous foreign agent. Thus, the datagrams in process which had already tunneled to the old care-of address of the mobile node are lost. This problem is solved in route optimisation by introducing smooth hand-offs. Smooth hand-off provides a way to notify the previous foreign agent of the mobile node's new mobility binding. If a foreign agent supports smooth hand-offs, it indicates this in its agent advertisement message. When the mobile node moves to a new location, it requests the new foreign agent to inform its previous foreign agent about the new location as part of the registration procedure. The new foreign agent then constructs a binding update message and sends it to the previous foreign agent of the mobile node. Thus, if the previous foreign agent receives packets from a correspondent node having an out-of-date binding, it forwards the packet to the mobile node's care-of address. It then sends a binding warning message to the mobile node's home agent. The home agent in turn sends a binding update message to the correspondent node. This notification also allows datagrams sent by correspondent nodes having out-of-date binding cache entries to be forwarded to the current care-of address. Finally, this notification allows any resources consumed by the mobile node at the previous foreign agent to be released immediately, instead of waiting for the registration lifetime to expire.

For managing smooth hand-offs, mobile nodes need to communicate with the previous foreign agent. This communication needs to be done securely as any careful foreign agent should require assurance that it is getting authentic hand-off information and not arranging to forward in-process datagrams to an unknown destination. For this purpose, a registration key is established between a foreign agent and a mobile node during the registration process. When a foreign agent receives a tunneled datagram for which it has no visitor list entry, it concludes that the node sending the tunneled datagram has an out-of-date binding cache entry for the mobile node. If the foreign agent has a binding cache entry for the mobile node, it should re-tunnel the datagram to the care-of address indicated in its binding cache entry. On the other hand, when a foreign agent receives a datagram for a mobile node for which it has no visitor list or binding cache entry, it constructs a special tunnel datagram. The special tunnel datagram is constructed by encapsulating the datagram and making the outer destination address equal to the inner destination address. This allows the home agent to see the address of the node that tunneled the datagram and prevent sending it to the same node. This avoids a possible routing loop that might have occurred if the foreign agent crashed and lost its state information.

> **Facts to Know!**
>
> Usually, the foreign agent allocates the care-of-address. However, it is possible that a mobile node moves to a network that has no foreign agents or all foreign agents are busy. A viable solution is that the mobile node may act as its own foreign agent by using colocated care-of-address.

For highly mobile nodes, the amount of registration traffic generated between the foreign and home networks can be quite large. In hierarchical routing, a hierarchy of foreign agents is established in a tree structure and multiple foreign agents are advertised in the agent advertisement message. In this way, registration can be localised to the foreign agent that is the lowest common ancestor of the care-of-addresses at the required points of attachment.

Deregistration If a mobile node wants to drop its care-of address, it has to deregister with its home agent. It achieves this by sending a registration request with the lifetime set to zero. There is no need for deregistering with the foreign agent as registration automatically expires when lifetime becomes zero. However, if the mobile node visits a new network, the old foreign network does not know the new care-of address of the mobile node. Thus, datagrams already forwarded by the home agent to the old foreign agent of the mobile node are lost.

14.6.5 Mobile IPv6

The Internet's explosive growth has made the TCP/IP protocol suite's technology limitations increasingly apparent mainly because of the weaknesses of the Internet Protocol (the IPv4 in TCP/IP) itself. IPv4 lacks two critical attributes: scalability and security. IP's scalability problems centre around three key issues: address allocation, backbone routing table growth, and host configuration. Perhaps IPv4's most widely reported deficiency is its limited addressing architecture. IPv4's Class A, B, C address allocation architecture is fundamentally inefficient. Given historical Internet growth trends, unassigned addresses are expected to be exhausted soon.

IPv6 is an evolutionary step from IPv4. In addition to functions provided by IPv4, the primary changes from IPv4 to IPv6 includes expanded addressing and routing, autoconfiguration, authentication and confidentiality capabilities. The IPv6 addresses are 128 bits long. The autoconfigure means to divide the IP address into two distinct parts: network identifier and a unique host identifier, using the host's IEEE-802 48-bit MAC address. IPv6 establishes three important security services: packet authentication, packet integrity, and packet confidentiality. All security functions are achieved using IPv6's optional extension header.

Facts to Know!

The Mobile IP protocol allows location-independent routing of IP datagrams on the Internet. Mobility support in IPv6 solves many of the problems of basic Mobile IP.

Fixed IPv6, however, cannot be used with Mobile IP. When an IPv6 node changes its location, it might also change its link and IPv6 address in order to maintain connectivity. Due to address changes, the existing connections of the mobile node that are using the address assigned from the previously connected link cannot be maintained and are ungracefully terminated.

The key benefit of Mobile IPv6 is that even though the mobile node changes locations and addresses, the existing connections through which the mobile node is communicating are maintained. To accomplish this, connections to mobile nodes are made with a specific address that is always assigned to the mobile node, and through which the mobile node is always reachable. Mobile IPv6 provides transport layer connection survivability when a mobile node moves from one link to another by performing address maintenance for mobile nodes at the Internet layer. Changes from IPv6 to Mobile IPv6 are a set of mobility options to include in mobility messages, a new Home Address option for the Destination Options header, a new Type 2 Routing header, new Internet Control Message Protocol for IPv6 (ICMPv6) messages to discover the set of home agents and to obtain the prefix of the home link, and changes to router discovery messages and options and additional neighbour discovery options. Some advantages of Mobile IPv6 over Mobile IPv4 are the following:

- Route optimisation is built as a fundamental part of Mobile IPv6 unlike Mobile IPv4 where it is an optional set of extensions that may not be supported by all nodes.
- Foreign agents are not needed in Mobile IPv6. The enhanced features of IPv6 like neighbour discovery and address auto-configuration enable mobile nodes to function in any location without the services of any special router in that location.
- In Mobile IPv4, when a mobile node communicates with a correspondent node, it puts its home address as the source address of the packet. Thus, ingress filtering routers used to filter out the packets as the source address of the packet is different from the network from which the packet originated. This problem is tackled in Mobile IPv6 by putting the care-of address as the source address and having a home address destination option, allowing the use of the care-of address to be transparent over the IP layer.

14.6.6 Applications of Mobile IP

Mobile IP is most often found in wired and wireless environments where users need to carry their mobile devices across multiple LAN subnets. It may, for example, be used in roaming between overlapping wireless systems such as Digital Video Broadcasting (DVB), WLAN, WiMAX and Broadband Wireless Access (BWA). Currently, Mobile IP is not required within cellular systems including 3G digital cellular systems, to provide transparency when Internet users migrate between cellular towers, since these systems provide their own data link layer, handover and roaming mechanisms. However, it is often used in 3G systems to allow seamless IP mobility between different Packet Data Serving Node (PDSN) domains.

In many applications such as Virtual Private Network (VPN) and Voice over Internet protocol (VoIP), sudden changes in network connectivity and IP address can cause problems.

14.7 MOBILE TCP

The Transmission Control Protocol (TCP) supports reliable and in-sequence data transport by establishing a connection between the source and destination in Internet. TCP is a reactive control method. A congestion window with dynamically adjusted size is used by the protocol to regulate the traffic flow. Although TCP was initially designed and optimised for wired networks, the growing popularity of wireless and mobile data applications in 3G wireless networks requires extending TCP to wireless and mobile communications as well. Incorporating end-to-end congestion control for wireless networks necessitate the use of TCP over wireless networks too.

> **Facts to Know!**
>
> As signaling system #7 (SS #7) is the glue for telecommunication network, whereas TCP/IP is the glue to the Internet.

The initial objective of TCP was to efficiently use the available bandwidth in the network and to avoid overloading the network and the resulting packet losses due to network congestion. TCP was designed to work well in wired networks with low channel errors. TCP keeps track of the round-trip time for the data packets and the corresponding acknowledgement packets within specified timeout value.

14.7.1 Limitations of TCP

- TCP provides reliable data transfer, but it transmits data in a sequence. However, some applications may need reliable data transfer, though not necessarily in a strict sequence. The strict sequence maintenance in TCP not only makes partial ordering of data impossible, it also causes unnecessary delay in the overall data delivery.
- Some applications prefer partial ordering of data, wherein ordering is maintained only within sub flows of data. Moreover, if a single packet is lost, delivery of subsequent packets is blocked until the lost TCP packet is delivered.
- TCP transmits data in a stream. This requires that applications add their own record marking, to delineate their messages. Applications must ensure that a complete message is transferred in reasonable time.
- In a TCP connection, each host includes a single network interface, and a connection is established between the network interfaces of the two hosts. As a result, if the connection breaks because of a path failure, data becomes unavailable until the connection is re-established.
- TCP is vulnerable to Denial of Service (DoS) attacks, such as SYN flood attacks. A DoS occurs when a malicious host forges an IP packet with a fake IP address and sends a large number of TCP SYN messages to the victim host. When the TCP stack is flooded with multiple SYN messages, the victim host can run out of resources and fail to service the new legitimate SYN messages.
- Wireless networks are characterised by high channel error rates and frequent hand-offs. When TCP is used in wireless networks, all the losses irrespective of whether they are due to channel errors or hand-offs are

treated as congestion losses. In response, the sending rate is unnecessarily reduced in an attempt to relieve the congestion, resulting in degraded performance.

- Delay Spike is another problem in wireless links, especially wireless WAN. When Delay Spike occurs, TCP suddenly suffers one extremely large delay. As a result, TCP will experience one long period without ACK arrives when Delay Spike occurs. The main reasons for Delay Spike are handover, recovery from link outage and blocking by circuit-switched call and high priority data. These events occur frequently in wireless links.
- TCP performance is often unsatisfactory when used in wireless and mobile networks, because of considerable variation in the radio link quality in time due to channel fading, user mobility, and hand-off procedures.
- On wireless links, TCP assumes congestion if packets are dropped due to transmission errors. In addition, mobility including hand-off can also cause packet loss. For example, if a mobile node roams from one access point (for example, foreign agent in Mobile IP) to another access point while there are still data packets in transit to the undesired access point and packet forwarding is not possible. The TCP congestion may unnecessarily degrade the overall throughput. Delay Spike and high variable delays and limited bandwidth are the most severe problems.

14.7.2 Modified TCP for Mobile Applications

Several solutions have been proposed to improve the performance of TCP over wireless and mobile networks. The simple technique is to detect and avoid spurious timeouts as well as spurious retransmissions by implementing time stamping. Since delay variability induces spurious timeouts, one effective way to maintain throughput performance is to absorb the delay variability by a sufficiently large timeout value by artificially injecting additional delay to some.

These solutions can be categorised broadly as

- Indirect TCP (split-connection approach)
- Snooping TCP (TCP-aware link layer protocols)
- Explicit loss notification approaches
- M-TCP (end-to-end protocols)

Indirect TCP It uses the split-connection approach. It is a simple protocol in which the connection between a fixed sender and a mobile receiver is split into two connections—one between the sender and the base station and the other between the base station and the mobile receiver with a proxy serving as a common point between the two connections. By doing this the loss recovery due to channel errors on the wireless link is local. This helps isolate impacts of packet errors and delay variability for the wireless link from the wired connection so that TCP congestion control, timeout and retransmission mechanisms in the wired link do not suffer from the fluctuating quality of the radio channel. Also, in case of hand-offs, the original serving base station transfers the connection information to the new target base station in a manner transparent to the sender. The split-TCP solution may violate the end-to-end security protection between the transmitter and receiver. In addition, TCP performance for the connection between the terminal and the proxy (which includes a radio link) may not be satisfactory.

Snooping TCP In the TCP-aware link layer approach, the losses due to channel errors are recovered locally, while maintaining the end-to-end semantics of TCP, by using a link layer that is aware of TCP semantics. In snooping TCP, the base station snoops TCP packets and retransmits it locally, when it receives three duplicate ACKs from the mobile host. Also, retransmission interference between the link layer and TCP is avoided by suppressing the duplicate ACKs from reaching the sender, if the lost packet can be recovered by link layer retransmission. Thus, it effectively reduces the number of end-to-end retransmissions and also prevents the sender from unnecessarily reducing the sending rate. However, since the link layer protocol needs to look into the TCP headers to identify a packet loss, it cannot be used when headers are encrypted. Also, this approach will not work in case of asymmetry in the network routes. Networks with higher bit-error rates, such as those with wireless links and mobile hosts, violate many of the assumptions made by TCP, causing degraded end-to-end performance. The snoop protocol improves TCP performance in wireless networks. The TCP-aware link layer schemes require the link layer at base station to cache the packets and perform local recovery in case of packet losses. Since there are no dedicated router/base station nodes in an ad-hoc network, the requirements of this scheme make it an unlikely solution in case of mobile ad-hoc networks.

Explicit-loss Notification Approach It relies on the identifying the type of loss and sending an indication to the TCP sender regarding the type of the loss. The TCP protocol at the sender can then be modified to respond differently to different types of losses. It is not quite possible to accurately identify the type of loss. Also, this requires several modifications in the core routers of the network and/or at both the sender and the receiver. The explicit-loss notification schemes could be used if the loss type can be identified. However, identifying the type of loss is a non-trivial problem, and so these schemes are not suited for implementation on the mobile ad-hoc networks.

M-TCP The end-to-end schemes modify the TCP protocol at the sender and/or receiver to improve its performance over wireless networks, with little or no help from the network elements, thus maintaining true end-to-end semantics. The improved performance in the presence of losses due to channel errors is achieved by modifying the way in which TCP controls the window and hence the sending rate. In this protocol, a rate estimator is used to estimate the fair rate for the connection by sampling and exponentially filtering the acknowledgements. This rate is used to guide the modifications to the window and hence sending rate.

The best performance improvements can be achieved by using link-layer protocols aware of TCP semantics. While the end-to-end protocols are not as effective as the local recovery schemes, they provide comparable results. Since the end-to-end schemes do not require support from the network infrastructure, they are more desirable for large-scale implementation on the Internet. End-to-end schemes seem to be the only possible solution, since they do not require any support from the network infrastructure. However, the mobile hosts are normally small hand-held devices and heavyweight protocols that require lots of computation and state maintenance may not be easily implemented on mobile hosts.

14.8 WIRELESS SENSOR NETWORKS (WSNS)

The technology required for Wireless Sensor Networks (WSNs) is located somewhere between IEEE 802.15.1 or IEEE 802.15.4 technology and the RFIDs. WSNs can be viewed as an extreme form of wireless ad-hoc networking with very low-power devices. Figure 14.18 presents an overview of a typical wireless sensor network.

The networks consist of several thousands or more tiny immobile sensors, also called *motes*, that are densely deployed in the service area on an ad-hoc basis to sense and transmit regularly some defined characteristics of the surrounding environment. An associated base station collects the information forwarded by the wireless sensors on a data-centric basis. WSNs have very limited computing capabilities, change their topology quite frequently as per the need of a particular application, and are prone to failures.

On the basis of mode of operation or functionality and the type of intended applications, wireless sensor networks can be broadly classified into two main categories:

Proactive WSNs The sensor nodes periodically switch on their transmitters, sense the parameter, and transmit the data to the network. In this way, proactive WSNs collect a snapshot of the relevant parameters at regular intervals of time and process the data for further use. These networks are well suited for applications requiring periodic monitoring of the data.

Reactive WSNs The sensor nodes react immediately to sudden and significant changes in the value of a sensed parameter. Therefore, these types of networks are well suited for applications which are time critical.

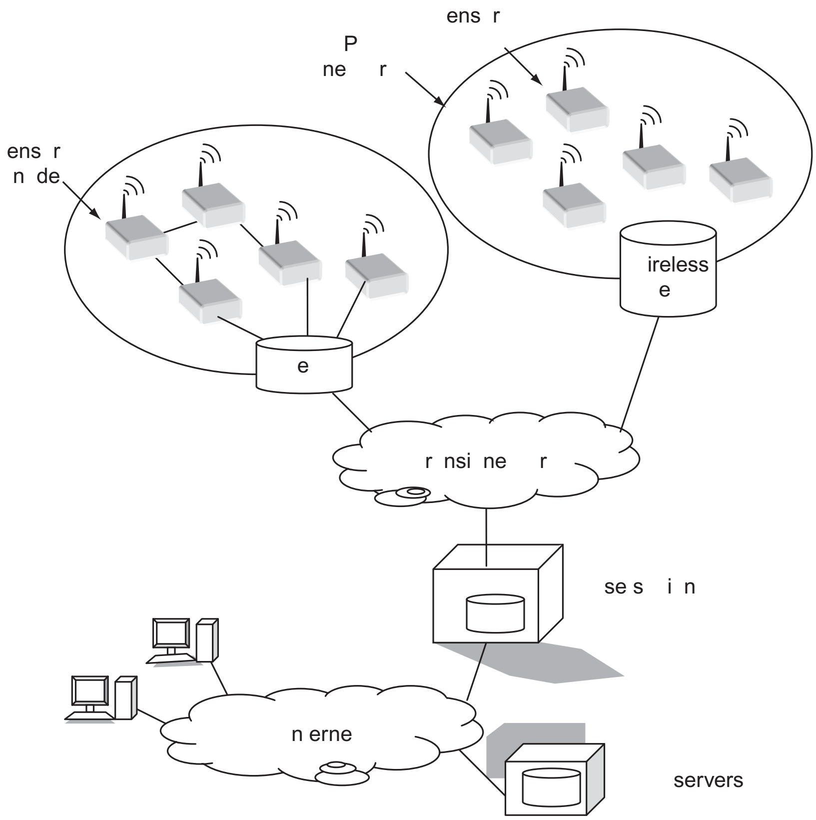

Fig. 14.18 *A typical wireless sensor network*

EXAMPLE 14.11 MANETs versus WSNs

Compare and contrast various features of MANETs versus WSNs.

Solution

Table 14.10 gives a comparative study of similarities and differences in some of the major features offered by MANETs versus WSNs.

