Wireless and Mobile Communication

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Dedicated to my husband Devang
and my daughters Parima and Jahnavee
– Upena Dalal

Dedicated to my respected
parents and lovely family
– Manoj K. Shukla

Features of

Theme of the Chapter

k mainly deals with the concepts of wireless digital communication. Though it is assumed that readers of th
familiar with the basic theory of communication, many required concepts are revised as a ready reference
tudents must be familiar with wireless communication systems—both conventional and latest. This chapte
ith a brief revision of the basics of communication and moves on to discuss wireless systems. It explains
digital wireless link with all the necessary blocks that form the basis for wireless systems. The chapter als
transmission rate, channel capacity, bandwidth, and signal-to-noise ratio parameters deciding the perfor
this link along with the types of signals useful for communication theory. It further explores the need fo
e of the best developments in wireless communications, which is possible only if the standards used toda
ess systems are known. Evolution of a system is linked with the previous systems, and the new system i
by analysing the problems of the previous systems and eliminating them. Hence, it is also necessary to know
opment scenario of the first to fourth generation systems. Once this background is provided and student
lying these from the root level of the wireless link, considering each and every stage of the wireless link
t of the theory and its application to the system can be correlated and the best solutions can be identified fo
here, *anytime* communication scenario.

Theme of the Chapter

Provides a glimpse of the topics that the readers are going to read and understand from a chapter

Sidebar

Interesting facts are presented as sidebars throughout every chapter to raise curiosity in readers.

Queuing of handoff request is a method to decrease the probability of forced termination of a mobile call due to lack of available channels.

The D layer disappears at night; hence, low frequencies like amplitude modulation (AM) can be received better at night compared to day.

Note: It is observed that an analog signal consumes less spectrum compared to its digital counterpart and hence requires lesser bandwidth, because the digital counterpart is the result of sharp transitions.

Note: Due to amplitude compression, logarithmic increase in quantization noise throughout the dynamic range of a sampled signal will keep the SQNR constant throughout this dynamic range.

Note: For 4 kHz voice digitization, the standard word size used is 8 bits. If an input analog signal is sampled 8000 times/s and each sample is given a code word that is 8 bits long, then the maximum transmission bit rate for telephony systems using PCM will be 64,000 bits/s.

Note

Highlights important statements so that readers don't miss them while reading

Box

Throughout each chapter, boxes provide a brief description of some key concepts and their significance

FRAUNHOFER DISTANCE

Shorter antennas with dimensions less than half the wavelength are treated as point antennas. Typical wavefronts are generated due to their radiations near the antenna and at far distance. The definition of near and far fields is different for them. In the free space propagation model, though long transmit antennas have physical dimensions larger than half the wavelength, they are considered too small compared to the distance. Hence, simplified assumption as a point antenna. Thus, it adds an isotropic source of radiated power [Fig. 3.6(a)]. Hence, near and far fields are distinguished in terms of Fraunhofer distance.

The far field is called *Fraunhofer region*, from the name of its inventor. It is defined as the region beyond the threshold value of the far-field distance d_f, which relates the largest linear dimension of the transmitter antenna aperture A_t and the carrier wavelength λ as given in Eq. (3.1). The Fraunhofer distance is given by

$$d_f = \frac{2A_t^2}{\lambda} \tag{3.1}$$

In addition, to be in the far-field region, $d_f >> A_t$ and $d_f >> \lambda$.

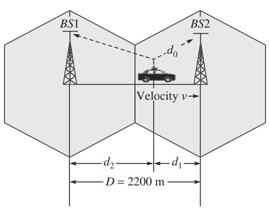

Fig. 2.13 Two base station scenario for handoff calculations

Illustrations

Important topics have been well-supported with suitable illustrations to allow easy visualization of difficult concepts. The book contains close to 370 self-explanatory illustrations.

the Book

Examples

Every chapter contains plenty of solved examples to demonstrate the applicability of the concepts discussed.

Example 5.2 The following are the readings for the measurement of quantization error in five consecutive samples. The number of quantization levels in the dynamic range of 2 V is eight.

Sample 1: 1.2 V
Sample 2: 1 V
Sample 3: 0.95 V
Sample 4: 1.41 V
Sample 5: 1.65 V

Find the quantization error in terms of its mean square value.

Solution If the dynamic range is 2 V, then the smallest step size will be 2/8 = 0.25 V.

Hence, the eight quantization l
0.5, 0.75, 1.0, 1.25, 1.50, 1.75, and 2
The measured samples will be a
ing quantization values:

Sample 1: 1.25 V quantization
Sample 2: 1.0 V quantization
Sample 3: 1.0 V quantization
Sample 4: 1.5 V quantization
Sample 5: 1.75 V quantization

Mean square error
= $[(0.05)^2 + 0 + (0.05)^2 + ($
= [0.0025 + 0.0025 + 0.00
= 0.0231/5 = 0.00462

Root mean square error = 0.0678

6.4 CHANNEL CODING AND TRADE-OFF

Error-correcting codes can be regarded
Figure 6.5 compares two curves depic
noise density ratio (E_b/N_o) (which is th
a digital signal representing the signal
scheme without coding; the other repre
Though channel coding is incorpora

Coverage

The book covers topics related to wireless communications as well as wireless networks. Topics such as source coding, channel coding, multiple access, modulation techniques, spread spectrum, diversity, and equalization are covered in great detail.

Introduces Newer Technologies

Addresses newer technologies such as the Next Generation Network (NGN) and Light Fidelity (Li-Fi) for the students to be able to appreciate the technological updates.