Table 14.10 *MANETs versus WSNs*

S. No.	*eature/Parameter*	*MANETs*	*WSNs*
1.	Communication reliability	Unreliable	Unreliable
2.	Network configuration requirement	Self-configuration	Self-configuration
3.	Bandwidth requirement	Constrained	Very constrained
4.	Energy requirement	Constrained	Very constrained
5.	Node mobility	Typically mobile	Typically immobile
6.	Interaction among nodes	Competitive	Cooperative
7.	Nodes identification	Address centric	Data centric
8.	Performance metric	QoS: delay	Application specific QoS

14.8.1 Routing in Wireless Sensor Networks

Each application has its own problems like topology discovery, power-efficient routing, data recovery, and security. There are many challenging issues such as medium-access control mechanisms, management of the nodes, routing of data within the sensor network, fault tolerance, reliability, and low-power design which need to be resolved for optimum performance of the wireless sensor networks.

Designing a routing protocol is one of most challenging tasks in wireless sensor networks. Existing routing protocols in wireless sensor networks could be classified into two main categories: hierarchical clustering routing, and data centric routing. In *hierarchical clustering routing* protocols, the cluster formation is based generally on the sensor's proximity to the cluster head without taking in consideration the nature of sensed data which may lead to inefficient data aggregation and losing energy. In addition, the cluster formation does not take in consideration the position of the observed event. The intra-cluster routing between nodes inside the cluster uses flooding technique which is not energy efficient. The possibility of the mobility of the observed event is not covered in the existing hierarchical routing protocols, which is necessary in some applications such as tracking an object.

In *data-centric routing* protocols, despite the in-data processing between the selected sensor nodes like data aggregation and data fusion, these operations lack organisation between nodes. Data fusion is the process of aggregating similar data gathered by adjacent nodes in a certain region before sending it. Furthermore, the large numbers of messages that flood the sensor network in order to find the specific nodes and establishing the paths between them and the sink consume too much energy.

There is a need for a general middleware architecture in wireless sensor networks which could work equally well with or without fixed infrastructure support with minimum hardware and software requirements. The middleware architecture keeps the optimal performance of the system by dynamically distributing processing power between more capable nodes within the network. It not only prolongs the life of the network but also ensures that the data provided by the sensors will be available for a longer period of time.

With the advances in Micro Electro-Mechanical Systems (MEMS) technologies, wireless sensor networks have enabled the development of tiny battery-driven sensors for a wide range of application domains. They also need to provide self-organisation, environment sensing and distributed processing. One of the main challenges is how to coordinate mobile sinks effectively with large distributed sensors and transmit data over dynamic networks. The unpredictable delays are caused by network congestion, unreliable communication, power losses of sensor nodes, and data collision. This affects the execution of tasks dependent on these messages.

The other challenges include the need of distributed processing algorithms that are not centralised, to efficiently move large amounts of sensor data for processing in low bandwidth, large scale coordination as many independent sensor nodes need to act in a concert with one another, and real-time computation because new data is always coming which requires faster processing than it is generated.

14.8.2 WSNs Applications

Wireless sensor networks are a promising technology for applications ranging from environmental monitoring to industrial asset management. This type of networks are expected to change our life in many ways, in schools, hospitals, houses, and many other places. In health, for example, sensor networks may be used to monitor the patient health state, by deploying many small sensor nodes in different parts of his/her body. WSNs have a wide range of applications that can be categorised as indoor and outdoor applications. Some of the major application areas of WSNs are

Environment Weather prediction; climate monitoring; distributed computing; pollution tracking; seismic detection

Urban Transportation and traffic systems; automatic identification by driving license; parking availability; security monitoring in shopping malls, parking garages, city streets; home security

Industrial Inventory tracking; assembly lines; RFID tags

Medical Tele-monitoring of human physiological data, tracking and monitoring doctors and patients inside a hospital and insurance cards, aids for the visually handicapped

Military To track and monitor movements of enemy troop or terrorists

14.9 RFID TECHNOLOGY

Radio Frequency Identification (RFID) technology is similar to barcode labels but uses radio frequency waves instead of laser light to read the product code. The RFID stores product information in electronic tags that contain an antenna and a chip. RFIDs can respond to a radio signal and transmit their tag. They can store additional data, employ collision avoidance schemes, and comprise smart-card capabilities with simple processing power. The size of the memory in a tag varies between 16 bits and hundreds of kilobits.

RFID tags are initially programmed with a unique identification code obtained from EPCglobal standards. Electronic Product Code (EPC) is a standardised numbering scheme so that it can be identified electronically. EPC is either 64 or 96 bits long and is usually represented in hexadecimal notation. RFID tags are also commonly known as transponders—a combination of transmitter and responder and includes an integrated circuit that contains some non-volatile memory and a simple microprocessor which can store data that is transmitted in response to an interrogation from a reader.

There are two classes of tags—Class 0 tags are read-only, and Class 1 tags are read/write. Tags and readers use different transmission mechanisms in each frequency band of HF (13.56 MHz) and UHF (400 to 900 MHz). There are basically four types of RFID tags:

(a) *Passive tags* (most common type) which are small, use the electromagnetic energy in the RF waves and do not require battery power. These can be produced in large quantities at low cost. The amount of data stored in a typical passive RFID tag is relatively small. Data transmission rates for the tags are also low.
(b) *Active tags* are equipped with a battery, and thus have a limited life due to the battery. They can transmit the signal farther away, and beacons transmit on a periodic basis.
(c) *Semi-active tags* use a built-in battery to power the circuit only when a reader first energises the tag.
(d) *Sensory tags* can be equipped with various kinds of sensors to monitor and record information.

While RFIDs are not communication devices, they are essentially RF controllers without any MAC layer, and thus simply act as modems. Collisions have to be detected on higher layers. More and more computing power is available on small, embedded low-power RF systems. They offer transmission rates of up to 115 kbps (wireless extension of a serial interface) and operate on many different ISM bands such as 27 MHz, 315 MHz, 418 MHz, 426 MHz, 433 MHz, 868 MHz, and 915 MHz. The potential uses for RFID are practically unlimited.

Facts to Know!

Security for RFID systems is a complex issue. There is no single solution that addresses all possible situations. Applications of RFID are tremendous, and so RFID usage will continue to expand. Ultimately, RFID will be present in nearly every aspect of our lives.

A *reader* is a device that captures and processes the data received from the tags. A reader has separate transmitter and receiver circuits. HF tag readers can read 200 tags per second. For tags containing just an EPC, the actual rates will likely be between 500 and 800 tags per second. UHF specifications define the tag-to-reader data rate as twice that of the reader-to-tag. Tag-to-reader data rate can be up to 140.35 kbps.

Main applications of RFID devices include garage-door openers, wireless mice/keyboards, Automobile security and car locks, health care and pharmaceuticals, transportation, military, sports and entertainment, people monitoring and crowds, etc.

14.10 SECURITY REQUIREMENTS FOR WIRELESS NETWORKS

In the recent times there has been an explosion in demand for security measures motivated by the ever-increasing use of mobile and wireless networks and emergence of new services such as e-commerce. Intrusions in network security are defined as any malicious activities that could compromise the integrity, confidentiality, or availability of networks or information sources. Intrusion Detection Systems (IDSs) are kind of tools to detect and make alarms against such intrusions. Unlike the firewalls which are designed to prevent the occurrence of intrusions, IDSs only work after the intrusions have occurred or even succeeded. That is why IDSs are thought as the second line of defense. The main advantage of IDSs over firewalls is that IDSs can detect not only the attacks launched outside a network, but also inside attacks.

14.10.1 Security Threats in Wireless Networks

Traditional IDSs, which were originally developed for wired networks, are not suitable for wireless networks due to their unique characteristics and inherent vulnerabilities. This insufficiency directly compromises the effectiveness and efficiency of IDSs. For example, with the concept of wireless networks, it is believed that there must be lots of devices with tiny sizes in order to achieve unaware deployment. Inevitably they will have limited energy supply and storage space. An apparent issue is about how to implement an IDS in such an environment in a cost-effective way. Another key aspect is related to the system architecture. Existing monolithic IDSs are difficult to be applied directly because of the network nodes' capacity constrains. Meanwhile, the ad-hoc characteristic (flexibility in system architecture) of wireless networks makes hierarchical network-based IDSs face problems as well.

The security threats to wireless networks can be categorised as follows:

Accidental Attack Exposure due to frequent failure of components or some weakness in the system, such as software bug or hardware failure. The protection is to keep the possible damages as small as possible.

Passive Attack Here, the goal of the intruder is simply to monitor or obtain information that is being transmitted. Attacks may take the form of releasing message contents or traffic characteristics. Since no data is altered, passive attacks are difficult to detect.

Active Attacks In this type of attack, modification of data or false data transmission takes place, giving rise to masquerade or replay, to produce unexpected effects.

Vulnerability A weakness in some aspect or feature of a system that makes a threat possible.

Unauthorised Usage This attack takes place because of the growing use of the Internet, which leaves the network vulnerable to hackers, viruses and intruders. It can be prevented by using proper user-authentication techniques.

Broadcast Based As mentioned earlier, an eavesdropper is able to tap the communication into the wireless communication channels, by positioning itself within transmission range. The approaches to mitigate this problem involve use of directional antennas, low-power transmissions and frequency hopping and spread spectrum technologies at the physical layer and encryption techniques at the higher layers.

Device Vulnerability Mobile devices can be hijacked easily, and if secret IDs or codes are embedded in the device, hijackers may get access to private information stored on it, thereby accessing other resources of the network.

Heterogeneity Mobile nodes need to adjust to potentially different physical communication protocols as they move to different locations.

Limited Resources and Computational Ability To keep the mobile unit portable, that is, small and lightweight, compromises on CPU speed and battery power is made. This may result in the exclusion of techniques such as public key cryptography during normal operations to conserve power. Also, it leaves the unit open to resource depletion and exhaustion attacks, where the battery life is shortened considerably. In ad-hoc networks, such attacks can cause key routing nodes in the network to fail, leaving parts of it unreachable.

14.10.2 Security in WLAN, WPAN and WMAN

A lack of effective security standards has slowed down the business adoption of Wi-Fi networks. The combination of an essentially useless security protocol implemented on loose access points creates a huge potential security hole in any business infrastructure. While the unauthorised entry to conventional wired LANs can be blocked by deploying firewalls and taking other measures at specific locations, Wi-Fi networks offer access to anyone who can get physically close enough to the access point. Since Wi-Fi is the dominant wireless networking technology at the moment, with a high promise to provide seamless mobility among current subscribers as well as future potential users, authentication and data security issues, other than high-speed reliable data transmission service, remain the main concerns.

A common approach to provide security over Wi-Fi link is to bypass WEP (Wired Equivalent Privacy) and use the corporate Virtual Private Networks (VPN). VPNs manage data confidentiality by encrypting network traffic, but they don't always have authentication systems or access controls that work well in wireless environments, especially when the access point may be publicly accessible. If a VPN is not set up with strong mutual authentication on both ends, subscribers may be open to an attack by a hacker on the Wi-Fi networks, monitoring traffic to the access point, intercepts attempts to connect to the corporate VPN and manages to masquerade as user VPN server, may be long enough to steal logon credentials.

Compared to WEP in IEEE 802.11 WLAN standards, Bluetooth offers a lot more security. Bluetooth devices can transmit private data, for example, schedules between a mobile phone and a PDA. A user does not want anyone to eavesdrop the data transfer. Bluetooth offers mechanisms for authentication and encryption on the MAC layer. The Bluetooth includes a challenge-response routine for authentication, a stream cipher for encryption, and a session key generation. Each connection may require a one-way, two-way, or no authentication using the challenge-response routine, and for each transaction, a new random number is generated on the Bluetooth chip. Key management is not done at the MAC layer and it is carried out at higher layers.

The security algorithms use the public identity of a device, a secret private user key, and an internally generated random key as input parameters. To set up trust between the two devices for the first time, a user can enter a secret PIN into both devices. This PIN can have a length of 4 bytes 16 bytes. Based on the PIN, the device address, and random numbers, several keys can be computed which can be used as link key for authentication. Link keys are typically stored in a persistent storage. The authentication is a challenge-response process based on the link key, a random number generated by the device that requests authentication, and the address of the device that is authenticated.

Facts to Know!

The use of numerous security features including verifiable digital certificates, the most advanced encryption mechanisms (AES, 3-DES. and RSA), and secure key exchange protocols make WiMAX technology the most suitable for quick and world-wide adaptability.

An encryption key of a maximum size of 128 bits is generated based on the link key, values generated during the authentication, and a random number. Based on the encryption key, the device address, and the current clock, a payload key consisting of a stream of pseudo-random bits is generated for ciphering user data. The security features included in Bluetooth helps to set up a local domain of trust between devices. If Bluetooth devices are switched on, they can be detected unless they operate in the non-discoverable mode. The PINs are quite often fixed. Some of the keys are permanently stored on the devices and the quality of the random number generators has not been specified.

Security in WMANs is a major concern. The MAC layer in WMAN includes a privacy sublayer which provides a client/server authentication and key management protocol using digital certificates. Data-encryption algorithms include 3-DES, RSA with 1024-bit key, and AES with 128-bit key. Efficient bandwidth-saving protocols along with use of adaptive modulation in the physical layer, and QoS enable WiMAX reduce jitter and latency, and maintain a consistent bandwidth.

14.11 IEEE 802.21 STANDARD – AN OVERVIEW

The purpose of the IEEE 802.21 standard is to enhance user experience of mobile devices by supporting handovers between heterogeneous networks. IEEE 802.21 is an emerging wireless networking standard which supports algorithms enabling seamless handover between networks of the same type as well as handover between different network types. The 802.21 standard provides information to allow handing over to and from cellular (GSM, UMTS), IEEE 802.11, Wi-Fi, Bluetooth, and IEEE 802.16 wireless networks through different handover mechanisms. Digital cellular networks and IEEE 802.11 networks employ handover mechanisms for handover within the same network type, termed as horizontal handover. Mobile IP provides handover mechanisms for handover across subnets of different types of networks, but can be slow in the process. Current IEEE 802 standards do not support handover between different types of networks. They also do not provide triggers or other services to accelerate Mobile IP-based handovers. Moreover, existing IEEE 802 standards provide mechanisms for detecting and selecting network access points, but do not allow for detection and selection of network access points in a way that is independent of the network type.

The salient features of IEEE 802.21 standards include roaming between 3G cellular networks and IEEE 802.11 wireless networks, ad-hoc teleconferencing, applicable to both wired and wireless networks, compatibility and conformance with other IEEE 802 standards, adaptability by multiple vendors and users. The standard include definitions for managed objects that are compatible with management standards like SNMP. Although security algorithms and security protocols will not be defined in the standard, authentication, authorisation, and network detection and selection will be supported by the protocol.

Vertical hand-offs among a range of wired and wireless access technologies including WiMAX can be achieved using Media-Independent Handover (MIH) which is standardised as IEEE 802.21. MIH enables the handover of IP sessions from one layer to access technology to another layer, to achieve mobility of end-user devices. The importance of MIH derives from the fact that a diverse range of broadband wireless access technologies is available and in course of development, including GSM, UMTS, Cdma2000, Wi-Fi, WiMAX, Mobile-Fi and WP Multimode wireless devices that incorporate more than one of these wireless interfaces require the ability to switch among themselves during the course of an IP session, and devices such as laptops with Ethernet and wireless interfaces need to switch similarly between wired and wireless access.

Vertical hand-off refers to a network node changing the type of connectivity it uses to access a supporting infrastructure, usually to support node mobility. For example, a suitably equipped laptop might be able to use both a high-speed wireless LAN and a cellular technology for Internet access. Wireless LAN connections generally provide higher speeds, while cellular technologies generally provide more ubiquitous coverage. Thus, the laptop user might want to use a wireless LAN connection whenever one is available, and to switch over to a cellular connection when the wireless LAN is unavailable. Vertical hand-offs refer to the automatic switchover from one technology to another in order to maintain a communication link. This is different from

> **Facts to Know!**
>
> Vertical hand-offs between WLAN and UMTS (Cdma2000) have attracted telecom engineers and scientists in all the research areas of the 4G wireless network, due to the benefit of utilising the higher bandwidth and lower cost of WLAN as well as better mobility support and larger coverage of UMTS.

a horizontal hand-off between different wireless access points that use the same technology in that a vertical hand-off involves changing the data-link layer technology used to access the network.

When a session is handed off from one access point to another access point using the same technology, the handover can usually be performed within that wireless technology itself, even without involving IP or MIH functionality. For instance a VoIP call from a Wi-Fi handset to a Wi-Fi access point can be handed over to another Wi-Fi access point within the same corporate network using Wi-Fi standards such as IEEE 802.11f and IEEE 802.11r. However if the handover is from a Wi-Fi access point in a corporate network to a public Wi-Fi hotspot, then MIH is required, since the two access points cannot communicate with each other at the link layer, and are in general on different IP subnets. When a session is handed off from one wireless technology to another, MIH can provide the handover by passing messages among the wireless technologies and IP.

14.12 INTEROPERABILITY OF WIRELESS NETWORKS

The requirement of interoperability among emerging diverse wireless networks including 3G (UMTS, CDMA EV-DO) is of utmost importance. The popular emerging wireless networks include Wi-Fi, WiMAX, Mobile Ad-hoc Networks (MANETs), Wireless Mesh Networks (WMNs) and Wireless Sensor Networks (WSNs). Moreover, due to an explosive increase in wireless data applications, such as real-time multimedia services including VoIP and video streaming, there has been all-round development of numerous types of wireless systems, architectures and standards. The ability to employ standard protocols such as TCP/IP with IPv6 provision, is a key factor in the development of any new networking specifications.

However, the interworking of these diverse technologies as well as the efficient delivery of value-added applications and services over these emerging wireless systems lead to several challenging issues. Some of these issues include security, mobility management, architectures, QoS provisioning, and resource allocation. To solve these different issues related to heterogeneity of wireless access technologies and networks, researchers, engineers and technocrats have been putting various efforts and coming up with innovative ideas in term of novel protocols, architectures and algorithms.

For example, relatively wideband interference, such as that generated by IEEE 802.11b wireless networks appears like white noise to an IEEE 802.15.4 low-rate WPAN receiver because only a fraction of the 802.11b power lies within the 802.15.4 receiver bandwidth. To an IEEE 802.11b receiver, the signal from an 802.15.4 transmitter appears like narrowband interference. In addition, the low duty cycles typical of ZigBee devices further reduce the impact of interference. Similarly, the impact of interference from IEEE 802.15.1 Bluetooth devices should be minimal due to much smaller bandwidth of each frequency channel. And 802.15.4 devices should only interfere with approximately three out of the 79 frequency hops of a Bluetooth transmission, which is approximately 4% only.

> **Facts to Know!**
>
> UWB can interfere with 802.11a networks. ZigBee and WiMedia products should be able to coexist with 802.11b/g without any serious problems. Potential applications include connecting digital cameras to printers and kiosks or connecting laptops to multimedia projectors and sound systems.

Of all the issues with WPANs, spectrum conflict is potentially the most serious. Spectrum conflict, in general, is the potential for competing technologies using the same frequency bands to interfere with each other to the extent that their performance degrades significantly when used within close operating range of each other. For example, 802.11b/g WLANs perform poorly in environments where a

2.4 GHz cordless phone is in use nearby. This problem affects all DSSS and FHSS-based wireless networking technologies operating in the same frequency band. To avoid the technical support issues related to spectrum conflict, manufacturers of cordless phones have started introducing models that operate in another unlicensed ISM 5.8 GHz band. Likewise, applying UWB technology to all of the WPAN technologies may considerably reduce the problem of spectrum conflict.

One of the concerns of end users and service providers when considering wireless data transmission in the unlicensed bands is that as the number of simultaneous transmitters increases, interference also increases in the same proportion. Eventually, this can make the technology unusable. It is of particular importance considering the distances achievable with IEEE 802.16 WiMax technology. WiMax is different from 802.11 technologies in that it is not limited to the 2.4 GHz or the 5 GHz bands. The ISM band offers approximately 80 MHz bandwidth whereas the U-NII band offers about 300 MHz bandwidth and 12 channels which can be shared by users and service providers. Depending on the distance between transmitters, interference may not be a serious problem since WiMax signals are limited to approximately 48–56 km under ideal line-of-sight conditions. Moreover, with the use of adaptive modulations, variable data rates, and FEC, the concerns about interference can be easily resolved by testing the performance of a communication link. The deployment of smart antenna systems is also another viable solution to this type of problem.

Bluetooth can conflict with other technologies such as IEEE 802.11b and 802.11g WLANs that also use the ISM 2.4 GHz band for transmission. Using Bluetooth and 802.11b/g WLAN devices in close proximity to each other may cause the WLAN to drop the connection if it detects that another device is sharing its frequency. One simple way to resolve this problem is to move the Bluetooth device away from the 802.11b/g device. It is also recommended in IEEE 802.15.1 standard that Bluetooth and 802.11b/g WLAN devices share the spectrum by first checking to see if the air medium is clear for transmission. The 802.11a WLAN standard uses a different frequency band which helps to avoid the spectrum conflict all together.

DS-UWB can interfere with 802.11 network, whereas MB-OFDM-based UWB devices can easily avoid using the 5-MHz band that conflicts with 802.11a. Both WiMedia and ZigBee products may co-exist with 802.11b/g without any serious implications. However, the actual performance may vary when these products are deployed in large volume and in many different operating environments.

Requirements of next-generation or emerging wireless networks open even more new opportunities for many interesting and comprehensive research topics targeting at concepts, methodologies, and techniques to support advanced mobile value-added services. Clearly, the development of new mechanisms, protocols, algorithms, applications, architectures, and systems will have a significant impact for the successful deployment of emerging wireless networks.

Key Terms

- Access Point (AP)
- Active mode
- Active scanning
- Active tags
- Ad-hoc mode
- Advanced Encryption Standard (AES)
- Authentication
- Authorisation
- Bluetooth
- Broadband
- Care-of-address
- Data link layer
- Distributed Coordination Function (DCF)
- Distribution Service (DS)
- Encapsulation
- Extended Service Set (ESS)
- Foreign agent
- Foreign network
- HIPERLAN
- Home address
- Home agent
- Home network
- IEEE 802.11
- IEEE 802.11a/b/g/e/i
- IEEE 802.16
- Independent Basic Service Set (IBSS)

- Industrial, Scientific and Medical (ISM) band
- Infrared Data Association (IrDA)
- Infrastructure mode
- MANET
- Master device
- Media Access Control (MAC)
- Mobile IP
- Motes
- Neighbour piconets
- Park mode
- Parked devices
- Passive scanning
- Passive tags
- Peer-to-peer mode
- Piconet
- Piconet Coordinator (PNC)
- Point Coordination Function (PCF)
- Radio Frequency Identification (RFID)
- Scatternet
- Service Set Identifier (SSID)
- TCP
- Tunneling
- Ultra Wide Band (UWB)
- Unlicensed National Information Infrastructure (UNII)
- Voice over Internet Protocol (VoIP)
- Wireless Bridge
- Wireless Fidelity (WiFi)
- Wireless Local Area Network (WLAN)
- Wireless Metropolitan Area Network (WMAN)
- Wireless NIC
- Wireless Personal Area Network (WPAN)
- Worldwide Interoperability for Microwave Access (WiMAX)

Summary

The lack of a uniform set of standards, the inconsistency in the quality of service, and the diversity in the networking approaches make it difficult for a mobile computing environment to provide seamless mobility across different wireless networks. With the advancement in mobile information technologies like ultra high-speed transmission, wireless Internet protocol IPv6, user-controllable software defined radios, the potential mobile users would be able to use a cellphone or laptop or any other PDA as mobile communication terminal, choose freely the services, applications and service-providing networks, exploit advanced mobile e-commerce applications with higher levels of data security and integrity during business transactions, and access the Internet as they do in the office—anywhere, anytime while on move. This chapter has presented an overview of current and emerging wireless network technologies. IEEE 802.11 WLAN family of standards offers an alternative to existing LANs. The key advantages of the wireless LAN are that it eliminates the laying of cables and wiring cost, and that it accommodates mobile PC workstations. Wi-Fi is the most popular WLAN technology. Bluetooth is one of the fastest growing technology standards for use within personal area. The specifications and technology is defined by IEEE 802.15 Wireless PAN Standards. Compared to the WLAN technologies, Bluetooth technology aims at ad-hoc piconets which are local area networks with a very limited coverage and without the need for an infrastructure. Wireless MAN technology such as WiMAX is used to provide services in a very large area. Mobile ad-hoc networks and wireless sensor networks offer a variety of applications. A MANET is an autonomous system of mobile nodes connected by wireless links, with a capability of organising themselves dynamically. WSNs are a special class of wireless ad-hoc networks comprising of unmanned tiny immobile wireless devices with the purpose of gathering useful information. Finally, types of security threats and security requirements in wireless networks are discussed in this chapter.

Short-Answer Type Questions with Answers

A14.1 List the essential requirements of IEEE 802.11 standards.

IEEE 802.11 standards should meet the following requirements:

1) There should be one common MAC layer to support multiple physical layers.
2) There should be mechanisms to allow multiple overlapping networks in the same area.

3) Provisions should be provided to counter the effect of interference from other ISM band radios.
4) There should be appropriate protocols and algorithm mechanisms to allow multiple overlapping networks in the same area.

A14.2 Why are ad-hoc networks useful in day-to-day applications
Ad-hoc networks do not depend on pre-existing infrastructure. They are easy to install and commission the services. These networks can be deployed faster, and offer Anytime-Anywhere-Any device (AAA) network paradigm. These features make ad-hoc networks useful in day-to-day applications.

A14.3 What is meant by radio always-on' mode operation of WLAN interface cards
The WLAN interface cards could be operated in continuous aware mode which means its transceiver is always on. The cards can also operate in power-saving polling mode by keeping the radio in sleep state to save more power in which the wireless access point keeps data in its buffer for the wireless terminals and sends a signal to wake up the wireless terminals.

A14.4 Define wireless Access Point (AP).
A wireless access point in WLAN is an entity which provides associated wireless terminals an access to the distribution system via the wireless medium. The AP also serves as the WLAN hub that functions as a bridge and relay point between wireless terminals and the existing wired LAN.

A14.5 Distinguish between BSS and ESS.
Base Service Set (BSS) is a group of wireless terminals controlled by a single coordination function provided by an access point. Extended Service Set (ESS) is a group of one or more interconnected BSSs and integrated IEEE 802 wired LANs that appear as a single BSS to the LLC layer at any wireless terminal associated with one of these BSSs.

A14.6 What is the difference between an access point and a portal
An access point is a part of IEEE 802.11 wireless LAN which enables to connect a BSS with a backbone DS. The portal is a part of IEEE 802 wired LAN which is implemented in a device such as a bridge or router, and attached to the Distribution System (DS) for providing an interface with wireless LAN and wired LAN.

A14.7 Why are multiple access points installed in a building
Multiple access points are typically installed in order to provide seamless connectivity to wireless terminals as they move from one location to another. Similar to the cellular structure, several in the building installed in the building at various strategic locations can serve a larger area. There may be overlapped access areas. The access points track the movement of wireless terminals within a coverage area and decide on whether to allow them to communicate among themselves via one of the closeby access points.

A14.8 What are the advantages of IEEE 802.11 FHSS WLAN system
The IEEE 802.11 FHSS system operates at 1 and 2 Mbps with a transmission bandwidth of 1 MHz, with a frequency hop rate of 2.5 hops per second. In a fade, a packet of duration of up to 20 ms has a high probability of success in a few attempts. In a deep and long persisting frequency selective fading, the maximum delay jitter is around 400 ms. The IEEE 802.11 FHSS system provides a low-power simple implementation as compared with DSSS alternative capable of taking advantage of time diversity.

A14.9 Suppose the mobile phone units are equipped with Bluetooth devices. What impact will it have on the piconet
When the mobile units connected with Bluetooth devices move out of the range of a particular master device in the piconet, the wireless communication link between master–slave combination may break due to out-of-range conditions. If the master device itself moves out of the radio coverage area, the functioning of a piconet may fail completely.

A14.10 Can different routing protocols applicable to ad-hoc networks be applied for a scatternet
Yes, all the ad-hoc network routing protocols can be applied to scatternet provided every piconet is assumed to be a node. Since there may be only one bridge node between two overlapping piconets

and not all the nodes of a piconet have information about other piconets, the information is required to be distributed in the piconet using master–slave communication.

A14.11 How do ad-hoc networks differ from cellular networks
In cellular networks, the wireless communication between two mobile phone users takes place through fixed base stations serving as access points and wired backbone infrastructure. The cellular network architecture does not change frequently. In an ad-hoc wireless network, there is no such fixed infrastructure and communication between two mobile nodes is established directly. Since the mobile nodes are free to move in an arbitrary direction with a random speed, the ad-hoc network may change its topology dynamically in an unpredictable manner.

A14.12 Is it desirable to assign priority to traffic in a MANET
Yes, it is definitely desirable to prioritise traffic in a MANET. The control messages such as route request or route repair should be assigned a higher priority over data transfer. If it is discovered that a particular route has been broken then the file transfer on a route can be deferred.

A14.13 Mention few similarities and dissimilarities between mobile ad-hoc networks and wireless sensor networks.
Mobile ad-hoc networks and wireless sensor networks have similar characteristics of bandwidth-constrained and energy-constrained operation. Mobile ad-hoc networks have dynamic network topologies because mobile nodes are free to move. A wireless sensor network is a collection of tiny immobile disposable and low power devices. Moreover, wireless sensor networks are data-centric.

A14.14 What are specific challenges in wireless sensor networks
The specific challenges in wireless sensor networks include distributed processing which need algorithms that are not centralised and do not require all of the data, have low bandwidth communication, efficiently move large amounts of sensor data for processing, large scale coordination, many independent sensor nodes need to act in concert with one another, and the need of real-time faster data computation than data generation.

A14.15 Is RFID better than using bar codes
RFID is not necessarily better than bar codes. The two are different technologies and have different applications, which sometimes overlap. The big difference between the two is that bar code is a line-of-sight technology, whereas RFID doesn't require line of sight. RFID tags can be read as long as they are within range of a reader. Standard bar codes identify only the manufacturer and product, not the unique item.

A14.16 How much information can an RFID tag store
It depends on the vendor and the application, but typically a tag carries no more than 2 kB of data, enough to store some basic information about the item it is on. The simple tags are cheaper to manufacture and are more useful for applications where the tag will be disposed of with the product packaging.

A14.17 What's the difference between passive and active tags
Active RFID tags have a transmitter and their own power source (typically, a battery). The power source is used to run the microchip's circuitry and to broadcast a signal to a reader. Passive tags have no battery. Instead, they draw power from the reader, which sends out electromagnetic waves that induce a current in the tag's antenna. Semi-passive tags use a battery to run the chip's circuitry, but communicate by drawing power from the reader.

A14.18 Suggest some measures which can be taken to address the security issues in wireless networks.
First of all, proper authentication of the users needs to be done in order to avoid unauthorised usage. One way could be to assign passwords to each user so that each time the user queries the network in its wireless device, the network will ask the user to enter the password. Moreover, public/private key cryptography could be used to achieve increased security against intruder.

Self-Test Quiz

S14.1 The ______ are installed as an add-on unit with the wireless terminals to provide wireless communications.
(a) access points
(b) wireless access interface cards
(c) distribution systems
(d) BSSs

S14.2 The ______ may be a logic within a wireless terminal that provides access to the DS.
(a) access point (b) BSS
(c) ESS (d) transceiver

S14.3 At any given time, a wireless terminal can be associated with
(a) up to seven access points
(b) three access points
(c) two access points
(d) one access point only

S14.4 The IEEE 802.15.1 Bluetooth system has a typical frequency-hop rate of ______ hops per second.
(a) 2.5 (b) 1600
(c) 3200 (d) one access point only

S14.5 The data transfer time for a fixed size file ______ in the IEEE 802.11 WLAN system operating at 2 Mbps as compared to that of at 1 Mbps.
(a) increases by ten times
(b) increases by two times
(c) decreases by two times
(d) decreases by ten times

S14.6 The size of a file transferred in 8 seconds in the IEEE 802.11 WLAN system operating at 2 Mbps data transmission rate is
(a) 2 MB (b) 4 MB
(c) 16 MB (d) 32 MB

S14.7 In practical WLAN installation, the radio coverage area of the AP is usually ______ that of the wireless terminal in the same operating environment.
(a) larger than (b) identical to
(c) smaller than (d) independent to

S14.8 In IEEE 802.11 WLAN standard, the physical layer specifies 2 Mbps data rate. If the baseband modulation used is DSSS then the carrier modulation scheme is
(a) DBPSK (b) DQPSK
(c) 2-GFSK (d) 4-GFSK

S14.9 The Wi-Fi technology is specified in
(a) IEEE 802.11 WLAN standards
(b) IEEE 802.11a WLAN standards
(c) IEEE 802.11b WLAN standards
(d) IEEE 802.11g WLAN standards

S14.10 Bluetooth technology has been adopted as the IEEE ______ standards.
(a) 802.11b (b) 802.15.1
(c) 802.15.3 (d) 802.16a

S14.11 In a piconet, each of the slave devices has an assigned ______ address.
(a) 1-bit (b) 3-bit
(c) 7-bit (d) 8-bit

S14.12 The IEEE 802.15.3 standard, also called WiMedia, physical layer operates in the unlicensed ______ frequency band.
(a) 2.402–2.480 GHz (b) 915 MHz
(c) 868 MHz (d) 5 GHz

S14.13 The IEEE 802.15.1 WPAN standard uses ______ technique for separation of piconets.
(a) DSSS (b) OFDM
(c) FHSS-TDMA (d) FHSS-CDMA

S14.14 The WiMAX technology uses multicarrier OFDMA scheme in 2 GHz–11 GHz band, to achieve transmission data rates of
(a) 11 Mbps (b) 54 Mbps
(c) 155 Mbps (d) 2 Gbps

S14.15 Assuming each mobile node to be connected to exactly four adjacent mobile nodes in a MANET of 100 nodes, the total number of wireless links are
(a) 100 (b) 200
(c) 400 (d) 800

S14.16 Agent discovery protocols entails
(a) first listening to foreign agent advertisement for COAs and if not found then agent solicitation at defined intervals
(b) registering with a foreign agent and waiting for registration reply from the home agent
(c) first agent solicitation and then if COA is not found, listening to foreign agent advertisement
(d) requesting foreign agent advertisements

S14.17 During tunneling from the HA to the FA, minimum encapsulation combines encapsulation header and IP header words into
(a) five words
(b) six words and checksum of the header
(c) six words and checksum of the packet through the tunnel
(d) seven or eight words

S14.18 Quality of service in a mobile network is affected by
(a) transmission errors and the quality of audio received
(b) network connectivity, effective bandwidth availability, connection reliability, and data loss probability
(c) atmospheric conditions and the number of simultaneous active mobile users
(d) network load and the data lost per second

Answers to Self-Test Quiz

S14.1 (b); S14.2 (a); S14.3 (d); S14.4 (b); S14.5 (c); S14.6 (a); S14.7 (a); S14.8 (b); S14.9 (c); S14.10 (a); S14.11 (b); S14.12 (a); S14.13 (d); S14.14 (c); S14.15 (b); S14.16 (a); S14.17 (d); S14.18 (b)

Review Questions

Q14.1 Compare the usefulness of WLANs, WPANs, and WMANs.

Q14.2 What is the fundamental difference between Wi-Fi and Bluetooth technologies?

Q14.3 How is it possible for two adjacent piconets not to use the same frequency-hopping sequences?

Q14.4 What is the difficulty of implementing CSMA/CD technique in a wireless environment?

Q14.5 List any three advantages of an infrastructure topology over an ad-hoc topology. Compare peer-to-peer and multihop ad-hoc network topologies.

Q14.6 What are the five major challenges for implementation of wireless LANs?

Q14.7 Give the physical specification summary of the DSSS and FHSS used by the IEEE 802.11.

Q14.8 Name the four states that a Bluetooth device can have.

Q14.9 What is Wireless broadband WiMAX? How is it different from Wireless LAN Wi-Fi?

Q14.10 How does IEEE 802.1x overcome the security vulnerabilities of WEP?

Q14.11 Explain, with diagram, how a correspondent mobile node on a visit sends and receives IP packets to and from another MN also on a visit at another foreign network.