12.7 NEXT GENERATION NETWORK

In general, the Next Generation Network (NGN) is defined to be the communication network which may replace the current PSTN network around the globe which is employed for carrying voice, fax, modem signals, etc. In terms of definition, the NGN is essentially a managed IP-based which is, in fact, packet-switched network, ultimately employed for enabling a wide range of services including are VoIP, videoconferencing, Instant Messaging, e-mail, and all other kinds of packet-switched communication services. The standards for fixed NGN were developed by the ETSI technical committee TISPAN (Telecommunications and Internet converged Services and Protocols for Advanced Networking). In 2012, TISPAN was closed and work was transferred to ETSI Network Technologies which is known as TC NTECH. The IP Multimedia Subsystem (IMS) is a standardized NGN architecture for an Internet media-services capability defined by the European Telecommunications Standards Institute (ETSI) and the 3rd Generation Partnership Project (3GPP).

The ITU has also defined the term NGN in Recommendation Y.20
NGN is a packet-based network able to provide telecommunicatio
make use of multiple broadband, QoS-enabled transport technologies
related functions are independent from underlying transport-related te

12.8 LIGHT-FIDELITY

Light fidelity is popularly known as Li-Fi. It was researched by Professor Harald Hass in 2011. He explained to the people in a talk that how data may be transferred using light. The data transfer speed of Li-Fi is estimated to be much higher compared to Wi-Fi due to the reason that the bandwidth of visible frequency is approximately 10,000 times greater than that of radio frequencies. Figure 12.26 demonstrates the visual presentation of data transfer speed of Wi-Fi and Li-Fi technologies in symbolic manner.

Li-Fi is based on concept of VLC (visual light communication) technique. To understand the concept of VLC, let us consider that a constant power is provided to a LED as presented in Fig. 12.27 and Fig. 12.28. In this case, a constant output power is achieved at the output also. However, output will be fluctuated if there is any variation in input. As soon as input changes, the output of light also changes which will help in data transfer.

EXERCISES

Multiple-choice Questions

3.1 The electric field of an EM wave at a point in free space is in the positive *Y* direction and the magnetic field is in the positive *X* direction. The direction of power flow will be in the
(a) positive *X* direction (c) positive *Z* direction
(b) positive *Y* direction (d) negative *Z* direction

3.2 In isotropic propagation, if f_c is the critical frequency and θ is the angle of incidence at the ionosphere measured with respect to the normal, the MUF equals
(a) $f_c\cos\theta$ (c) $f_c\sec\theta$
(b) $\sqrt{f_c}\cos\theta$ (d) $\sqrt{f_c}\sec\theta$

3.3 The skip distance is

3.7 The virtual height of an ionospheric layer is
(a) more than the height a wave actually reach
(b) less than the height a wave actually reache
(c) the same as the height a wave actually reac
(d) none of these

3.8 The ionosphere roughly extends from
(a) 50 km to several earth radii
(b) 50 km to 80 km
(c) 50 km to 400 km
(d) 50 km to 150 km

3.9 The total noise of a satellite earth station rece system consists of
(a) sky noise
(b) parametric amplifier noise
(c) antenna and feeder noise

Exercises

Has a rich set of end-chapter exercises with close to 250 review questions, more than 200 Multiple-choice Questions, and 100 unsolved problems.

Preface

A communication system is used for transmitting information or data from one point to another. Earlier communication systems, such as telegraphy and telephony, were wired systems which transmitted information through wires. A major breakthrough in the field came with the advent of wireless technology, which uses radio waves to transmit data, as opposed to wires, to carry signals over the communication path. Wireless communication deals with the principles, techniques, and analytical tools underlying wireless systems, other emerging systems, and standards in the field.

Wireless communication has become the fastest growing segment of the telecommunications industry and has led to exciting technological advances over the last few decades. This has radically changed the way people communicate across the world. Initially, wireless communication was mainly used in military applications. With the commercialization of wireless systems and resource-sharing networks, communicating with people over mobile phones even on the move has become easy, and access to social media and applications provided by mobile service providers has made it much more convenient. In the future, the telecommunications industry is set to become all-wireless with an 'anywhere, anytime, and for anybody' communications scenario. Users will have a single and unique identification number—universal telecommunication number (UTN)—which would enable users to receive as well as make calls from any terminal on any network. The stage is set for 4G, the fourth generation of mobile telephones, which in addition to the usual services would also enable streaming multimedia, HDTV content, digital video broadcasting (DVB), and ultra-broadband Internet access. Mobile companies are already aiming for 5G technology, which has been visualized as the convergence of network access technologies.

ABOUT THE BOOK

This book is primarily designed for undergraduate students of electronics and communications engineering as well as computer engineering, and is suitable for courses on mobile communication, wireless communication, and mobile networks. Basic knowledge of the concepts of communication, signal processing, and probability theory is assumed to be a pre-requisite. A lucid approach, both in terms of language and content, has been adopted throughout the text. Beginning with the fundamental concepts of wireless communication, the book comprehensively covers the various aspects of wireless systems.

As the majority of wireless communication systems today are completely digital, this text focuses only on all the aspects of digital communication in the context of wireless channels, and analog methods have been completely omitted from the text.

The book is divided into four parts which represent the four important aspects of practical wireless systems—Wireless Communication Prerequisites (Chapters 1 and 2), Wireless Channels and Modelling (Chapters 3 and 4), Wireless Communication Techniques (Chapters 5–9), and Wireless Networks (Chapters 10–12). Each chapter begins with a theme and key topics, and gradually explores concepts through detailed explanations and illustrations. A large variety of solved examples have been added to elucidate the application of the theory covered in each chapter. Review questions, multiple-choice questions, and numerical exercises add value to the rich content of the book. The book also contains appendices on the additional topics associated with the subject.

CONTENTS AND COVERAGE

Chapter 1 is the introductory chapter, which describes the basic terminology associated with wireless communication in the present scenario and trends in wireless systems. It discusses various types of wireless systems in terms of major advancements identified in different generations and also classified according to range such as short range and long range systems. The chapter helps to develop a basic understanding of the subject, so that concepts in later chapters can be understood easily.