Q14.12 What are the functions of home and foreign agents in the mobile IP protocol?

Q14.13 How does agent advertisement differ from agent solicitation? When and how does a mobile node solicit an agent?

Q14.14 What is the difference between the care-of-address and co-located care-of-address?

Q14.15 Explain mobile TCP. How does a supervisory host send TCP packets to the mobile node and to a fixed TCP connection?

Q14.16 What information is stored on RFID tags?

Analytical Problems

P14.1 The IEEE 802.11 WLAN system operates at 1 Mbps. Calculate the data transfer time of a 20-kB file.

P14.2 The IEEE 802.11 WLAN system operates at 2 Mbps data transmission rate. Calculate the size of the data file that can be transferred in 32 seconds.

P14.3 In an IEEE 802.11a WLAN system, the OFDM baseband modulation scheme uses 48 data subcarriers. OFDM symbol duration including guard interval for ISI mitigation is given as 4 μs. If the system uses FEC code rate and 16-QAM carrier modulation scheme then determine the achievable transmission data rate.

P14.4 A given MANET consists of 200 mobile nodes. The mobility of the nodes is such that four existing wireless links are broken, while four new wireless links are established every one second. Assume that each mobile node is connected to exactly four adjacent mobile nodes. Find the total number of wireless links at any time in the network.

P14.5 In a MANET comprising of 100 mobile nodes, the mobility of the nodes is such that two existing wireless links are broken, while two new wireless links are established every one second. If the updated message is sent every 5 seconds, find the number of messages initiated periodically. Assume table-driven routing protocol.

P14.6 Consider a 2.4-GHz WLAN transmitter, radiating +20 dBm power from an isotropic antenna, is located at a workstation in an office building. It is separated from the network access node (receiver) from a distance of 35 m. The wireless signal crosses a 5-m open space of the office itself, then through a plasterboard wall (which causes 6-dB signal attenuation), and finally a large open area with a path-loss exponent of 3.1. Assuming 0-dB receiver antenna gain, establish that the received signal will be well above the receiver sensitivity level of −75 dBm.

P14.7 Find out the transmission range of a wireless node operating at 2.4 GHz with transmitter power of +7 dBm and receiver sensitivity of −81 dBm. Assume free-space propagation.

P14.8 How much reduction in transmission range is obtained for the scenario described in P14.7, if the two-ray propagation model is used?

P14.9 Compare the transmission range in the following cases:

Case I - Tx Power: +7 dBm; Rx sensitivity: −81 dBm

Case II - Tx Power: +15 dBm; Rx sensitivity: −91 dBm

Assume the wireless node operates at 2.4 GHz in mobile environment having propagation path-loss exponent as 4.

P14.10 A wireless station transmits data at the rate of 19.2 ksymbols/s using DSSS modulation technique. The chipping rate is 1.2288 Mcps. Calculate the bit interval, chipping frequency interval, DS spread factor, and bandwidth of the DS spread spectrum signal.

P14.11 A Bluetooth node has a transmit power of 0 dBm and Rx sensitivity of −80 dBm. The RF propagation path has a moderate clutter (corresponding to propagation path-loss exponent of 3). If it is communicating with another Bluetooth node with the same radio parameters, what is the estimated range? Ignore other losses.

P14.12 A Wi-Fi access point transmits with +20 dBm power. The receiver sensitivity of a wireless device in the network is specified as −85 dBm. What is the maximum allowable path loss, ignoring other losses?

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Appendix 1
Abbreviations and Acronyms

1G	First-Generation Analog Cellular Systems
2G	Second-Generation Digital Cellular Systems
2.5G	Intermediate (2G – 3G) Generation Digital Cellular Systems
3G	Third-Generation Digital Cellular Systems
3GPP	Third-Generation Partnership Project (3GPP – UMTS/WCDMA)
3GPP2	Third-Generation Partnership Project Group 2 (3GPP2 – Cdma2000)
4G	Fourth-Generation Digital Cellular Systems
802.11	Original IEEE standard for WLAN at 2.4 GHz (FHSS or DSSS)
802.11a	IEEE standard for WLAN at 5.2 GHz (OFDM)
802.11b	IEEE standard for High Rate WLAN at 2.4 GHz (DSSS)
802.11g	IEEE standard combining 802.11a and 802.11b
802.15	IEEE standards for Wireless PANs (WPAN)
802.16	IEEE standards for Wireless MANs (WMAN)
8-PSK	8-level Phase Shift Keying
AAA	Authentication, Authorisation, and Accounting
ACELP	Algebraic Code Excited Linear Prediction
ACI	Adjacent Channel Interference
ACK	Acknowledgement
ADPCM	Adaptive Differential Pulse Code Modulation
ADSL	Asymmetrical Digital Subscriber line
AES	Advanced Encryption Standard
AGCH	Access Grant Channel
AKA	Authentication and Key Agreement
AM	Amplitude Modulation
AMPS	Advanced Mobile Phone System
AMR	A hybrid speech coder using ACELP
AODV	Ad-hoc On-demand Distance Vector routing protocol
AP	Access Point
APC	Adaptive Predictive Coding
ARFCN	Absolute Radio Frequency Channel Numbers
ARQ	Automatic Retransmission Request
AS	Authentication Server
ASK	Amplitude Shift Keying
ATM	Asynchronous Transfer Mode
AuC	Authentication Centre
AV	Authentication Vector
AWGN	Additive White Gaussian Noise
BCCH	Broadcast Control CHannel
BCH	Broadcast Channel
BER	Bit-Error Rate
BFSK	Binary Frequency Shift Keying
BG	Border Gateway
BPSK	Binary Phase Shift Keying
BRAN	Broadband Radio Access Networks
BS	Base Station
BSC	Base Station Controller
BSIC	Base Station Identity Code
BSS	Basic Service Set (used in WLAN)
BSS	Base Station Subsystem (used in cellular networks)
BSSID	Basic Service Set Identifier
BTS	Base Transceiver Station
CAMEL	Customised Application for Mobile network Enhanced Logic
CAP	CAMEL Application Part
CCCH	Common Control CHannel
CCH	Control Channel
CCI	Co Channel Interference
CCK	Complementary Code Keying
CCITT	Consultative Committee on International Telegraphy and Telephony
CDMA	Code Division Multiple Access
CDMA 1xEV-DO	cdmaOne EVolution for high speed Data Only
CDMA 1xEV-DV	cdmaOne EVolution for high speed Data and Voice
Cdma2000	Code Division Multiple Access 2000
CDVCC	Coded Digital Verification Colour Code
CELP	Code Excited Linear Predictive
C/I	Carrier-to-Interference ratio
CM	Call Management
CN	Core Network, Correspondent Node

COA Care Of Address
codec coder/decoder
COST Cooperative for Scientific and Technical research
CPCH Common Packet CHannel
CPE Customer Premises Equipment
CPFSK Continuous Phase Frequency Shift Keying
CRC Cyclic Redundancy Check
CS Circuit Switched
CSMA Carrier Sense Multiple Access
CSMA/CA Carrier Sense Multiple Access Collision Avoidance
CSMA/CD Carrier Sense Multiple Access Collision Detection
CT2 Codeless Telephone-2
CTCH Common Traffic Channel
CTCH Common Transport CHannel
CVSDM Continuous Variable Slope Delta Modulation
DA Destination Address
DAB Digital Audio Broadcasting
DAMPS Digital Advanced Mobile Phone System
dB decibel
dBd dB gain with respect to a half wave dipole
dBi dB gain with respect to an isotropic antenna
DBPSK Differential Binary Phase Shift Keying
DCA Dynamic Channel Allocation
DCCH Digital Control CHannel
DCF Distributed Coordination Function
DCH Dedicated channel
DCS1800 Digital Cellular System (1800 MHz)
DECT Digital Enhanced Cordless Telephone
DES Data Encryption Standard
DFE Decision Feedback Equaliser
DFWMAC Distributed Foundation Wireless MAC (see MAC also)
DHCP Dynamic Host Configuration Protocol
DLC Data Link Control
DoS Denial-of-Service
DPSK Differential Phase Shift Keying
DQPSK Differential Quadrature Phase Shift Keying
DS Distribution Service
DSAT Digital Supervisory Audio Tone
DS-CDMA Direct Sequence CDMA (see CDMA also)
DSCH Downlink Shared CHannel
DSDV Destination Sequenced Distance Vector algrotithm
DS/FHMA Direct Sequence/Frequency Hopped Multiple Access
DSL Digital Subscriber Line
DSSS Direct Sequence Spread Spectrum
DST Digital Signaling Tone
DTCH Dedicated Traffic (Transport) CHannel
DVB Digital Video Broadcasting
E-DCH Enhanced Dedicated CHannel
EDGE Enhanced Data Rates for GSM Evolution (see GSM also)
EGPRS Extended GPRS (see GPRS also)
EIA Electronic Industries Alliance (Association)
EIR Equipment Identity Register
EIRP Effective Isotropic Radiated Power
E_b/N_0 Bit Energy-to-Noise power density
EPC Electronic Product Code
ERMES European Radio MEssage System
ERP Effective Radiated Power
ERP-PBCC Extended Rate Physical PBCC (see PBCC also)
ESN Equipment Serial Number
ESS Extended Service Set
ESSID Extended Service Set Identifier
ETACS European or Enhanced TACS (see TACS also)
ETSI European Telecommunications Standards Institute
EY-NPMA Elimination Yield Non-Preemptive Multiple Access
FA Foreign Agent
FACCH Fast Associated Control CHannel
FACH Forward Access Channel
FAF Floor Attenuation Factor
FAUSCH FAst Uplink Signaling Channel
FBF Feedback Filter
FCC Federal Communications Commission
FCCH Frequency Correction Channel
FCDMA Hybrid FDMA/CDMA (see FDMA and CDMA also)
FDD Frequency Division Duplexing
FDMA Frequency Division Multiple Access
FEC Forward Error Correction
FH-CDMA Frequency Hopping Code Division Multiple Access
FHMA Frequency Hopped Multiple Access
FHSS Frequency Hopping Spread Spectrum
FLEX 4-level FSK based paging standard
FM Frequency Modulation
FN Frame Number
FSE Fractionally Spaced Equaliser
FSK Frequency Shift Keying

FVC	Forward Voice Channel
GERAN	GSM/EDGE Radio Access Network (see GSM and EDGE also)
GFSK	Gaussian Frequency Shift Keying
GGSN	Gateway GPRS Support Node (see GPRS also)
GMSC	Gateway MSC (see MSC also)
GMSK	Gaussian Minimum Shift Keying
GOS	Grade of Service
GPRS	General Packet Radio Service
GPRS-MS	GPRS Mobile Subscriber (see GPRS also)
GPS	Global Positioning System
GSM	Global System for Mobile Communications
GTP	GPRS Tunneling Protocol (see GPRS also)
HA	Home Agent
HCF	Hybrid Coordination Function
HDLC	High-level Data Link Control
HiperLAN	High Performance Wireless LAN - Versions 1 and 2 (European Standard)
HiperLAN/2	High Performance Radio Local Area Network 2 (ETSI/BRAN standard at 5.2 GHz using OFDM) (see OFDM also)
HLR	Home Location Register
HR-WPAN	High Rate Wireless Personal Area Network
HSCSD	High-Speed Circuit-Switched Data
HSUPA	High-Speed Uplink Packet Access
IBSS	Independent Basic Service Set
ICMP	Internal Control Message Protocol
ID	Identifier
IDS	Intrusion Detection System
IEEE	Institute of Electrical and Electronic Engineers
IETF	Internet Engineering Task Force
IM	Inter-Modulation
IMEI	International Mobile Equipment Identity
IMS	IP-based Multimedia Services
IMSI	International Mobile Subscriber Identity
IMT-2000	International Mobile Telecommunications for year 2000 and beyond
IMTS	Improved Mobile Telephone Service
IP	Internet Protocol
IPTV	Internet Protocol Television
IPv4	Internet Protocol version 4
IPv6	Internet Protocol version 6
IrDA	Infra-red Data Association
IS-136	Interim Standard–136 for USDC with digital control channels (see USDC also)
IS-54	Interim Standard–54 for USDC with analog control channel (see USDC also)
IS-95	Interim Standard–95 for US CDMA (see CDMA also)
ISDN	Integrated Services Digital Network
ISI	Inter-Symbol Interference
ISM	Industrial, Scientific and Medical
ISP	Internet Service Provider
ISUP	ISDN Signaling User Part (signaling) (see ISDN also)
ITU	International Telecommunications Union
Iu	UMTS Air Interface Standard (see UMTS also)
IWU	InterWorking Unit
JDC	Japanese Digital Cellular
JTACS	Japanese Total Access Communication System
LA	Location Area
LAI	Location Area Identification
LAN	Local Area Network
LAPD	Link Access Protocol D-channel
LCC	Lost Call Cleared
LCD	Lost Call Delayed
LCR	Level Crossing Rate
LLC	Logical Link Control
LMDS	Local Multipoint Distribution Service
LMS	Least Means Square
LOS	Line-of-sight
LPC	Linear Predictive Coding
LR-WPAN	Low Rate-Wireless Personal Area Network
LTE	Linear Transversal Equaliser; Long Term Equipment
LTP	Long Term Prediction
MAC	Media Access Control
MAHO	Mobile Assisted HandOff
MAI	Multiple Access Interference
MAN	Metropolitan Area Network
MANET	Mobile Ad-hoc NETwork
MAP	Mobile Application Protocol (GSM)
M-ary	Multiple level modulation
MBMS	Multimedia Broadcast and Multicast Services
MC	Muti-Carrier
MCC	Mobile Country Code
Mcps	Mega chips per second
MCS	Multiple Modulation and Coding Schemes
MFSK	Minimum Frequency Shift Keying

MH	Mobile Host
MIMO	Multiple In Multiple Out
MIN	Mobile Identification Number
MIP	Mobile IP (see IP also)
MLSE	Maximum Likelihood Sequence Estimation
MM	Mobility Management
MMAC	Multimedia Mobile Access Communication Systems (Japan)
MMDS	Multichannel Multipoint Distribution Service
MMS	Multimedia Messaging Service
MN	Mobile Node
MNC	Mobile Network Code
MOS	Mean Opinion Score
MPE	Multi-Pulse Excited
MPSK	Minimum Phase Shift Keying
MS	Mobile Station or Mobile Subscriber
MSB	Most Significant Bits
MSC	Mobile Switching Center
MSE	Mean Square Error
m-sequence	Maximum length sequence
MSIN	Mobile Subscriber Identity Number
MSISDN	Mobile Subscriber ISDN number
MSK	Minimum Shift Keying
MSRN	Mobile Station Roaming Number
Msps	Mega symbols per second
M-TCP	Mobile TCP (see TCP also)
MTP	Message Transfer Part
MTSO	Mobile Telecommunications Switching Office
MUX	Multiplexer
NADC	North-American Digital Cellular
NAMPS	Narrow Band Advanced Mobile Phone System
NB	Node-B
NIU	Network Interface Unit
NLOS	Non-Line-of-Sight
NMT	Nordic Mobile Telephone
NPMA	Non Preemptive Multiple Access
NSS	Network and Switching Subsystem
NTT	Nippon Telephone and Telegraph
OCCCH	ODMA Common Control CHannel
ODCH	ODMA Dedicated CHannel
ODCCH	ODMA Dedicated Control CHannel
ODMA	Opportunity Driven Multiple Access
ODTCH	ODMA Dedicated Traffic CHannel
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency Division Multiple Access
OLSR	Optimized Link State Routing Protocol
OMAP	Operations Maintenance and Administration Part
OMC	Operation Maintenance Center
OQPSK	Offset Quadrature Phase Shift Keying
ORACH	ODMA Random Access CHannel
OSS	Operational Support System
OVSF	Orthogonal Variable Spreading Factor codes
PABX	Private Automatic Branch Exchange
PACS	Personal Access Communication System
PBCC	Packet Binary Convolution Coding
PBCCH	Packet Broadcast Control CHannel
PC	Personal Computer
PCCCH	Packet Common Control CHannel
PCCH	Paging Control CHannel
PCF	Point Coordination Function
PCH	Paging CHannel
PCN	Personal Communication Network
PCS	Personal Communications Service
PDA	Personal Digital Assistant
PDC	Personal Digital Cellular or Pacific Digital Cellular
PDSN	Packet Data Serving Node
PDU	Packet Data Unit
PHS	Personal Handyphone System
PHY	Physical
PID	Protocol Identifier
PIN	Personal Identification Number
PLMN	Public Land Mobile Network
PM	Phase Modulation
PN	PseudoNoise
PNC	Piconet Coordinator
POCSAG	Post Office Code Standard Advisory Group
POTS	Plain Old Telephony Service
PPM	Pulse Position Modulation
PPP	Point to Point Protocol
PR	Packet Radio
PRMA	Packet Reservation Multiple Access
PS	Packet Switched
PSK	Phase Shift Keying
PSTN	Public Switched Telephone Network
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QoS-GRD	QoS Guided Route Discovery
QPP	Quadrature Permutation Polynomial
QPSK	Quadrature Phase Shift Keying
RAC	Routing Area Code
RACH	Random Access CHannel
RAN	Radio Access Network
RANAP	Radio Access Network Application Port
RC4	A variable key-size stream cipher with byte-oriented operations

RCC	Reverse Control Channel
RELP	Residual Excited Linear Predictor
RF	Radio Frequency
RFID	Radio Frequency Identification
RLC	Radio Link Control
RLP	Radio Link Protocol
RLS	Recursive Least Square
RNAS	RAN Access Server (see RAN also)
RNC	Radio Network Controller
RNS	Radio Network Subsystem
RNSAP	Radio Network Subsystem Application Port
RPE-LTP	Regular Pulse Excited Long Term Prediction
RRC	Radio Resource Control
RRM	Radio Resource Management
RS	Reed Solomon
RSA	Rivest, Shamir and Adleman
RSSI	Received Signal Strength Indication
RTS/CTS	Request to Send/Clear to Send
RTT	Radio Transmission Technology
RVC	Reverse Voice Channel
Rx	Receiver
SAC	Service Area Code
SACCH	Slow Associated Control CHannel
SAI	Service Area Identifier
SAP	Service Access Point
SAT	Supervisory Audio Tone
SCCP	Signaling Connection Control Port
SCH	Synchronization CHannel
SCM	Station Class Mark
SDCCH	Standalone Dedicated Control CHannel
SDMA	Space Division Multiple Access
SGSN	Serving GPRS Support Node (see GPRS also)
SHCCH	SHared channel Control CHannel
SIM	Subscriber Identity Module
SIR	Signal-to-Interference Ratio
SMS	Short Message Service
SMS-GMSC	Short Message Service Gateway MSC (see MSC also)
SNR	Signal-to-Noise Ratio
SS7	Signaling System 7
SSB	Single Sideband
SSID	Service Set Identifier
SSMA	Spread Spectrum Multiple Access
ST	Signaling Tone
STP	Short Term Prediction
SYN	Synchronisation
TACS	Total Access Communications System
TCAP	Transaction Capability Application Port
TCDMA	Time Division CDMA (see CDMA also)
TCH	Traffic CHannel
TCM	Trellis Code Modulation
TCP/IP	Transmission Control Protocol/Internet Protocol
TDD	Time Division Duplexing
TDFH	Time Division Frequency Hopping
TDMA	Time Division Multiple Access
TD-SCDMA	Time Division Synchronous CDMA (see CDMA also)
TIA	Telecommunications Industry Association
TMSI	Temporary Mobile Subscriber Identity
Tx	Transmitter
UE	User Equipment
UNII	Unlicensed National Information Infrastructure
UMTS	Universal Mobile Telecommunications Service/System
USCH	Uplink Shared Channel
USDC	United States Digital Cellular
UTRA	UMTS Terrestrial Radio Access
UTRAN	UMTS Terrestrial Radio Access Network
UWB	Ultra Wide Band
UWC-136	Universal Wireless Communications Consortium - 136
VLR	Visitor Location Register
VLSI	Very Large Scale Integration
VoIP	Voice over Internet Protocol
VPN	Virtual Private Network
VSELP	Vector Sum Excitation Linear Predictive
UHF	Ultra High Frequency
USIM	USer Identity Module
UTRA	Universal Terrestrial Radio Access
WAP	Wireless Application Protocol
WASP	Wireless Access Service Provider
WCDMA	Wideband Code Division Multiple Access
WDP	Wireless Datagram Protocol
WEP	Wired Equivalent Privacy
Wi-Fi	Wireless Fidelity
WiMAX	Worldwide Interoperability for Microwave Access
WISP	Wireless Internet Service Provider
WLAN	Wireless Local Area Network
WLL	Wireless Local Loop
WMAN	Wireless Metropolitan Area Network
WNIC	Wireless Network Interface Card
WOFDM	Wideband OFDM (see OFDM also)
WPAN	Wireless Personal Area Network
WSN	Wireless Sensor Network
WT	Wireless Terminal
WWAN	Wireless Wide Area Network

Appendix 2
Glossary of Key Terms

1G	First-Generation Cellular Mobile Communications Cellular mobile communications services that use analog techniques, providing voice communications.
1-bit Tags	RFID devices that do not include a chip or memory and cannot store an EPC. 1-bit tags are only used to activate an alarm at retail store entrances as a means of preventing theft.
1/3 Rate FEC	1/3 rate Forward Error Correction An error-correction scheme that repeats each information data bit three times for redundancy.
1xRTT	Single Carrier Radio Transmission Technology A wireless technology based on the CDMA platform, capable of data rates up to 144 kbps. Also referred to as Cdma2000.
2G	Second-Generation Cellular Mobile Communications Cellular mobile communications services that use digital techniques to provide voice communications and relatively low transmission rate for data.
2.5G	Second-Generation plus Cellular Mobile Communications Cellular mobile communications services that enhance 2G techniques to provide higher data transmission rates. It is an interim step between 2G and 3G digital cellular networks. 2.5G sends data up to 384 kbps.
3G	Third-Generation Cellular Mobile Communications Cellular mobile communications services that provide broadband applications such as high quality multimedia up to 2 Mbps with advanced global roaming feature, and is expected to synchronise all of the different specifications used around the world into one universal standard. Known as IMT 2000 by the ITU and implemented in Europe as UMTS and Cdma2000 in North America.
3GPP	3rd-Generation Partnership Project A cooperation of standards organisations throughout the world to expedite the development of open, globally accepted technical specifications for UMTS.
3GPP2	3rd-Generation Partnership Project 2 A cooperation of standards organisations set up to expedite the development of open, globally accepted technical specifications for Cdma2000.
4-PPM	4-Pulse Position Modulation A modulation technique that translates two data bits into four light impulses.
16-PPM	16- Pulse Position Modulation A modulation technique that translates four data bits into 16 light impulses.
802.11	An IEEE standard that defines wireless local area networks at a rate of either 1 Mbps or 2 Mbps. All WLAN features are contained in the PHY and MAC layers (also see IEEE 802.11a/b/g/e/g/i).
802.15	IEEE standards for wireless PANs (WPAN).

8-PS — **8-level Phase Shift eying**
A modulation technique in which the phase of the carrier is shifted in 45-degree increments and 4 bits can be transmitted per phase change.

AC — **Ad acent-Channel nterference**
It is interference caused by inadequate filtering in an adjacent channel.

AC — Acknowledgment—A protocol message sent to acknowledge correct receipt of a transmission.

Active Antenna — A passive antenna with an amplifier built-in.

Active Mode — A state in which a Bluetooth device actively participates on a channel.

Active Scanning — The process of sending frames to gather information.

Active ags — RFID tags that include a battery.

Ad-Hoc Mode — A WLAN mode in which wireless terminals communicate directly among themselves without using an access point.

Ad-Hoc etwork — A network configuration defined for 802.11 wireless LANs where two wireless terminals communicate directly with one another without using an access point. Also called an Independent Basic Service Set (IBSS) or a peer-to-peer connection.

A S — **Advanced ncryption Standard**
The latest encryption standard, developed by the National Institute of Standards and Technology (NIST) to replace the United States data encryption standard (DES). It uses an encryption algorithm used with the 802.11i security protocol.

Air nterface — Air-modulation scheme, equivalent to physical layer in OSI model.

ALOHA — Algorithm that allows multiple terminals to share the same communication channel. Newly arriving packets are transmitted immediately. Retransmitted if no acknowledgement is received. First used in Hawaii.

AMPS — **Advanced Mobile Phone System**
The standard used for 1G analog cellular mobile communication systems based on FDMA technique. It operates in the 800-MHz band.

AM — **Amplitude Modulation**
A change in the amplitude of the carrier signal in proportion to the instantaneous value of the modulating signal.

Analog — The transmission of signals in the form of continuously varying waves. This is the natural form of the energy produced by the human voice.

Analog Modulation — A method of encoding an analog baseband signal onto a high-frequency carrier signal.

Analog Signal — A signal in which the amplitude varies continuously and smoothly over a period of time.

Angle Modulation — Modulation in which the angle of a sine wave carrier signal is varied in proportion to the instantaneous value of the modulating signal. Frequency and phase modulation are particular forms of angle modulation.

Antenna — A copper wire rod, or similar device that has one end up in the air and the other end connected to the ground through a receiver.

Antenna Pattern — A graphic that shows how a signal radiates out of an antenna.

Antenna Polarisation — An indication of the horizontal or vertical orientation of the sine waves leaving an antenna.

AP — **Access Point**
A base station in a wireless LAN. Access points are typically stand-alone devices that plug into an Ethernet hub or server. Like a cellular phone system, users can roam around with their mobile devices and be handed off from one access point to the other.

A Q — **Automatic etransmission equest**
An error-correction scheme that continuously retransmits until an acknowledgment is received or timeout value is exceeded.

AS — **Amplitude Shift eying**
A binary modulation technique in which 1 bit has a carrier signal while a 0 bit has no carrier signal.

Attenuation — A loss of signal strength.

AuC — **Authentication Centre**
Process of verifying the identity of the user at the other end of a link. Part of HLR in 3G systems, to perform computations to verify and authenticate mobile phone users.

Authentication — A process that verifies that the client device asking to join the piconet has permission to access the network.

Authorisation — Process of deciding if a requesting device is allowed to have access to a service on another device. Authorisation always includes authentication.

Bands — Sections of the radio frequency spectrum.

Bandwidth — In the internet industry, bandwidth refers to the capacity of a connection to carry information, while in radio communications it is the range of frequencies that can be transmitted. It is also the difference in Hz between the upper and lower frequencies of a spectrum.

Baseband — A transmission technique that treats the entire transmission medium as only one channel.

Baud ate — The number of times that a carrier signal changes per second.

BCCH — **Broadcast Control CHannel**

Beamwidth — Beamwidth of an antenna pattern is the angle between half-power points of the main lobe, referenced to the peak effective radiated power of the main lobe. Usually expressed in degrees.

B — **Bit- rror ate**
The probability that a transmitting bit is received in error.

Block rror Correction Code — A technique in which a k-bit block of data is mapped into an n-bit block ($n > k$) called a code word, using an FEC encoder,

Bluetooth — A wireless standard that enables devices to transmit data at up to 1 Mbps over a maximum distance of 10 meters.

bps — **bits per second**
The basic unit for rate of transfer of data, which signifies the number of bits that can be transmitted per second.

Broadband — A transmission technique that sends multiple signals at different frequencies. General term used for any type of technology that provides a high data rate (greater than 200 kbps up to 100s of Mbps) capability. Broadband services are usually always-on'. Capable of supporting a variety of voice and data applications, such as voice telephony, internet access, pay TV and multimedia services.

BS — **Base Station**
A land station at a fixed location at the centre or on the edge of a coverage region supporting radio access by mobile users to a fixed communication infrastructure.

BSC	Base Station Controller A network node in cellular systems that supervises the functioning and control of multiple Base Transceiver Stations.
BTS	Base Transceiver Station The equipment housed in cabinets and co-located with antennas, also known as Radio Base Station.
Burst Error	A burst error is a contiguous sequence of bits in which the first and last bits and any number of intermediate bits are received in error.
Carrier Frequency	A continuous frequency capable of being modulated with an information-carrying signal.
Carrier Signal	A transmission over a radio frequency that carries no useful information.
CCK	Complementary Code Keying A table containing 64 8-bit code words used for transmitting at speeds above 2 Mbps.
CDMA	Code Division Multiple Access Multiple access method based on spread spectrum in which different users transmit on the same carrier frequency, but use different spreading codes.
Cdma2000 IxEVD	The 3G digital cellular technology that is a migration from Cdma2000 IxRTT.
Cdma2000 IxRTT	A 2.5G digital cellular network technology that is a migration from CDMA (1xRTT stands for l-times Radio Transmission Technology).
cdma ne	2G CDMA (IS 95) standard.
Cell	The area covered by radio signals from a base station, and in which a mobile station can successfully transmit to a base station.
Cell Sectorisation	Splitting (theorectically hexagonal) cells into multiple independent sectors (typically 3 or 6) that each have their own transmit and receive facilities.
Cell Splitting	A method of increasing capacity by reducing the size of the cell.
Cellular Network	A wireless communication network in which low-power fixed base station transmitters are arranged in a hexagonal pattern and mobile subscribers communicate through nearby base stations.
Channel	A channel may signify wither a one-way or a two-way path for transmitting electrical signals.
Channels	Another name for frequencies.
Channel Capacity	The maximum possible information rate through a channel subject to the constraints of that channel.
Circuit Switching	A dedicated communication path is established between two devices through one or more intermediate switching nodes.
Cluster Size	Number of different channels needed in a particular frequency reuse plan. Related to reuse distance.
Code Division Multiple Access	A multiplexing technique used with spread spectrum.
Coherence Bandwidth	The coherence bandwidth is a statistical measure of the range of frequencies over which the channel can be considered flat.
Coherence Time	Coherence time is the time duration over which two received signals have a strong potential for amplitude correlation.

Collision	The garbling of data that occurs when two computers start sending messages at the same time in a shared channel.
Constellation Diagram	A graphical representation that makes it easier to visualise signals using complex modulation techniques such as QAM.
Control Channel	The radio channel used for transmission of call initiation and other control signaling purposes.
Convergence	The act of bringing the Internet to devices that previously couldn't access it. This could include the merging of the Internet with television, cellular phones or even household appliances.
Coverage Area	Geographical area within which mobile phone calls can be made. Coverage can be increased by installing radio base stations in new areas or by installing equipment to extend the range of coverage.
CRC	Cyclic Redundancy Check A common technique for detecting data transmission errors in which the code is the remainder resulting from dividing the bits to be checked by a predetermined binary number.
Data Link Layer	The layer responsible for the transfer of data between nodes in the same network segment and that also provides error detection.
Data Rate	The volume of data that is able to be transmitted over a period of time. Data rates are usually measured in bits per second.
dB	Decibel – A ratio between two signal levels. A measure of the relative signal strength of two signals. The number of decibels is 10 times the log of the ratio of the power of two signals.
dBd	dB dipole The relative measurement of the gain of an antenna when compared to a dipole antenna.
dBi	dB isotropic The relative measurement of the gain of an antenna when compared to a theoretical isotropic radiator.
dBm	A relative way to indicate an absolute power level in the linear watt scale with respect to 1 mW.
DCF	Distributed Coordination Function The default access method for WLANs.
DECT	Digital Enhanced Cordless Telephony It is a European cordless telephone standard. Designed for wireless PABX connections to provide voice and data services using TDMA technique.
Delay Spread	The delay spread determines to what extent the channel fading at two different frequencies are correlated.
DES	Data Encryption Standard The encryption standard used in the United States until the adoption of AES (see Advanced Encryption Standard).
Dial- p Subscribers	Subscribers who connect to the internet via modem and dial-up software utilising the public switched telecommunications network, including ISDN connections that require the user to dial-up.
Dibit	A signal unit that represents two bits.
Diffraction	Diffraction is referred to the change in wave pattern caused by interference between waves that have been reflected from a surface or a point.
Digital Convergence	The power of digital devices, such as desktop computers and wireless handheld devices, to combine voice, video, and text-processing capabilities, as well as to be connected to business and home networks and to the Internet.

Digital Data	Data consisting of a sequence of discrete elements.
Digital Modulation	A method of encoding a digital signal onto an analog carrier wave for transmission over media that does not support direct digital signal transmission.
Digital Signal	A discrete or discontinuous signal, such as voltage pulses.
Digital Transmission	The transmission of digital data, using either an analog or digital signal.
Dipole Antenna	An antenna that has a fixed amount of gain over that of an isotropic radiator.
Direct Wave Path	It is a radio signal path from the transmitter to the receiver that is clear form of terrain contour.
Directional Antenna	An antenna that radiates the electromagnetic waves in one direction only. As a result, it can help reduce or eliminate the effect of multipath distortion, if there is a clear line of sight between the two antennas.
Directional Gain	The effective gain that a directional antenna achieves by focusing RF energy in one direction.
Diversity	Technique of receiving a radio signal through multiple channels, to improve reliability of the signal reception.
Doppler Shift	The effective change of frequency of a received signal due to the relative velocity of a mobile receiver with respect to a transmitter.
Doppler Spread	One half times the width of the spectrum of a received signal when a sinusoidal wave is transmitted over a time dispersive channel.
DoS	Denial-of-Service A type of security attack on a networked device in which the attacker sends so many frames to a single device that the device is unable to communicate with other devices.
Downlink	The communication link from base station to mobile subscriber.
DSSS	Direct Sequence Spread Spectrum A spread spectrum technique that uses an expanded, redundant code to transmit each data bit. Each bit in the information signal is represented by multiple bits in the transmitted signal, using a spreading code.
Duplex	Method of operating a network in which transmission is possible simultaneously in both directions of a telecommunications channel.
EDGE	Enhanced Data rates for GSM Evolution A 2.5G digital cellular network technology that boosts GPRS data rate transmissions.
EIR	Equipment Identity Register A database used to verify the validity of equipment being used in mobile networks. It can provide security features such as blocking of calls from stolen mobile phones and preventing unauthorised access to the network.
Encapsulation	The addition of control information by a protocol entity to be obtained from a protocol user.
Encryption	The process of encoding communications to ensure that the transmissions cannot be easily intercepted and decoded.
EPC	Electronic Product Code A standardised numbering scheme that can be programmed in a tag and attached to any physical product.
Equalisation	Signal processing intended to undue channel dispersion and avoiding undesirable noise enhancements.
Erlang	Unit of telephone traffic intensity.

ESS	Extended Service Set A WLAN mode that consists of wireless clients and multiple Access Points.
ETSI	European Telecommunications Standards Institute European organisation responsible for establishing common industry-wide telecommunication standards.
Fade Rate	The number of times that the received signal envelope crosses the threshold value in a positive going direction per unit time.
Fade Duration	The time period for which the received signal is below a specified received signal level.
Fading	Fading describes the rapid fluctuations of the amplitudes, phases, or multipath delays of a radio signal over a short period of time or travel distance so that large-scale path loss effects may be ignored.
Fast Fading	Fast fading refers to rapid fluctuations in received signal strength occur over distances of about one-half a wavelength as the mobile subscriber moves in an urban environment.
FCC	Federal Communications Commission The primary U S regulatory agency for telecommunications.
FDD	Frequency Division Duplexing A mechanism that uses one frequency for uplink and another frequency for downlink transmissions.
FDMA	Frequency Division Multiple Access A radio transmission technique that divides the bandwidth of the frequency into several smaller frequency bands. Multiple access method in which different users transmit at different carrier frequencies.
FHSS	Frequency Hopping Spread Spectrum A spread spectrum technique that uses a range of frequencies and changes frequencies during the transmission over a seemingly random sequence of radio frequencies, hopping from one frequency to another frequency at fixed intervals.
Flat Fading	Flat fading is that type of fading in which all frequency components of the received signal fluctuates in the same proportions simultaneously.
FM	Frequency Modulation A change in the frequency of the carrier signal in proportion to the instantaneous value of the modulating signal.
Forward Channel	In a cellular or cordless system, the communication link from the base station to the mobile unit.
Frame	A data link layer packet that contains the header and trailer required by the physical medium.
Free Space Loss	The signal loss that occurs as a result of the tendency of RF waves to spread, resulting in less energy at any given point, as the signal moves away from the transmitting antenna.
Frequency	A measurement of radio waves that is determined by how frequently a cycle occurs.
Frequency Reuse	Assignment of the same frequency channel in multiple cells following defined rules, and simultaneous use of these channels allowed by propagation losses between spatially separated areas.
Frequency Selective Fading	Frequency selective fading affects unequally the different spectral components of a radio signal.
Fresnel one	An elliptical region spanning the distance between two directional antennas that must not be blocked more than 40% to prevent interference, with the RF signal.