Chapter 2 is related to the infrastructure development of cell-based wireless communication in multi-user environments. An understanding of cell theory is necessary for deciding the size of the cell, locating the transmitter in a cell, and splitting the cell to cover a higher population density. Frequency reuse is the key concept to utilize the available channels efficiently, but it leads to co-channel interference. By utilizing cellular theory, all these problems can be solved. The chapter briefly discusses traffic engineering as well.

Chapter 3 describes radio propagation over a wireless channel. Starting from the free-space propagation model, different types of long-distance radio propagations are discussed in the chapter. Path loss model and multipath effect are also explained; these are necessary to understand the behaviour of the channel in certain frequency ranges. It also delves into the different types of fading effects (such as delay spread and Doppler effect), which are very common in the multipath environment.

Chapter 4 covers the different channel models represented in their mathematical forms. The chapter discusses popular channel models such as the Rayleigh model, the Rician model, and Nakagami, which are all characterized by their probability density functions (PDFs). It also covers popular urban models such as the Okumura and Hata models.

This chapter ends with discussion of underwater wireless communication channel modelling where RF and related waves cannot be used for communication.

Chapter 5 mainly deals with the concept of source coding and waveform coding. Most real-time signals are analog in nature. Beginning from the digitization of analog signals, further processing must be applied to the source signal to compress or convert it into a standard format. The chapter describes the analog-to-digital conversion process, as well as the errors which result from the conversion, for example, aliasing. Digital transmission formats, special voice coders for low bit rate signals, and data compression methods are also discussed in this chapter.

Chapter 6 describes error handling over a noisy channel. As the wireless channel is more susceptible to noise and multipath effects, error-correcting codes are required. Hence, in this chapter, we describe most of the error-correcting schemes with their error-correction capabilities. The chapter also demonstrates the latest developments, such as Turbo codes, which are increasingly becoming popular as they approach Shannon's limit for bit error rate (BER) performance.

Chapter 7 helps in the understanding of all the basic single- and multi-carrier digital modulation schemes along with their mathematical representations, block diagrams, constellation diagrams, and other important parameters. The chapter comprises conventional methods such as Amplitude shift keying (ASK), frequency shift keying (FSK), binary phase shift keying (BPSK), M-PSK, and quadrature amplitude modulation (QAM) as well as the modified versions of the conventional modulation schemes, such as differential PSK (DPSK), offset keyed quadrature PSK (OKQPSK), minimum shift keying (MSK), Gaussian MSK (GMSK), and M-FSK. Finally, the chapter explains spread spectrum modulation (SSM) and orthogonal frequency division multiplexing schemes, which are especially suitable for the 3G and 4G systems, and elucidates how these techniques are superior to conventional digital modulation techniques.

Chapter 8 illustrates the diversity techniques, equalization methods, and channel estimation to mitigate channel effects. Most of these techniques are important at the receiver's end and help improve the quality of signal reception. An understanding of these concepts is very important because, as a result of these techniques, phase ambiguity due to multipath, frequency-dependent effects, or fading effects can be considerably reduced at the receiver side, and BER performance can be improved. Multiple input, multiple output (MIMO), the latest diversity-based technique, which is based on spatial diversity, is also covered in this chapter.

Until Chapter 8, all the basic theories and fundamentals for establishing a single wireless digital link are described. From Chapter 9 onwards, the focus shifts to the multi-user system environment.

Chapter 9 is related to multiple access techniques. There are numerous ways in which multiple users are allowed to access the available wireless channels on a sharing basis, so that all the users can communicate successfully without any partiality and without interference from one another. This chapter throws light on some of the schemes for multi-user environments in which an individual user's information is transmitted independently, such as FDMA, TDMA, CDMA, OFDMA, Single Carrier FDMA (SC-FDMA), Interleave Division Multiple Access (IDMA), and space division multiple access (SDMA). For packet radio systems, random access schemes such as ALOHA, slotted ALOHA, and carrier sense multiple access with collision detection (CSMA/CD) are used for sharing packets over a channel, rather than complete information transmission at a time.

Chapter 10 summarizes the concepts of conventional networking and its applications in a wireless networking environment. Starting with the OSI reference model and layered concept of protocol design, the chapter discusses TCP/IP protocol. It also describes the basic constraints of networking and gives some basic solutions, such as MAC scenario, routing protocols, and transport scenario, along with their applications, and highlights the importance of mobile computing.

The last two chapters provide an introduction to all the existing wireless digital systems, which have been developed on the basis of certain standards and protocols.

Chapter 11 describes the infrastructure-based/cell-based networks which are established permanently and support mobility, such as GSM, CDMA, UMTS, WLL, and LTE.

Chapter 12 describes special categories of wireless systems like ad-hoc networks (e.g., bluetooth), ad-hoc networks with the support of cellular concept, and networks mainly designed for data access or transfer (e.g., Wi-fi and WiMAX). The chapter also expounds on Zigbee, which is a special protocol for the wireless sensor network, and UWB, which is used for ultra-high speed indoor communication. The chapter ends with discussion of Next Generation Network (NGN), considered to be a promising alternative to the existing public switched telephone network (PSTN) and Light Fidelity (Li-Fi), a bidirectional, high speed futuristic technology in wirless communication.

Appendices A to *G* deal with linear systems theory, algebra for the linear system, probability theory, DSP fundamentals applied to OFDM processing, satellite communication aspects, and Erlang and Poisson traffic tables, and Interleaver.

ONLINE RESOURCES

To aid the faculty and students using this book, additional resources are available at www.india.oup.com/orcs/9780199475001.

For Faculty
Solutions manual
PowerPoint presentations
For Students
MATLAB codes

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Suggestions for improving the presentation and contents of the book can be sent to the publisher through their website www.oup.co.in or to the authors at upena_dalal@yahoo.com and manojkrshukla@gmail.com.

Upena Dalal
Manoj K. Shukla

Detailed Contents

List of Symbols

$\alpha(t)$ = Rayleigh amplitudes of the multipath signals
α (f) = absorption coefficient
δ = handoff margin
ε = permittivity of free space (capacitance per unit length measured in farads/meter)
ε_0 = dielectric constant of vacuum
ε_r = relative dielectric constant of the earth
$\phi(t)$ = different phases due to different delays of multipath signals
Φ = phase of noise vector
λ = wavelength (of carrier normally)
λ = call arrival rate
μ = permeability of free space (inductance per unit length measured in henries/meter)
μ = mean number of call arrivals
η = aperture efficiency
η_{fdma} = spectral efficiency of FDMA scheme
$\eta_{fdma\text{-}tdma}$ = spectral efficiency of FDMA-TDMA scheme
η_{tdma} = spectral efficiency of TDMA scheme
σ = standard deviation
σ_g = conductivity of ground
σ_x = standard deviation from the mean value
θ = phase variable
θ = directions variable
θ_{err} = phase error
θ_{max} = maximum allowable phase margin
τ = time variable for autocorrelation function
τ_{max} = maximum delay spread on multipath channel
τ_1 = delay between received multipath signals
ω = angular frequency of a signal
ω_c = angular carrier frequency
π_k = interleaver sequence for k th user
$\xi_k(j)$ = distortion including interference and noise
$\ddot{o}(t)$ = signal attenuation in underwater environment
a's = filter coefficients of an FIR filter
a = sea water coefficient
a_n = symbol for transmission
a_n = value of nth chip in PN code
A = traffic in erlangs
A_0 = peak amplitude
A_e = effective receiving aperture area of antenna
A_m = amplitude of the signalling element
A_r = actual receiving aperture area of antenna
$A(d, f)$ = attenuation over a distance d for a signal with frequency f
a(f) = absorption coefficient
Atm = transmission anomaly
$A_p(t)$ = time-varying path amplitude
B = spread bandwidth
B = busy period in ISMA scheme
B = grade of service
B_c = coherence bandwidth
B_d = Doppler spread
b_g = number of bits in each guard interval
B_j = interference signal bandwidth
B_m = message signal bandwidth
b_{OH} = overhead bits in TDMA frame
b_p = number of bits in each slot preamble
b_r = number of overhead bits per reference burst
b_T = total bits in TDMA frame
B_s = average bits per symbol
B_s = PN signal bandwidth
c = average number of calls in period of observation
c = speed of sound
c_k = code rate for k th user
c_n = amplitude of nth reflected component
$c(t)$ = PN sequence (subscript t or r represents transmitter PN sequence and received PN sequence)
C = channel capacity in bits/s
C = total effective number of duplex channels available in cell area
$C_{auto}(k)$ = autocorrelation function for speech for kth sample
C (ʹI, t) = channel impulse response for a time-varying multipath underwater acoustic channel

d = distance between isotropic source and receiver antenna
d = distance between two antenna elements
d_1 = initial period of the busy period in ISMA
d_2 = processing delay within busy period after packet duration in ISMA
d_f = far-field distance
d_k = input data sequence for k th user
d_{min} = minimum weight
d_0 = close-in distance
$d(U,V)$ = hamming distance between the codes U and V
A_t = largest physical dimension of the transmitter antenna
D = frequency reuse distance
$e(n)$ = estimation or equalization error signal in discrete form
E = electric field strength w.r.t. transmitter
E_b = bit energy
E_{ms} = mean square error
E_o = field strength while propagation in free space
E_s = signal energy dissipated in time T
$E\left(r(j)\right)$ = estimation of received signal for j th chip
f = frequency variable
f_1 = spacing between consecutive hopping frequencies in FHSS
f_1, f_2, etc. = subcarrier frequencies in OFDM
$f_1, ..., f_N$ = frequency groups within a cluster f_1 for cell 1 and so on
f_c = carrier frequency
f_d = Doppler spread
f_{max} = highest frequency content of analog in
f_o = centre frequency of the PN signal spectrum
f_s = sampling frequency or Nyquist frequency
$f(t)$ = basic pulse function
$f(x,y)$ = image function (2D)
Δf = carrier spacing between orthogonal subcarriers in OFDM
F = flux density
F = fade margin
$F(x)$ = companding function to compress x integer value
g (t) = impulse response of transmit filter
G = offered traffic
G_r = gain of receiving antenna
G_t = gain of transmitting antenna
$G(\theta)$ = gain of antenna at angle θ
$R_S(f)$ = PSD function
H = average call holding time
h_R = receive antenna height
h_T = transmit antenna height
$h(n)$ = channel impulse response (discrete form)
$h(t)$ = channel impulse response
$h(x)$ = polynomial for LFSR design
$h_e(n)$ = estimated channel impulse response in discrete form
h_k = channel coefficient for k th user
$H(\omega)$ = a channel impulse response in frequency domain
H = complex N × M matrix representing the MIMO channel impulse response, subscripts LOS, NLOS represents corresponding components
H or $H(x)$ = entropy (in outcome x)
$h_{antenna}$ = height of antenna
$i(t)$ = current as a function of time t (variable current)
$i(t)$ = interference signal
$I(t)$ or I = in-phase component of the received modulated (complex) signal
I = mean value of exponentially distributed Poisson arrivals
I_i = interference power of ith cell
I_{int} = total number of interfering cochannel cells
I_o = interference density (per bit)
$I(x_i$ or $x_j)$ = information content in outcome x_i or x_j
$'I_p(t)$ = time-varying path delay
J = total interference power in CDMA
J = chip number
k = input bits applied to channel coder
k = ratio of message bit duration to chip duration (hops per message bit)
k = number of channels within a group
k = occupied trunks
k = power expansion factor
K = frequency multiplier in FHSS
K = Rician factor
K = number of users in DS-CDMA
K = variable for smart antenna elements in general
Ks = spreading factor