FSK	Frequency Shift Keying A binary modulation technique that changes the frequency of the carrier signal.
Full-Duplex Transmission	Transmissions that enable data to flow in either direction simultaneously.
Full-Wave Antenna	An antenna that is as long as the length of the wave it is designed to transmit or receive signal.
Gain	A relative measure of increase in a signal's power level.
GGSN	Gateway GPRS Support Node One of the two main GPRS nodes, which provides the interface between the radio network and the IP network.
GHz	Gigahertz – 1,000,000,000 Hertz.
GFSK	Gaussian Frequency Shift Keying A binary signaling technique that uses two different frequencies to indicate whether a 1 or a 0 is being transmitted in addition to varying the number of waves.
GPRS	General Packet Radio Service A 2.5G cellular network technology as an enhancement for GSM and TDMA core networks that introduces packet data transmission. GPRS uses radio spectrum very efficiently and provides users with 'always on' connectivity and greater bandwidth that can transmit up to 114 kbps.
GPS	Global Positioning System A system of 24 satellites for identifying earth locations, launched by the U S Department of Defense. By triangulation of signals from three of the satellites, a receiving unit can pinpoint its current location anywhere on earth to within a few metres.
GSM	Global System For Mobile Communication Pan-European 2G digital cellular network standards currently operate GSM networks in the 900-MHz band, with planned use in the 1800-MHz band. The standard employs a combination of FDMA and TDMA.
GTP	GPRS Tunneling Protocol Creates a secure connection in the IP environment by encapsulating encrypted data in an IP packet.
HA	Home Agent A router on a mobile node's home network that serves as a point for communications with it when away from home, and maintains current location information for the mobile node.
Half-Duplex Transmission	Transmission that occurs in both directions but only one way at a time.
Half-Wave Antenna	An antenna that is half as long as the wavelength of the signal it is designed to transmit or receive signal.
Hamming Distance	The number of digit positions in which two binary numbers of the same length are different.
Hand-off	The transition that occurs when a mobile phone or wireless terminal connects with a new cell or access point and disconnects from the previous one in cellular or piconet communications.
Header	System-defined control information that precedes user data.
Hidden Node	A mobile node in a CSMA network actively transmitting data, but which is not noticed by another node with data ready for transmission.

HIPERLAN	High-Performance Radio Local Area Network An European WLAN standard that is similar to the IEEE 802.11a WLAN standard.
HLR	Home Location Register A permanent database used in cellular mobile systems to identify subscribers and to contain subscriber data related to features and services.
HSCSD	High-Speed Circuit Switched Data This technology affords increased mobile network bandwidths.
Hz	Hertz – The number of cycles per second.
IBSS	Independent Basic Service Set A WLAN mode in which wireless terminals communicate directly among themselves without using an AP.
IEEE	Institute of Electrical and Electronics Engineers A standards body that establishes standards for telecommunications.
IEEE 802.11a	A family of WLAN transmissions standards, covering wireless short-range communications networks with speeds up to 54 Mbps in the ISM frequency band of 5 GHz.
IEEE 802.11b	An addition to the lEEE 802.11 standard for WLANs that added two higher speeds, 5.5 Mbps and 11 Mbps in the 2.4 GHz unlicensed ISM frequency band. Also known as Wi-Fi.
IEEE 802.L1e	A standard for WLAN applications that requires QoS and provides for improvements in the capabilities and efficiency of the protocol.
IEEE 802.11g	A standard for WLAN transmissions for networks with speeds up to 54 Mbps using the ISM band.
IEEE 802.11i	An enhancement to 802.11 that deals with security enhancements of the original standard.
IEEE 802.16	A set of Fixed Broadband Wireless standards for fixed and mobile broadband wireless communications that allows computers to communicate at up to 75 Mbps and at distances of up to 56 km in a point-to-point configuration in both licensed and unlicensed frequencies.
IETF	Internet Engineering Task Force A standards body that focuses on the lower levels of telecommunications technologies.
IM Noise	Intermodulation Noise Noise due to the nonlinear combination of signals of different frequencies.
Impulse Noise	A high-amplitude, short duration noise pulse.
IMT 2000	International Mobile Telecommunications 2000 The ITU initiative for standardising radio access to the 3G global telecommunications infrastructure.
Infrared	Electromagnetic waves whose frequency range is more than that of microwave and below the visible spectrum, that is, $3x10^{11}$ to $3x10^{14}$ Hz.
Infrastructure Mode	A WLAN mode that consists of wireless terminals and at least one AP.
In-Phase Component	Component of a signal that has the same phase as a reference sinusoidal signal.
Interleaving	Intentional resequencing or reshuffling of the data bits in a signal according to a predefined method known by both transmitter and receiver, to avoid burst errors.
Ionosphere	That part of the earth's outer atmosphere (50 km to 400 km above the earth's surface) where ionisation caused by incoming solar radiation affects the transmission of radio waves.

P **nternet Protocol**
Protocol for transmission of data over the internet. The IP part of the TCP/IP communications protocol. IP implements the network layer (layer 3) of the protocol, which contains a network address and is used to route a message to a different network or sub network. IP accepts packets from the layer 4 transport protocol (TCP or UDP), adds its own header to it and delivers a datagram to the layer 2 data link protocol. It may also break the packet into fragments to support the maximum transmission unit of the network.

Pv4 **nternet Protocol version four**
The version of IP most commonly deployed today.

pv6 **nternet Protocol version six**
This will, among other things, add significantly to the address capacity, security and real-time capability of IP.

rDA **nfrared Data Association**
It is a set of specifications for wireless infrared communications.

S-136 **nterim Standard – 136**
Second-generation TDMA standard, also called Digital AMPS or D AMPS.

S-95 **nterim Standard – 95**
First generation digital cellular CDMA standard (cdmaOne).

SD **ntegrated Services Digital etwork**
A technology that enables digital transmission of voice and data over the telephone lines at a maximum of 256 Kbps data rate in Public Switched Telephone Network (PSTN).

SM **ndustrial, Scientific and Medical band**
An unregulated radio frequency band approved by the FCC in 1985 as ISM band. Unlicensed spectrum typically in the 900-MHz, 2.4-GHz and 5.7-GHz bands. Requires spread spectrum techniques at 1 watt.

SP **nternet Service Provider**
Service provider offering internet access to the public or another service provider.

U **nternational elecommunications Union**
Part of the United Nations responsible for co-coordinating global telecommunications activities, especially in the area of standards.

u UMTS interface between the core network and the RAN (radio access network).

Jamming Deliberate radiation of electromagnetic energy with the intent to jam the desired signals by sinusoidal (CW), noise-like or broadband transmitters, specific deceptive signals that imitate messages.

B **kilobyte**
A thousand bytes. 1 kB = 1024 Bytes. Also see **byte**

bps **kilobits per second**
Data communications transmission rate of 1,000 bits per second.

LA **Local Area etwork**
A group of computers and associated devices that share a common communications line and typically share the resources of a single processor or server within a small geographic area. Usually, the server has applications and data storage that are shared by multiple PC users.

Latency Delays caused by signals that must travel over a long distance. Also, the amount of time delay that it takes a packet to travel from source to destination device.

LLC **Logical Link Control**
One of the two sub layers of the IEEE Project 802 Data Link layer.

LMDS	Local Multipoint Distribution Service A fixed broadband technology that can provide a wide variety of wireless services. Licensed spectrum above 20 GHz. Range is about 6-7 kilometers.
L S	Line-of-Sight The direct radio path between transmitter and receiver without any obstruction.
Loss	A relative measure of decrease in a signal's power level.
MAC	Media Access Control One of the two sub layers of the IEEE Project 802 Data Link layer.
MAC Address	A 3-bit address used to distinguish devices within a piconet. 3 bits allows for only 8 unique address, thus the limit on the number of devices allowed to participate in a piconet
MAP	Mobile Application Part Part of the SS7 protocol used in GSM. MAP standards address registration of roamers and intersystem hand-off procedures.
Master Device	This is the device within a piconet whose clock and frequency-hopping sequence is used to synchronize all slave units within the piconet.
Maximum-Ratio Combining	Method of pre-filtering and adding signals arriving through different branches of a diversity receiver, by weighing a signal proportionally to its amplitude.
MB	Megabyte One million bytes. Also, see byte kB.
Mbps	Megabits per second Data communications transmission rate of one million bits per second. Also see bps kbps.
Mesh Networking	A network topography in which each device connects to all other devices within range.
Microcell	Cell with relatively small size, typically a few hundreds of meters, used in a dense cellular network with many subscribers.
MIM	Multiple In Multiple ut A technology that uses multiple antennas and also uses multipath reflected signals to extend the range of WLAN by attempting to correctly decode a frame from multiple copies of it received at different times.
MMDS	Multichannel Multipoint Distribution Service A fixed broadband wireless technology that transmits at 1.5 Mbps over distances of 56 kilometres in licensed spectrum of 2.5–2.6 GHz band.
MMS	Multi-Media Messaging Mobile telecommunications data transmission service for sending messages with a combination of text, sound, image and video to MMS-capable mobile handsets.
Mobile Subscriber	A user terminal in a cellular radio network intended to be used while in motion or during halts at unspecified locations.
Modem	Modulator + DEModulator A device used to convert digital signals into an analog format, and vice versa.
Modulation	The process of changing the characteristics of a carrier signal.
Modulation Index	The amount that the characteristics of a carrier signal varies.
Monopole Antenna	An antenna formed of a straight piece of wire, usually a quarter of the wavelength with no ground point or reflecting element.

Motes	Remote sensors used for collecting data from manufacturing equipments or for scientific research that can communicate using wireless technology.
MSC	Mobile Switching Centre Telecommunications node connecting and controlling several cellular base stations.
MTS	Mobile Telecommunications Switching ffice The connection between a cellular network and wired telephone network.
Multipath	The propagation phenomenon that results in signals reaching the receiving antenna by two or more signal paths.
Multipath Distortion	The same signal being received from several different directions and also at different times in wireless transmissions.
Multipath Interference	Multipath interference is the reflection of radio signals from concrete structures that results in multiple copies of the received signal.
Multiple Access	Method that allows multiple spatially separated users to share the same communication channel to a common base station receiver.
Narrowband	Narrowband describes a class of telecommunications services such as dial-up internet access that offer a data rate of up to 64 kbps.
Narrow band Transmissions	Transmissions that use one radio frequency or a very narrow portion of the frequency spectrum.
Neighbour Piconets	Separate piconets that have their own PNC, but that depend on the original piconet's PNC to allocate a private block of time when their devices are allowed to transmit.
NI	Network Interface nit A device that connects an LMDS modem to a LAN or telephone system.
NL S	Non-Line-of-Sight When the transmitter antenna cannot be seen from the receiver end or vice-versa.
NMT	Nordic Mobile Telephone System First analog cellular telephone system in 1979 in Scandinavian countries, operates at 450 or 900 MHz.
Noise	Interference with a signal. Unwanted signal that combine with the signal and distort it which was intended for transmission and reception.
Nomadic ser	A user that moves frequently but does not use the device while on move.
bstructive Path	It is a radio signal path between the transmitter and receiver when the terrain contour blocks the direct wave path.
FDMA	rthogonal Frequency Division Multiple Access A multiple access technique, based on OFDM that divides the frequency channel into 1,536 data sub carriers.
FDM	rthogonal Frequency Division Multiplexing A transmission technique in which the frequency band is divided into a number of frequencies (called sub-frequencies or channels) that do not interfere with each other. Multi-carrier modulation method with partially overlapping orthogonal sub carriers.
mnidirectional Antenna	An antenna that radiates the signal in a uniform pattern in all directions.
-QPSK	ffset Quadrature Phase Shift Keying A digital modulation technique that uses two carrier waves of the same frequency but with a phase difference of 90 degrees between them. This technique modulates even numbered chips in the in-phase (I-phase) signal and odd numbered chips in the quadrature phase (Q-Phase) signal.

SS	perations Support System Methods and procedures that directly support the daily operation of a telecom network.
Packet	A piece of data transmitted over a packet-switching network such as the Internet. A packet includes not just data but also its destination.
Packet Switching	A method of subdividing long messages into short packets and then transmitting them through a communication network.
Page	A brief message broadcast over the whole service area in a simulcast manner by many base stations simultaneously.
Paging	Communication service that offers one-way transmission of short messages in the form of an audible tone or alphanumeric text.
Parity Bit	A binary digit appended to an array of binary digits to make the total sum of bits odd (odd parity) or even (even parity).
Parked Devices	Those devices within a piconet which are synchronised but do not have a MAC address.
Park Mode	A state in which a Bluetooth device is still synchronised to the piconet but it does not participate in the traffic.
Passive Antenna	The most common type of antenna, which can only radiate a signal with the same amount of energy that appears at the antenna connector.
Passive Scanning	The process of listening to each available channel for a set period of time.
Passive Tags	The most common type of RFID tags. They never initiate a transmission. They do not include a battery and are powered by the electromagnetic energy in the RF waves transmitted by the reader.
Patch Antenna	A semi-directional antenna that emits a wide horizontal beam and an even wider vertical beam.
Path Loss	Average propagation attenuation between transmitter and receiver. Depends on transmitter power, distance, antenna heights, atmospheric and terrain characteristics among other parameters.
PCF	Point Coordination Function The 802.11 optional polling functions.
PCS	Personal Communications Service New digital cellular phone systems in the 2-GHz range.
PDA	Personal Digital Assistant A handheld computer or personal organiser device used for taking notes, making appointments, creating to-do lists, and communicating with other devices.
PDC	Personal Digital Cellular Second-generation digital cellular phone system in Japan, operates in the 900-MHz and 1.5-GHz band. The system uses both full, and half-rate speech code (5.6 kbps) and allows high speed transmission at 9.6 kbps to ensure efficient spectrum utilization.
Peer-To-Peer Mode	A WLAN mode in which wireless terminals communicate directly among themselves without using an AP.
PH	Physical Layer – The layer that is responsible for converting the data bits into an electromagnetic signal and transmitting it on the medium.
Piconet	A small network composed of two or more Bluetooth devices that contain one master and at least one slave exchanging data using the same channel.
PM	Phase Modulation A change in the phase of the carrier signal in proportion to the instantaneous value of the modulating signal.

PNC	Piconet Coordinator – A device that provides all the basic communications timing in an 802.15.3 piconet.
PN Code	PseudoNoise Code – a code that appears to be a random sequence of 1s and 0s but actually repeats itself. Used in CDMA cellular telephone technology.
Point-To-Multipoint Wireless Link	A link in which one central site uses an omnidirectional antenna to transmit to multiple remote sites, which may use omnidirectional or directional antennas to maximise the range and quality of the signal.
Point-To-Point Wireless Link	The most reliable link between two antenna sites using directional antennas to maximise the range and quality of the signal.
Polling	A channel access method in which each computer is asked in sequence whether it wants to transmit.
Power Management	An 802.11 standard that allows the mobile set to be off as much as possible to conserve battery life but still not miss out on data transmissions.
Privacy	Standards that ensure that transmissions are not read by unauthorised users.
Probe	A frame sent by a client when performing active scanning.
Probe Response	A frame sent by an AP when responding to a probe from active scanning wireless terminals.
Propagation	Natural mechanism of dissemination of radio energy.
PSK	Phase Shift Keying A binary modulation technique that changes the starting point (phase angle) of the cycle.
PSTN	Public Switched Telecommunications Network Public telecommunications network operated by a carrier to provide services to the public.
QAM	Quadrature Amplitude Modulation A combination of phase modulation with amplitude modulation to produce different signals.
QPSK	Quadrature Phase Shift Keying A digital modulation technique that combines quadrature amplitude modulation with phase shift keying.
QoS	Quality f Service A feature of some wireless PANs that allows devices to request more channel access time in order to prioritise high-volume, time-sensitive such as a voice stream.
Quad Bit	A signal unit that represents four bits.
Quadrature Component	Component of a signal that is orthogonal to (90 degrees out of phase with) a reference sinusoidal signal.
Quarter-Wave Antenna	An antenna that is one-fourth as long as the wavelength of the signal it is designed to transmit or receive.
Radiation Pattern	A graphical representation of the radiation properties of antennas as a function of space coordinates.
Radio Frequency Spectrum	The entire range of all radio frequencies that exist.
Radio Modules	Small radio transceivers built onto microprocessor chips that are embedded into Bluetooth devices and enable them to communicate.
Radio Path	A radio path is a path traveled by the radio signal in the wireless medium from the transmitter to the receiver.

Radio Wave	An electromagnetic wave created when an electric current passes through a wire and creates a magnetic field in the space around the wire.
Rake Receiver	Special form of a matched filter which can optimally detect direct sequence spread spectrum signals over a multipath channel.
RAN	Radio Access Network The portion of a mobile network that handles subscriber access, including radio base stations and concentration nodes.
Random Access	Method or algorithm that allows multiple terminals to share the same communication channel.
Rayleigh Fading	Fading characterised by a Rayleigh probability distribution function of the amplitude of an infinitely large number of reflected waves.
Reader	The RFID device that captures and processes the data received from the tags.
Reassociation	The process of a wireless terminal dropping a connection with one AP and reestablishing the connection with another.
Reflection	Reflection occurs when incident electromagnetic waves are partially reflected when they impinge on obstructions of different electrical properties.
Refraction	Refraction occurs when the velocity of the electromagnetic waves varies due to different density of the medium through which it travels.
Repeater	A device commonly used to repeat the radio signal to another location.
Reuse Distance	Distance between the centers of two cells using same frequency channel in a cellular configuration.
Reverse Channel	In a cellular or cordless system, the communication link from the mobile unit to the base station.
Rician Fading	Fading characterised by a Rician probability distribution density function of the amplitude in situations where a dominant line-of-sight wave plus an infinitely large number of reflected waves occur in mobile radio propagation environment.
RF	Radio Frequency Communications All types of radio communications that use radio frequency waves.
RFID	Radio Frequency Identification A technology that uses electronic, flexible tags, equipped with microprocessor chips and memory, to identify products. RFID tags can store significantly more information than the current barcode system.
RFID Device	A small tag placed on product packaging and boxes that can be remotely activated and read by remote sensors. The data about the product is then transferred directly to information processing system for inventory control, location, and counting.
RNC	Radio Network Controller Manages the radio part of the network in UMTS.
Roamer	A mobile subscriber operation in a service area other than that from which service has been subscribed.
Roaming	The automatic transfer of the RF signal when moving from one cellular network to another network.
Router	A device used to link two or more networks.
RTS/CTS	Request to Send/Clear to Send An 802.11 protocol option that allows a station to reserve the network for transmissions.