l = number of mobile terminals
L = dimension of vector quantization, number of samples in a block or vector
L = maximum resolvable paths for CDMA
L = total number of duplex channels available to the operator
L = number of multipath components received at each antenna element from 1 mobiles
m = shape factor or gamma distribution (Nakagami model)
$m(t)$ = transmitted message signal
$m'(t)$ = received message signal
M = number of quantization levels, number of signalling elements
M = number of hopping frequencies (in FHSS), number of hopping time slots (in THSS)
M = number of times cluster is repeated
M = a number of receiving antennas for MIMO
$M(f)$ = spectrum of message signal
n = number of bits per quantization level (i.e. number of bits per sample)
n = number of bits/symbol
n = output bits from channel coder
n = number of users/slots per channel in TDMA
n = path loss exponent
n = symbol index count
N = total noise power
N = maximum number of significant reflected components to model the channel
N = number of users in CDMA
N = one period comprising number of samples, IFFT bin size in OFDM
N = number of trunks
N = cluster size in cell theory
N = number of transmitting antennas for MIMO
N_c = number of chips in a full period of PN code
N_c = number of subcarriers in OFDM bandwidth
N_c = total number of carriers in FDM scheme
N_t = number of data slots per TDMA frame
N_o = noise power spectral density
N_r = number of reference bursts per TDMA frame
N_s = total number of slots in TDMA frame
$N_{shipping}$ = shipping noise
$N_{surface-wind}$ = surface noise due to wind
$N_{thermal}$ = thermal noise
N_{turb} = turbulent noise
N_u = number of users supported in FDM
N_{us} = number of users per sector
p = probability of occurrence of information in binary symmetric source
p = number of shift registers in an ML sequence generator
p = persistency
Pl = path loss
p_p = prime number to generate twin prime sequences
P = power (subscript differentiating 's' signal or 'n' noise power)
P_B = bit error probability
P_o = power radiated by test antenna
P_r = received power at the receiver front end
Pr_{handoff} = power level at which handoff is made
$Pr_{\text{min usable}}$ = minimum usable power for acceptable voice quality
P_t = (isotropic source) transmitted power
PG = processing gain
$P(\theta)$ = power radiated in a direction with angle θ
$P(t)$ = power as a function of time t (variable power)
$p(x)$ = probability function
$P(x_i \text{ or } x_j)$ = probability of outcome x_i or x_j
$P(Y/S^{(m)})$ = probability of maximum likelihood with *mth* sequence when Y sequence is received
$\overline{P}$ = local mean power
$\overline{\overline{P}}$ = area mean power
q = integer deciding the q-r sequences and Hall sequences
q_e = quantization error
$Q(t)$ or Q = quadrature component of the received modulated (complex) signal
r = integer deciding the sequences
Δr = resolution, step size in quantizer
$r(j)$ = received signal for j th chip

$r(n)$ = received signal (discrete form)
$r(t)$ = received time varying signal
R = resistance
R = data rate or information rate in bits/s
R = cell radius
R_b = bit rate in bits/s
R_c = ground reflection coefficient
R_d = data redundancy
R_{max} = maximum cross correlation
R_T, R_R = correlation matrices of transmitter and receiver, respectively
$R(D)$ = rate of vector quantization in bits per sample (for given distortion D)
$R_c(\tau)$ = autocorrelation function for PN codes
s = number of sectors in a cell
s = spreading factor
s_1, s_2, etc. = symbols, split from OFDM symbol (frame) to assign the carrier
s (t) = output signal
$s(t)$ = transmitted time varying signal
$s(t)$ = spread baseband at the transmitter
$s'(t)$ = spread baseband at the receiver
S = average signal power
$S(t)$ = frequency spectrum (signal in frequency domain)
$s(n)$ = transmitted signal (discrete form)
$s(k)$ = kth speech sample
S_k = kth frequency sample in the speech spectra
$S^{(m)}$ = mth possible sequence
$s_k(t)$ = kth subcarrier in multicarrier system
$S_q(t)$ = quantized speech signal
t = time variable
t_c = error-correcting capability of the code
t_{chip} = chip duration
t_d = error-detecting capability of the code
t_{di} = delay over the ith channel path in case of multipath reception for SSM system
t_m = message bit duration for spread spectrum system
T or T_o = period of a periodic signal (time for one cycle)
T = traffic observation duration
T_b = One bit interval (pulse duration)
T_{code} = period of PN code
T_f = frame duration containing all time slots
T_g = guard interval between consecutive OFDM symbols
T_m = maximum excess delay over channel
T_{mc} = (multicarrier system) symbol period
T_o = coherent time
T_{rms} = RMS delay spread
T_s = OFDM symbol period
T_{sc} = (single carrier system) symbol period
T^k = shift operator
ΔT_s = sampling interval
v = velocity of mobile
v = velocity of light/velocity of electromagnetic waves
v_f = voice activity factor
V = voltage (subscript differentiate with ‘s’ signal or ‘n’ noise voltage)
Var = variance
$V(t)$ = voltage as a function of time t (with appropriate identity subscript)
w = variable for channel taps or weights
w = wind speed
$w(n)$ = white noise (discrete form)
$w(t)$ = white Gaussian noise varying with time
W = bandwidth
$W_{channel}$ = available bandwidth
W_{guard} = guard band
W_{signal} = per user bandwidth
$W(U)$ = Hamming weight for code word U or V
x = actual number of call arrivals
X_k = interleaved data
Y = received sequence
z = number of standard deviations
z = threshold level (or noise threshold level)