Satellite	A satellite is a wireless receiver/transmitter that operates in orbit around the earth and acts as a microwave relay station, receiving signals sent from a ground-based station, amplifying them, and retransmitting them on a different frequency to another ground-based station.
Scanning	The process that a wireless terminal uses to examine the radio waves for information that it needs in order to begin the association process.
Scattering	Scattering is a special case of reflection caused by irregular objects result in many different angles of reflection and scatter waves in all directions in the form of spherical waves.
Scatternet	A group of piconets in which connections exist between different piconets.
Selective Fading	Fading that affects unequally the different spectral components of a radio signal.
Semi-Active Tag	RFID tags which include a battery that is only used when the tag is interrogated. The batteries in semi-active tags usually last for several years.
Sensory Tags	RFID devices that include a thermal or other kind of sensor and can record information about the environmental conditions to which a product has been exposed during transportation or storage.
Server	Device or application that passively waits for connection requests from one or more user terminals.
Service Area	Typically includes the coverage areas of multiple cells in which mobile communication can be established with a mobile station.
SGSN	Serving GPRS Support Node The SGSN handles the data traffic of users in a geographical service area and is one of the two types of node implemented in a GPRS environment.
Sidebands	The sum and the differences of the frequency of the carrier signal that serve as buffer space around the frequency of the transmitted signal.
SIM	Subscriber Identity Module Smart card that gives GSM phone its user identity. It permits cellphone equipment to be easily rented or borrowed.
Simplex Transmission	The transmission that occurs in only one direction.
Slave Device	A device on a Bluetooth piconet that takes commands from the master.
Sleep Mode	A power-conserving mode used by notebook computers.
Slow Fading	Slow fading refers to change in the average received power level about which the rapid fluctuations occur as the mobile subscriber covers distances well in excess of a wavelength due to changes in the urban environment over these longer distances.
Smart Antenna	An array of narrow-beam antenna elements and associated signal processing, used to track the user and improve the performance by minimising the effect of interference.
Smart Labels	Another name for flexible RFID tags that include a microprocessor chip and memory.
Smartphone	A device that combines a cellular phone with the capabilities of a Personal Digital Assistant (PDA). These devices provide the user with the ability to enter appointments in a calendar, write notes, send and receive e-mail, and browse websites, among other functions.
SMS	Short Message Service Mobile telecommunications data transmission service used in cellular phones and pagers that allows users to send short text messages to each other using the keypad.
SNR	Signal-to-Noise Ratio The measure of signal strength relative to the background noise.

Sniff Mode	A state in which the Bluetooth device listens to the piconet master at a reduced rate so that it uses less power.
Spectrum	Refers to an absolute range of frequencies.
Spread Spectrum	A transmission and modulation technique in which the information in a signal is intentionally spread over a wider bandwidth using a spreading code.
Spreading Code	A sequence of bits used to spread bandwidth in a spread spectrum system. Also called a spreading sequence or a chipping code.
SS7	Signaling System 7 The protocol used in the intelligent network for setting up calls and providing services.
SSID	Service Set Identifier A unique identifier assigned to an AP.
Subscriber	A user who pays subscription charges for using a communication system.
Superframe	A mechanism for managing transmissions in a piconet. The superframe is a continually repeating frame containing a beacon, contention access periods, channel-time allocation periods, and management-time allocation periods. Using the superframe is optional in 802.15.4 WPANS.
TACS	Total Access Communications System Analog cellular phone system in Europe based on AMPS.
Tags	Devices that include an antenna and a chip containing memory and can store information about products, such as the manufacturer, product category, and serial number, along with date and time of manufacturing.
TCM	Trellis Code Modulation A method of encoding a digital signal in a way that permits single bit errors to be detected and corrected.
TCP/IP	Transmission Control Protocol/Internet Protocol A global communications protocol developed to inter network dissimilar systems. TCP provides transport functions, which ensures that the total amount of bytes sent is received correctly at the other end. IP provides the routing mechanism. TCP/IP is a routable protocol, which means that all messages contain not only the address of the destination station, but the address of a destination network. This allows TCP/IP messages to be sent to multiple networks within an organisation or around the world, hence its use in the worldwide Internet. Every client and server in a TCP/IP network requires an IP address, which is either permanently assigned or dynamically assigned at startup.
TDD	Time Division Duplexing A mechanism that divides a single transmission into two parts, an uplink part and a downlink part.
TDMA	Time Division Multiple Access A transmission technique that divides the bandwidth into several time slots. Multiple access method in which different users transmit in different time intervals.
Time Slots	The smallest unit in a TDMA frame. The length of a time slot is predefined by the standard or specification for a particular system.
Transceiver	An equipment capable of transmitting and receiving radio signals simultaneously.
Transmission Medium	The physical path between transmitters and receivers in a communication system.
Troposphere	That part of the earth's atmosphere in which temperature generally decreases with altitude.
Trunking	Use of the radio spectrum in which multiple user groups share the same channels using multiple access technique, thus gaining efficiency.

wo-way Paging The ability to receive and send data to the Internet by way of the paging network. Also often called interactive paging.

UM S **Universal Mobile elecommunications System**
It is the European implementation of the 3G generation cellular technology for rapidly moving data and multimedia over wireless devices. UMTS provides service in the 2 GHz band and offers global roaming and personalised features. Designed as an evolutionary system for GSM network operators, multimedia data rates up to 2 Mbps are expected eventually in VPN's.

U- **Unlicensed ational nformation nfrastructure**
An unregulated band approved by the FCC in 1996 to provide for short-range, high-speed wireless digital communications.

Uplink The communication link between subscriber unit to base station.

U A **UM S errestrial adio Access etwork**
The name of the WCDMA radio network in UMTS.

UWB **Ultra Wide Band**
A wireless communications technology that allows devices to transmit data at hundreds of megabits per second at short distances up to 50 metres at lower speeds. The transmissions use low-power, precisely timed pulses of energy that operate in the same frequency spectrum as low-end noise, such as that emitted by computer chips and TV monitors.

ocoder Voice encoder in which speech is heavily compressed to reduce the channel bit rate required to transmit speech.

o P **oice over nternet Protocol**
A protocol for transmitting voice over packet-switched data networks. A technology that allows voice telephone calls to be carried over the same network used to carry computer data. Also called IP telephony.

S LP **ector Sum xcited Linear Predictive**
Commonly used method for speech coding.

WAP **Wireless Application Protocol**
An open, global specification that allows mobile users with wireless devices to access data services (including the Internet). Typically provides access to email and information such as news, sport, weather, flight schedules, stocks and shares, banking or shopping.

WASP **Wireless Application Service Provider**
Organisations that can design, create, and deliver a complete wireless application.

Wavelength The distance between two points in a periodic wave that has the same phase.

WCDMA **Wideband Code Division Multiple Access**
The 3D digital cellular technology that is a migration from EDGE. Technique that uses direct spreading of data. Supports mobile or portable voice, data and video communication at up to 2 Mbps (local area access) or 384 kbps (wide area access).

Wi-Fi **Wireless Fidelity**
A trademark of the Wi-Fi Alliance, often used to refer to IEEE 802.11b WLANs. It provides short-range, high data rate connections between mobile data devices and access points connected to a wired network.

WiMA **Worldwide nteroperability for Microwave Access**
Industry group organised to advance the IEEE 802.16 standards for broadband wireless access networks for multimedia applications with a wireless connection.

Wireless	Refers to electromagnetic transmission through space or air by means of an antenna.
Wireless Bridge	A networking component that connects two wired networks or extends the range of a WLAN.
Wireless Communications	The transmission of user data without the use of wires between devices.
Wireless NIC	Wireless Network Interface Card A device that connects to a PC to transmit and receive network data over radio waves. It includes an antenna for wireless communication between networked devices.
WLAN	Wireless Local Area Network Network using radio communications to connect computer terminals or other digital devices over relatively short distances. Its range extends to approximately 100 metres and has a maximum data rate of 54 Mbps by current standards. Today's WLANs are based on IEEE 802.11a/b/g standards. See also IEEE 802.11.
WLL	Wireless Local Loop Use of radio communications rather than cable to connect a fixed or mobile handset and a telecommunications base station connected to a telecommunications network.
WMAN	Wireless Metropolitan Area Network A wireless network that covers a large geographical area such as a city or suburb. The technology is based on the IEEE 802.16 set of standards, and can cover distances of up to 56 km.
WPAN	Wireless Personal Area Network A group of technologies that are designed for short-range communications, from few meters to about 10 metres. Due to its limited range, WPAN technology is used mainly as a replacement for cables. Sec also piconet and UWB.
WWAN	Wireless Wide Area Network A network that can encompass geographical area, even the entire world. WWANs use cellular phone technologies.

Model Test Paper - Type 1

Wireless Communications

Max. Time: 3 Hours Max. Marks: 100

N TE Attempt any FIVE questions. All questions carry equal marks.

Q1 (a) Compare and contrast the features of second-generation digital cellular standards—GSM and CDMA technologies.

(b) What are advantages of cellular mobile communication systems over conventional mobile telephone systems? 12 + 8

Q2 (a) A cellular architecture is configured with regular hexagonal cell geometry. The total service area is divided into cell clusters with frequency reuse. Prove that the distance D between the centres of two closest cochannel cells is given by $D = R\sqrt{3(i^2 + j^2 + i \times j)}$; where R is the cell radius, having same units as D; and i, j are non-negative integers which describe the geometry relation between adjacent cells.

(b) A large city with an area of 1000 km^2 is required to be covered by a finite number of cells with a radius of 2 km each. How many cell-sites would be required, assuming regular hexagonal-shaped cells? 16 + 4

Q3 (a) Explain fixed, dynamic, and hybrid channel-assignment strategies in a cellular system.

(b) A cellular mobile communication system operates at 900 MHz. Compute the propagation-path loss at a distance 5 km away from the cell-site. Assume the height of the cell-site transmitting antenna is 50 m and the mobile receiving antenna is 1.5 m above ground. 10 + 10

Q4 (a) Suggest at least five different means to increase the radio coverage of a cell.

(b) On what basis are multiple access radio protocols classified? Distinguish between various types of multiple access protocols. 10+10

Q5 (a) Describe the step-by-step procedure for placing a call from a calling mobile subscriber to a called landline telephone subscriber.

(b) In a US AMPS cellular system, a service provider is allocated a total spectrum bandwidth of 10 MHz. Out of this, 1 MHz spectrum is reserved for signaling and control channels. If a 7-cell reuse pattern is used, calculate the radio capacity of a cell. 12 + 8

Q6 (a) Explain the main properties of the basic multiple access techniques—FDMA, TDMA, and CDMA.

(b) Adaptive power control is the solution to the near–far interference problem in a CDMA cellular system. Describe two basic concepts of power control employed for CDMA. 8 + 12

Q7 (a) Explain the additional features which the next-generation wireless networks are likely to have over and above the present 3G wireless technologies.

(b) Why must an omnidirectional antenna be used at the mobile unit in a mobile radio environment? $16 + 4$

Q8 (a) What is the difference between a physical channel and a logical channel? Describe the important functions of various types of logical channels in GSM.

(b) What are the four major challenges for implementation of wireless LAN? $16 + 4$

Model Test Paper - Type 2

Wireless Communications

Max. Time: **3 Hours** Max. Marks: **100**

O Attempt F questions, taking at least WO questions each from Part A and Part B. All questions carry equal marks.

PA – A

Q1 (a) Derive an expression for received signal power using mobile point-to-point two-ray propagation model.

(b) How rapidly would the signal fade if a mobile operating at 800 MHz is in a vehicle moving at 100 kmph? If the vehicle is moving directly across a cell of 3-km radius, how long will it remain in that cell before it has to be handed off to the next cell? 12 + 8

Q2 (a) Describe the procedure for locating the cochannel cells in the first tier using a regular hexagonal pattern for cellular architecture. Illustrate the procedure for a cluster size of 12.

(b) A cellular system is designed with a total of 500 full duplex channels and 100 cell-sites, configured with a frequency reuse factor of 4. Determine the number of channels per cell, total number of channels in the system, and the minimum carrier-to-interference (C/I) ratio of the system in a mobile radio environment. 12 + 8

Q3 (a) What are the different types of digital modulation techniques employed for mobile communications? Explain the most preferred modulation technique.

(b) Distinguish between macroscopic and microscopic diversity. Describe the space diversity combining technique. 10 + 10

Q4 (a) Explain the concept of channel-sharing and channel-borrowing techniques.

(b) The transmitted power of a cell-site is increased by 6 dB. For the same minimum acceptable received signal power and all other factors remaining unchanged, compute the percentage increase in the coverage area. Assume the path-loss exponent value as 4. 10 + 10

PA – B

Q5 (a) Draw the GSM network architecture. Identify various interfaces used in its different entities.

(b) Explain how location of mobile users are tracked using various registers in GSM. 10 + 10

Q6 (a) What is meant by multi-carrier modulation? Describe the working principle of Orthogonal Frequency Division Multiple Access Scheme.

(b) Describe the format for digital voice channel used in US digital cellular TDMA system. 12 + 8

Q7 (a) How is GPRS technology different from GSM technology? Briefly describe the functions of those network elements in GPRS architecture which are different from GSM architecture.

(b) Consider a CDMA system which has a bandwidth of 1.25 MHz and transmits baseband data at 9.6 kbps rate. The system uses QPSK modulation and FEC convolution coding. The system requires E_b/N_o of 7 dB. Compute the number of subscribers that can be supported by the system. Assume a 3-sector antenna configuration with an effective gain of 2.6, power-control accuracy factor of 0.9, and an interference factor of 0.5. Ignore voice activity factor. 10 + 10

Q8 (a) What is the fundamental difference between Wi-Fi and Bluetooth technologies?

(b) Explain, with the help of a diagram, how a correspondent mobile node on a visiting network sends and receives IP packets to and from another mobile node. 10 + 10

Model Test Paper - Type 3

Wireless Communications

Max. Time: 3 Hours Max. Marks: 100

N TE All questions are compulsory. There are TW parts Part A has 10 questions of TW marks each and Part B has F R questions of 20 marks each.

PART – A

Q1 What are different techniques used to minimise the effects of fading?

Q2 What are the important parameters of the wireless communication network designed on cellular approach?

Q3 Differentiate between a high-gain omnidirectional antenna and a directional antenna having the same gain.

Q4 Define cell splitting.

Q5 What is the significance of using a pre-modulation low-pass filter with Gaussian characteristics in GMSK digital modulation technique?

Q6 What is meant by spectral efficiency of FDMA?

Q7 The basic TDMA frame structure of GSM cellular system comprises of 156.25 bits in a time slot, of which 40.25 bits are overhead (ignoring the 2 flag bits), compute the frame efficiency.

Q8 Why is it essential to implement power control on the reverse channel?

Q9 Suggest some measures which can be taken to address the security issues in wireless networks.

Q10 What are the main benefits of IEEE 802.11e WLAN standards?

PART – B

Q11 (a) Summarise the air-interface specification parameters of major digital cordless telephone standards.

(b) In a CDMA system, a user signal is transmitted with a chip interval of 814 ns. Find the code length that can be used to transmit video files at the high data rate of 384 ksymbols/s with a rate-matching reduction factor of 1.25. [16 + 4]

(OR)

(a) Briefly explain the evolution of the analog and digital cellular mobile systems.

(b) In IS-136 TDMA cellular system, the one-way allocated RF bandwidth is 12.5 MHz. The channel spacing is 30 kHz. There are 395 voice channels in the system. The TDMA frame duration is 40 ms, with 6 time slots per frame. The system offers an individual user data rate of 16.2 kbps in which the speech with error protection is @ 13 kbps. Compute the overall system efficiency. [12 + 8]

Q12 (a) Describe the methods to measure the cochannel interference at the cell-site and at the mobile unit.

(b) What is hand-off? Describe different types of hand-offs. 10 + 10

(OR)

(a) Distinguish between frequency-selective and time-selective fading. A mobile subscriber travels at a uniform speed of 60 kmph. Compute the time between fades if the mobile user uses a cellphone operating at 900 MHz.

(b) Explain briefly the characteristics of cell-site antennas and mobile antennas. 10 + 10

Q13 (a) Discuss the concept of frequency management including numbering the channels and grouping channels into subsets.

(b) Show that the frequency reuse factor, K for a cellular system is given by N/J, where N is the total number of channels available in the cellular network without frequency reuse and J is the average number of channels per cell. 10 + 10

(OR)

(a) Explain the concept of GSM superframe, multiframe, TDMA frame and time slot in GSM channel. Give suitable illustration for GSM frame hierarchy.

(b) In a cellular topology, an equilateral or a square geometry can also be used in addition to the regular hexagonal pattern. Given the same distance between the centre of the cell and the farthest perimeter points, compare the cell coverage areas among the three regular polygons, that is, equilateral triangle, square, and the regular hexagon. Comment on the advantages of using a regular hexagonal cell shape over the other two. 12 + 8

Q14 (a) Compare similarities and differences in the fundamental concepts of a DS-SS system versus FH-SS system.

(b) What are advantages and disadvantages of mobile TCP? How is mobile TCP different from traditional TCP? 10 + 10

(OR)

(a) In an omnidirectional CDMA cellular system, the required E_b/N_o is 10 dB. It is required to accommodate 250 number of users, each with a baseband data rate of 13 kbps. Determine the minimum channel chip rate of the spread spectrum sequence if the voice-activity factor of 0.4 is considered.

(b) What is wireless broadband WiMAX technology? How is it different from wireless LAN technology? 10 + 10

Model Test Paper - Type 4

Wireless Communications

Max. Time: **3 Hours** Max. Marks: **100**

O here are total H parts Part A, Part B, and Part C.

– Part A has 10 questions of WO marks each. Attempt ALL questions.

– Part B has F questions of 10 marks each. Attempt any FOU questions.

– Part C has H questions of 20 marks each. Attempt any WO questions.

PA – A Attempt all questions

Q1 (a) What is the value of propagation-path loss in dB at a distance of 50 km from the transmitter if it is given that the path loss is 110 dB at a distance of 25 km? Assume the mobile radio environment conditions.

(b) Why can free-space propagation model not be applied in a mobile radio environment?

(c) Distinguish between a cell and a cell-site.

(d) Under what circumstances, is static-channel assignment normally used?

(e) Distinguish between FDMA and FHMA systems.

(f) Compute the maximum throughput of a pure ALOHA network with a large number of subscribers and a transmission rate of 1 Mbps.

(g) How does a SIM card provide security against fraudulent use of GSM phone?

(h) Differentiate between the PN spreading codes and the orthogonal codes.

(i) Which methods can be used to temporarily or permanently disable an RFID tag?

(j) Suppose the mobile phone units are equipped with Bluetooth devices. What impact will it have on the piconet?

PA – B Attempt any FOU questions

Q2 Define Doppler spread. In an 800-MHz mobile radio system, the surrounding environment is relatively stationary and the mobile unit is traveling at a uniform speed of 80 kmph. What is the maximum Doppler spread and the baseband spectral bandwidth of the Doppler caused random signal? [4 + 6]

Q3 What are the tasks of PN sequences in multi-user CDMA systems? Why are Walsh codes preferred over *m*-sequences in practical CDMA systems? [5 + 5]

Q4 A cellular system has a total 500 duplex voice channels without frequency reuse. The service area is divided into 150 uniform cells. The required *C/I* value is 18 dB. Determine the cell-cluster size, number of cell clusters in the given service area, and maximum number of users at any instant in the service area. Assume the path-loss exponent as 3. [10]

Q5 State the conditions when hand-off is needed. Describe two-level hand-off algorithm. 5 + 5

Q6 What is the purpose of using RTS/CTS in CSMA/CA protocol? List the relative advantages and disadvantages of basic CSMA/CA with that of RTS/CTS protocols. 5 + 5

PART – C (Attempt any TW questions)

Q7 (a) The 'near–far interference' is a serious problem in a wireless cellular CDMA network. Explain its reason giving suitable example data.

(b) A mobile radio system operates in the 2.48-GHz ISM band. Calculate the absolute mean path loss at LOS free-space distance of 3 m and at a total distance of 22 m from the transmitter. Assume non-LOS propagation environment having a path-loss exponent of 3.5. 10 + 10

Q8 (a) A mobile receives 900 MHz transmissions while moving at a uniform speed for 10 seconds. The average fade duration for a signal level of 10 dB below the rms level is 1 millisecond. How far does the vehicle travel during the given interval? Assuming stationary environment, how many fades does the signal experience at the rms threshold value?

(b) What measures can be taken to reduce interference in a cellular system? 12 + 8

Q9 (a) Describe the functions of piconet and scatternet in Bluetooth architecture.

(b) Explain the concept and salient features of IEEE 802.11 MAC layer. 10 + 10

Hints/Answers to Model Test Papers

Model Test Paper – Type 1

Q1 (a) Refer Chapter 12 : Section 12.11.2
(b) Refer Chapter 9 : Section 9.1.1
Q2 (a) Refer Chapter 4 : Section 4.5 and 4.6
(b) Ans. 96 cell-sites
Q3 (a) Refer Chapter 6 : Section 6.3, 6.4 and 6.5
(b) Ans. 110.46 dB
Q4 (a) Refer Chapter 7 : Section 7.1
(b) Refer Chapter 8 : Section 8.8
Q5 (a) Refer Chapter 9 : Section 9.3.1, 9.3.2 and 9.3.4
(b) Ans. 51 approximately
Q6 (a) Refer Chapter 8 : Section 8.2, 8.3.1 and 8.4.1
(b) Refer Chapter 12 : Section 12.7
Q7 (a) Refer Chapter 1 : Section 1.6
(b) Refer Chapter 5 : Section 5.3
Q8 (a) Refer Chapter 11 : Section 11.4
(b) Refer Chapter 14 : Section 14.1.1

Model Test Paper – Type 2

Q1 (a) Refer Chapter 3 : Section 3.3.1
(b) Ans. 6.74 ms and 3.6 minutes
Q2 (a) Refer Chapter 4 : Section 4.5
(b) Ans. 125; 12500; 13.78 dB
Q3 (a) Refer Chapter 7 : Section 7.6.5 and 7.6.6
(b) Refer Chapter 7 : Section 7.8.2 and 7.8.3
Q4 (a) Refer Chapter 6 : Section 6.3.2, 6.3.3, 6.3.4 and 6.4.1
(b) Ans. 100%
Q5 (a) Refer Chapter 11: Section 11.1
(b) Refer Chapter 11: Section 11.2.6
Q6 (a) Refer Chapter 8 : Section 8.10.1
(b) Refer Chapter 10 : Section 10.8.3
Q7 (a) Refer Chapter 13 : Section 13.2.2
(b) Ans. 40 users
Q8 (a) Refer Chapter 14 : Section 14.1.2 and 14.3.1
(b) Refer Chapter 14 : Section 14.6.2

Model Test Paper – Type 3

Q7 Ans. 74.2%
Q11 (a) Refer Chapter 10 : Section 10.2.5
(b) Ans. 4
(OR)
(a) Refer Chapter 1 : Section 1.4.1, 1.4.2 and 1.4.3
(b) Ans. 76%
Q12 (a) Refer Chapter 4 : Section 4.7
(b) Refer Chapter 7 : Section 7.12.1
(OR)
(a) Refer Chapter 2 : Section 2.5.1 and 2.5.2; Ans. 10 ms
(b) Refer Chapter 5 : Section 5.1 and 5.2
Q13 (a) Refer Chapter 6 : Section 6.1 and 6.1.1
(b) Refer Chapter 4 : Section 4.4
(OR)
(a) Refer Chapter 11 : Section 11.5.1
(b) $1.3R^2$, $2R^2$, $2.6R^2$; $3.14R^2$
Q14 (a) Refer Chapter 12 : Section 12.1.2 and 12.1.3
(b) Refer Chapter 14 : Section 14.7
(OR)
(a) Ans. 1.3×10^7 Mcps
(b) Refer Chapter 14 : Section 14.4.1

Model Test Paper – Type 4

Q1 (a) Ans. 122 dB
(f) Ans. 184 kbps
Q2 Ans. 63 Hz and 126 Hz
Q3 Refer Chapter 12 : Section 12.2.1
Q4 Ans. 19; 7.895; 3947
Q5 Refer Chapter 14 : Section 14.4.1
Q6 Refer Chapter 14 : Section 14.4.1
Q7 (a) Refer Chapter 12 : Section 12.10.1
(b) Ans. 78 dB
Q8 (a) Ans. 443.1m and 2658.6 fades
(b) Refer Chapter 7 : Section 7.3
Q9 (a) Refer Chapter 14 : Section 14.3.1
(b) Refer Chapter 14 : Section 14.1.3

Answers to Analytical Problems

Chapter 2 Mobile Communication Engineering

P2.1 (a) 3.4 ms (b) 0.2 ms (c) 19.9 μs
P2.2 (a) 900,000,000.834 Hz and 899,999,999.166 Hz
(b) 900,000,004.167 Hz and 899,999,995.833 Hz
(c) 900,000,083.34 Hz and 899,999,916.66 Hz
P2.3 (a) 443.1 m (b) 2658.6 fades
P2.4 19.48 crossings/s and 5.1 ms
P2.5 (a) 20 times/s (b) 13.3 ms
P2.6 (a) 340 crossings (b) 9.28 ms (c) 0.543 ms
P2.7 (a) 6.74 ms (b) 3.6 minutes
P2.8 (a) 63.3 ns (b) 11.88
P2.9 13.65 crossings/s
P2.10 10 Hz
P2.11 34.7
P2.12 159.15 kHz

Chapter 3 The Propagation Models

P3.1 (a) 37.6 W (b) 266 W (c) 21.1 μW/m^2
P3.2 (a) +46 dBm (b) 91.52 dB (c) 0.028 μW
P3.3 (a) –47 dBm +17 dBW (b) –24.52 dBm
(c) –64.52 dBm
P3.4 7×10^{-9} W
P3.5 1.58×10^{-10} W
P3.6 891 m
P3.7 1×10^{-9} W
P3.8 110.46 dB
P3.9 (a) for $\gamma = 2$; –36 dBm; –44 dBm; –50 dBm; –56 dBm
(b) for $\gamma = 3$; –39 dBm; –51 dBm; –60 dBm; –69 dBm
(c) for $\gamma = 4$; –42 dBm; –58 dBm; –70 dBm; –82 dBm
(d) –54 dBm; –70 dBm; –82 dBm; –94 dBm
(e) –94.25 dBm; –107.93 dBm; –118.29 dBm; –128.64 dBm
P3.10 137.3 dB
P3.11 1.66 km and 933 m
P3.12 16 km
P3.13 29.8 km

Chapter 4 Principles of Cellular Communication

P4.1 3 and 21
P4.2 96 cells
P4.3 (a) $i = 3$, $j = 2$ (b) Yes (c) 15 km
P4.5 143 channels/cell, 50,050 channels/system
P4.6 (a) 30 km (b) 40 channels/cell
(c) 320 calls/cell
P4.8 (a) 44 channels/cell (b) 9 km
(c) 450 cells (d) 352 calls/cell
P4.9 (a) 10,656 each (b) 166 and 95
(c) 64 and 112 cells (d) 16,650 and 9,514
P4.10 (a) 165, 94 and 55 channels/cell respectively
(b) For $K = 4$, 160 voice channels and 5 control channels;
For $K = 7$, 5 cells of 92 voice channels each and 2 cells of 90 voice channels each;
6 cells of 3 control channels each and 1 cell of 2 control channels.
(c) For $K = 12$, 4 cells of 54 voice channels each and 8 cells of 53 voice channels each;
8 cells of 2 control channels each and 4 cells of 1 control channel each.

Chapter 5 Cellular Antenna System Design Considerations

P5.1 13.78 dB; 18.65 dB; 23.34 dB
P5.2 K = 19
P5.3 q = 4.6
P5.4 26.8 dB
P5.5 125 channels/cell; 12,500 channels; 13.78 dB
P5.6 24.76 dB, 29 dB, and 33 dB respectively
P5.7 By 7.5 dB
P5.8 17 dB and 16.79 dB respectively
P5.9 100 times
P5.10 By 26 dB

Chapter 6 Frequency Management and Channel Assignment

P6.1 (a) Minimum 71 channels/cell
(b) 7,150 channels/system
(c) Minimum 41 channels/cell; 4,166 channels/system
P6.2 (b) 992 links
P6.3 (a) 30 kHz one-way
(b) 790 voice and 42 set-up channels
(c) 829.5 MHz

Chapter 7 Cellular System Design Trade-offs

P7.3 100%
P7.4 12.5 W
P7.5 3.125 W

P7.6 (a) 2.5 km (b) 75% (c) 4 times
P7.7 (a) 360 channels (b) 51 approximately
P7.8 (a) One-half (b) +28 dBm
P7.9 (a) One-eighth (b) Two-times
(c) 1250 mW and 625 mW
P7.10 (a) 300 channels (b) 660 channels
(c) 1020 channels
P7.11 91.8 km^2
P7.14 5 taps

Chapter 8 Multiple Access Techniques

P8.1 800 channels
P8.2 13 μs and 7 bits
P8.3 (a) 0.577 ms (b) 4.615 ms
P8.4 (a) 156.25 bits (b) 1250 bits
(c) 40.25 bits (d) 74.24%
P8.5 (a) 820 kbps (b) 180 bits
P8.6 (a) 900 kbps (b) 360 bits
P8.7 33 μs and 92 bits/packet

Chapter 9 A Basic Cellular System

P9.1 (a) 2 km (b) 1 km
P9.2 16.67 μs
P9.3 (a) 10,656
(b) 166, 95 and 55 calls respectively
(c) 64, 112 and 192 cells respectively
(d) 16,600; 9,500; and 5,500 links respectively
P9.4 1062 users
P9.5 (a) 7 users (b) 111 users (c) 273 users
P9.6 System A – 43,071 users;
System B – 45,864 users;
System C – 48,462 users;
Total Capacity – 1,37,397 users
P9.7 (a) 32 cells; 95 channels/cell (b) 3040 users
(c) 84 Erlangs/cell (d) 2688 Erlangs
(e) 89,600 users
(f) 134 users/channel
P9.8 44.3%
P9.9 2558 calls/cell/hour
P9.10 1376 calls/cell/hour, and 739 calls/cell/hour
P9.11 (a) 1518; 769; and 403
(b) 1, 27, 512 users; 64, 596 users; 33, 852 users
P9.12 (a) 56 channels/cell
(b) 1376 Erlangs/cell and 172 Erlangs/km^2
(c) 36 calls/hour/cell
(d) 4.5 calls/hour/ km^2
P9.13 (a) 1026.7, 2223.3, 1620, 1106.7, 1273.3, 1260, 1086.7
(b) 0.9 calls/hour/subscriber
(c) 924, 2001, 1458, 996, 1146, 1134, 978
(d) 40, 78, 59, 43, 48, 48, 42
(e) 9597 subscribers
(f) 26.8 subscribers/channel
(g) 3 subscribers/km^2 approximately
(h) 287.9 Erlangs
(i) 2.78 calls/km^2
P9.14 (a) 23 cells (b) 832 channels
(c) 116 channels/cell
(d) 0.0714 channels/MHz/km^2
(e) 0.06426 Erlangs/MHz/km^2
P9.15 (a) 1000 users/cluster (b) 250 users/cell
(c) 0.7424 (d) 0.3375 bps/Hz/cell

Chapter 10 Wireless Communications Systems

P10.1 (a) −14 dBW, +16 dBm, 0.04 W
(b) No
(c) Class I : 3981 mW maximum and 6.3 mW minimum
Class II : 1585 mW maximum and 6.3 mW minimum
Class III : 63 mW maximum and 6.3 mW minimum
P10.2 (a) 12.5 km (b) 7.9 km
P10.3 9.225 km and 16.7 μs
P10.4 80.2%
P10.5 (a) 2.53 kbps (b) 506 bps
P10.6 (a) 48.6 kbps (b) 260 bits
(c) 40 ms (d) 80.25%
P10.7 (a) 48.6 kbps
(b) 1.8 kbps; 4.2 kbps; 0.9 kbps; 123.42 μs; yes; 13 kbps
P10.8 16.2 kbps
P10.9 (a) 80.25% (b) 8.1 kbps
P10.10 (a) 960 bits (b) 4800 bits
P10.11 (a) 149.6 dB (b) –115.46 dBm
(c) 169.6 dB; –135.46 dBm
P10.12 (a) 832 channels
(b) 118.8 channels/cell
(c) 624 user channels/cell
P10.13 (a) 24 dB (b) $K = 12$
(c) 2,080 users (d) 1783 users
P10.14 (a) 832 channels (b) 33 channels/MHz
(c) 208 and 118 channels/cell respectively
P10.15 (a) 870.09 MHz, 825.09 MHz
(b) 879.99 MHz, 834.99 MHz
(c) 893.97 MHz, 848.97 MHz
(d) 869.04 MHz, 824.04 MHz

P10.16 (a) 125 channels (b) 71 channels
(c) 42 channels
(d) For $K = 4$, 120 voice channels and 5 control channels;
For $K = 7$, 4 cells of 69 voice channels each and 3 cells of 68 voice channels each;
6 cells of 3 control channels each and 1 cell of 2 control channels;
For $K = 12$, 40 voice channels each; 8 cells of 2 control channels each and 4 cells of 1 control channel each.

Chapter 11 Global System for Mobile (GSM)

P11.1 (a) 124 channels (b) 40 channels/MHz
(c) 31 and 17.7 channels/cell respectively
P11.2 13 kbps, 9.8 kbps, 0.95 kbps and 10.1 kbps
P11.3 120 ms
P11.7 22.8 kbps
P11.8 (a) 33.854 kbps (b) 70.166% (c) 57.14%
P11.9 (a) 1000 users (b) 3.69 μs
(c) 4.616 ms (d) 4.616 ms
P11.10 27%

Chapter 12 CDMA Digital Cellular Standards (IS 95)

P12.1 +24 dBm
P12.2 52.1 μs
P12.3 (a) 7 users (b) 31 users
P12.5 31.25, 12.5, and 108 users/cell respectively
P12.6 (a) 13 users (b) 88 users
P12.8 (a) 130 Mcps (b) 52 Mcps
P12.9 (a) 51 Mcps (b) 20.4 Mcps
P12.10 21 users/sector
P12.11 172.8, 1.659 Mcps

Chapter 13 3G Digital Cellular Technology

P13.1 57.6 kbps
P13.2 68.4 kbps
P13.3 3.25×10^7 Mcps
P13.4 1.3×10^7 Mcps
P13.5 40 users
P13.6 0.307 bps/Hz
P13.7 480 ksymbols/s
P13.8 66.56 μs
P13.9 960 ksymbols/s
P13.10 2.4 kbps

Chapter 14 Emerging Wireless Network Technologies

P14.1 160 ms
P14.2 8 MB
P14.3 36 Mbps
P14.4 400 links
P14.5 20 messages
P14.6 –66.2 dBm
P14.7 250 m approximately
P14.8 Range reduces by about 12 m
P14.9 238 m and 670 m respectively
P14.10 52 μs; 0.814 μs; 64; 1.2288 MHz
P14.11 22 m
P14.12 105 dB

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