Acronyms

ANA	Application Network Adaptations
APP	A Posteriori Probability
BER	Bit Error Rate
BS	Base Station
CDMA	Code-Division Multiple-Access
CDS	Channel Dependent Scheduling
CIR	Channel Impulse Response
CP	Cyclic Prefix
DEC	Decoder
DFT	Discrete Fourier Transform
DSL	Digital Subscriber Line
DSP	Digital Signal Processing
DWDM	Dense Wavelength-Division Multiplexing
ETSI	European Telecommunications Standards Institute
FCC	Federal Communication Commission
FDE	Frequency Domain Equalization
FEC	Forward Error Correction
FFT	Fast Fourier Transform
GF	Galois Field
IDFT	Inverse Discrete Fourier Transform
IDMA	Interleave-Division Multiple-Access
IFDMA	Interleaved FDMA
IFFT	Inverse Fast Fourier Transform
IMS	IP Multimedia Subsystem
IP	Internet Protocol
ITU	International Telecommunication Union
LCU	Loop-Count with User
LFDMA	Distributed Subcarrier FDMA
LFDMA	Localized Subcarrier FDMA
Li Fi	Light Fidelity
LLR	Log-Likelihood Ratio
MAI	Multiple Access Interference
MC CDMA	Multi Carrier CDMA
MIMO	Multiple Input Multiple Output
MRI	Master Random Interleaver
MUD	Multiuser Detection
MUX	Multiplexer
NGN	Next Generation Network
NNA	Network to Network Adaptations
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency-Division Multiple-Access
PAPR	Peak to Average Power Ratio
PI	Prime Interleaver
PSD	Power Spectral Density
PSE	Primary Signal Estimator
PSTN	Public Switched Telephone Network
RI	Random Interleaver
RPS	Random Position Setter
RTP	Real Time Protocol
SC-FDMA	Single-Carrier Frequency-Division Multiple Access
SCTP	Stream Control Transport Protocol
SIP	Session Initial Protocol
SS7	Signaling System 7
TBI	Tree Based Interleaver
TD-SCDMA	Time Division-Synchronous CDMA
TISPAN	Telecommunications and Internet converged Services and Protocols for Advanced Networking
UDP	User Datagram Protocol
UNA	User Network Adaptations
UW	Under Water
UWA	Under Water Acoustic
UWB	Ultra Wide Band
VoIP	Voice over Internet Protocol
WCDMA	Wideband CDMA
Wi Fi	Wireless Fidelity

PART 1
Wireless Communication Prerequisites

Signal processing and basic principles of communication are the basis for wireless communication, and cellular theory forms the basis for wireless systems. Part 1 of this book consists of two chapters that provide the initial spark by introducing the basic terminologies and describing the existing wireless systems all over the world, along with their development phases called generations and major features.

In this part, the readers will also be introduced to the basic necessity of user's mobility for "anytime anywhere seamless connectivity", and that is cellular structure and associated issues.

This way, the section creates a bigger picture of the matters to be touched upon for the development of the complete wireless scenario.

1 Fundamentals and Present Scenario

Theme of the Chapter

This book mainly deals with the concepts of wireless digital communication. Though it is assumed that readers of the book are familiar with the basic theory of communication, many required concepts are revised as a ready reference. Today's students must be familiar with wireless communication systems—both conventional and latest. This chapter begins with a brief revision of the basics of communication and moves on to discuss wireless systems. It explains a complete digital wireless link with all the necessary blocks that form the basis for wireless systems. The chapter also discusses transmission rate, channel capacity, bandwidth, and signal-to-noise ratio parameters deciding the performance of this link along with the types of signals useful for communication theory. It further explores the need for and scope of the best developments in wireless communications, which is possible only if the standards used today for wireless systems are known. Evolution of a system is linked with the previous systems, and the new system is designed by analysing the problems of the previous systems and eliminating them. Hence, it is also necessary to know the development scenario of the first to fourth generation systems. Once this background is provided and students start studying these from the root level of the wireless link, considering each and every stage of the wireless link, every part of the theory and its application to the system can be correlated and the best solutions can be identified for the *anywhere*, *anytime* communication scenario.

Key Topics

- Fundamental terms of communication
- Wireless communication link model
- Bandwidth and signal-to-noise ratio
- Types of signals
- Types of communication systems
- Wired versus wireless communication
- Types of wireless systems
- Existing technologies and requirements
- Evolution of wireless systems
- First- to fourth-generation wireless systems
- Licensed and unlicensed band communication
- Spectrum policies

1.1 FUNDAMENTAL TERMS OF COMMUNICATION

Wireless communication is a diverse field and its study requires a basic knowledge of many other fields. The overall model of the learning system for wireless communication is shown in Fig. 1.1.

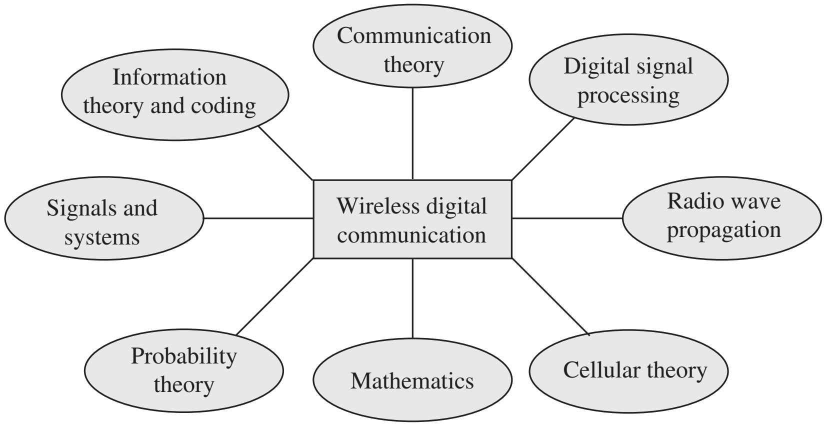


Fig. 1.1 Interdisciplinary learning model for wireless communication

Readers may be aware about many of these fields and may even be familiar with the basic theory of communications. However, in this chapter, we will brush up on all the fundamentals before we get into the details of wireless communication. This section will recapitulate the most frequently used terms in this subject.

Information Communication systems convey messages that originate from information sources. The information may be analog or digital, and accordingly, the communication system can be classified as an analog or a digital system. The sine wave is the fundamental analog information signal. A pure sine wave can be represented by three parameters—peak amplitude (A_0), frequency (f), and phase (θ)—in the form

$$s(t) = A_0 \sin(\omega t + \theta)$$

where $\omega = 2\pi f$, the angular frequency. The analog information may be voice or video (or real-time signals).

Information is to be transmitted by a sender and is to be received and interpreted by a receiver. If the information is in analog form, a conversion is required to process and transmit it into digital form. The smallest unit representing the digital form is a bit, which is a pulse. Digital information may be converted into words, groups of words (frames), code symbols, or any other prearranged units of bits. When no interpretation is applied, these units are called *data*, which may be a raw bit stream. When they are received and interpreted at the other end, they become *information*, which is conveyed.

For binary digital systems, the data or information transmission rate is measured in bits per second. If additional bits are added (for special purpose) to the required data, the efficiency of information transmission reduces. It must be understood that no real information is conveyed by a redundant message, but redundancy is not wasteful under all conditions, especially where error handling is concerned (which will be discussed shortly). In short, a set of information or data with respect to time is the time domain *input signal* for a system, whose frequency contents can be observed in the frequency domain by observing the spectrum. Information theory and the mathematical aspects of measurement of information are discussed in Chapter 5.

Antenna size and carrier follow inverse relationship; hence, for portable sizes of antenna, very high frequencies are used in wireless systems.

The transmitter and receiver systems are connected through a channel. These systems process the input signal in various ways to ensure proper communication. One of the important processes is *modulation*, with which the term *carrier* is associated.

Modulation This is the process by which a signal is transformed into waveforms that are compatible with the characteristics of the channel. Modulation may be of two types: analog and digital. In analog modulation, analog signal is modulated by a carrier while digital modulation is the process by which pulses are modulated into the required digital form or modulated by a carrier. These modulated waveforms usually take the form of shaped pulses [ideally the shape of a sinc function, which is $\sin(x)/x$), in the frequency domain]. However, in the case of *digital band-pass modulation*, the shaped pulses modulate a sinusoid called a carrier wave, or simply a carrier. For radio transmission, the carrier is converted into an electromagnetic (EM) field through an antenna for propagation to the desired destination.

Carrier The transmission of EM fields through space is accomplished using antennas. The size of the antenna depends upon the wavelength λ and the application. The antennas used for cellular telephones are typically small. Wavelength and frequency are related as $c = f\lambda$, where c is the speed of the EM wave in free space. Thus, antenna dimensions indirectly decide the frequency an antenna can transmit. A very large antenna would be required for sending a baseband signal of a very low frequency. To transmit a 3 kHz signal or voice signal through space, without carrier wave modulation, an antenna that spans 15 miles would be required. If the baseband information is first modulated on a high-frequency carrier (e.g., 900 MHz), then it would require an antenna with a diameter of only about 8 cm. Hence, for all portable applications, radio frequency (RF) conversion is necessary.

Another advantage of modulation with a carrier is the multi-user environment. If more than one signal or user utilizes a single channel, modulation with different carriers or the same carrier may be used to separate the different signals (these techniques are explained in Chapter 9). The reception will be based on the tuning of the carriers. Systematic allocation of frequency bands is possible due to the fixed bandwidth and dedicated allocation of carriers. Some modulations can be used to minimize the effects of interference. Such modulation schemes require a transmission bandwidth that is much greater than the minimum bandwidth that would be required by the message (wideband communication). Bandwidth concepts are discussed in Section 1.3.

Transmitter A transmitter performs various functions to make a source signal suitable for transmission. Examples of such functions are converting a non-electrical form of signal into an electrical signal, restricting the range of frequencies, compressing the amplitude ranges, and modulating the signal as per requirements. Not much processing is required in *baseband communication* or *carrierless communication*, such as the local loop wire telephony, as the mouthpiece of the handset gives analog electrical signals that can be directly transmitted for short distances on the wired lines. However, in long-distance communication, a transmitter is required to process, possibly encode, and to modulate the incoming information to make it suitable for transmission over the desired channel and subsequent reception. This is known as *broadband communication*. Eventually, in this type of transmitter, the information modulates the carrier, that is, the information is systematically superimposed on a comparatively high-frequency sine wave. RF upconversion may be followed by the modulator stage, especially for a wireless link, and then the power amplifier stage completes the transmitter part. The signal becomes ready for transmission through an antenna.

In wireless communication, the final form of transmission is always analog, irrespective of whether the modulation is analog or digital.

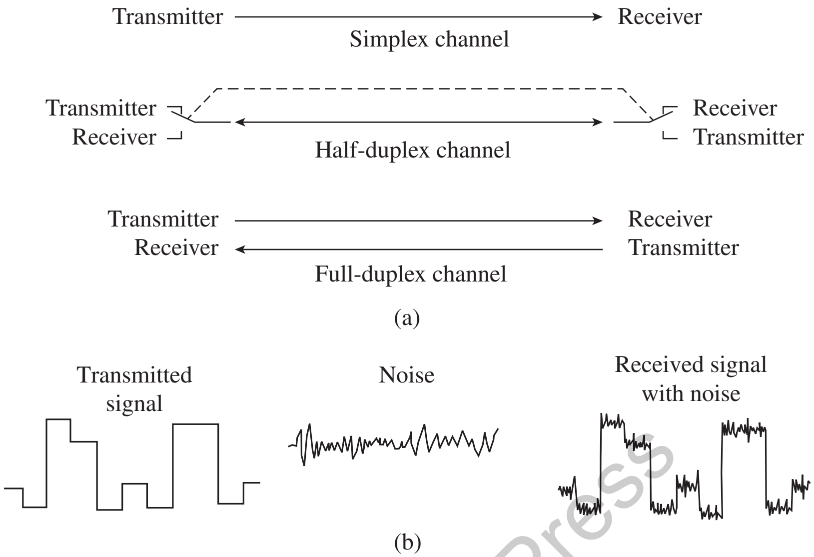


Fig. 1.2 (a) Channel types (b) Addition of noise to signal

Channel It should be noted that the term *channel* is often used to refer to the frequency range allocated to a particular service for transmission, such as a television channel (the allowable carrier bandwidth with modulation); however, in general, a channel is a medium through which a signal propagates towards its receiver. As shown in Fig. 1.2(a), channels may be of three types: *simplex*, *half duplex*, and *full duplex*. Simplex channel implies one way communication, half duplex implies bi-directions communication on a sharing basis one at a time, while full duplex channel implies simultaneous bi-directional communication. *Noise* and *interference* are the most serious problems associated with a channel. It is inevitable that a signal will deteriorate during the processes of transmission, propagation, and reception because of some distortion in the system or because of the introduction of noise. Noise is unwanted energy (usually of random nature) present in a transmission system due to a variety of causes. Since noise will be received together with the signal, as shown in Fig. 1.2(b), it places a limitation on the transmission system as a whole. When noise is severe, it may mask a given signal so much that the signal becomes unintelligible and therefore useless.

Though noise may interfere with a signal at any point in the communication system, its effect will be maximum when the signal is weak. Hence, the most noticeable noise is that in the channel or at the input to the receiver. Correspondingly, when the signal is strong, the noise effects are less. This is defined using the parameter *signal power to noise power ratio* or *signal-to-noise ratio* (SNR). Better the SNR, stronger the signal in the presence of noise. The different types of noise are discussed in Chapter 3.

Receiver There are many varieties of receivers in communication systems since the exact form of a particular receiver is influenced by the opposite tasks to that of the transmitter and many other requirements. Among these requirements are the modulation scheme used, the operating frequency and its range, error-handling tasks, and the type of output device required, which in turn depends on the destination of the intelligence received. Most of the wireless receivers are of superheterodyne type with the intermediate frequency (IF) stage and then the local oscillator and mixer stage for final RF upconversion. Receivers vary in complexity from a very simple crystal receiver, with headphones, to a far more complex *rake receiver*, explained in the chapter 7 of modulation techniques.

As already stated, the purpose of a receiver and the form of its output influence its construction. The output of a receiver may be fed to a loudspeaker, video display unit, radar display, television picture tube, pen recorder, or computer. In each instance, different arrangements must be made, each affecting the receiver design.

1.2 GENERAL MODEL FOR WIRELESS DIGITAL COMMUNICATION LINK

A study of wireless digital communication involves the in-depth study of the whole point-to-point link, covering the fundamentals of each block of the link. This section provides an introduction to the blocks and their importance. The blocks will be explained in detail in subsequent chapters.

Figure 1.3 provides a simplified block diagram of a digital communications link. A transmitter begins and ends with an analog signal (except the readily stored or generated digital base). The signal that comes out as multimedia information is analog in nature, which should be first converted into the digital form. Initially, wireless communication was used only for voice communication, but now any signal can be communicated. In the case of video communication, a huge storage capacity and high speed of communication are required, and hence, source encoding for compression of the database is necessary. Here, standard methods may be used to compress the data, and the stored files with standard extensions, such as .jpg, .avi, .mp3, .gif, .tif, and .dat, can be made available for transmission. The basic communication model, as shown in Fig. 1.3, is a *systematic assemblage* of the *forward path* and the *reverse path*.

In general, channel coding aspects need more attention in wireless communication, whereas line coding is important in wired communication.

Source coding/decoding stage The first step is to convert a continuous analog signal into a discrete or digital bit stream. This process is called *digitization*. The next step is to add information coding for data compression. The information to be transmitted from the source may be human-originated (speech) or machine-originated (data or image). The source encoder with compression eliminates the *inherent redundancy* in the information (thus compressing) to maximize the transmission rate, and the encrypter ensures secrecy of data. The encryption process is described in Chapter 5 of the source coding stage.

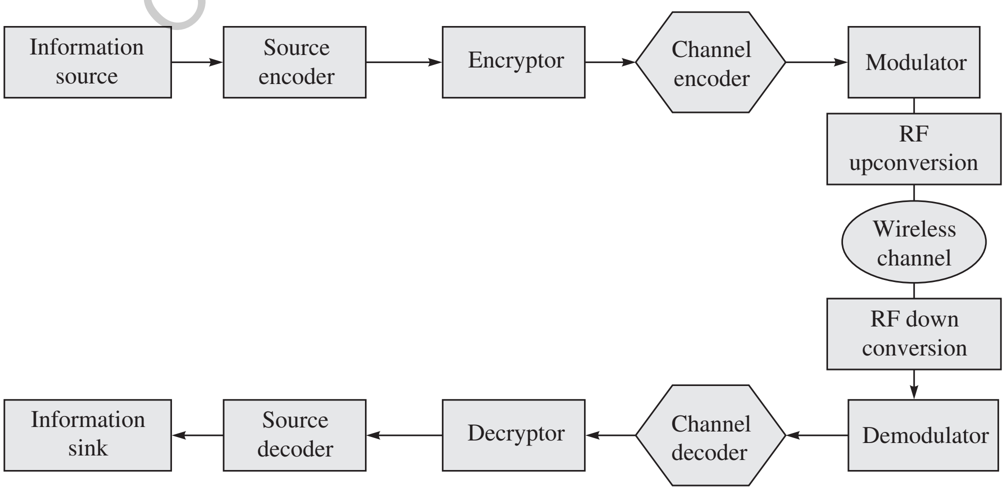


Fig. 1.3 Basic model of wireless digital communication link

Channel coding/decoding stage Data must be protected against perturbations introduced by the noisy channel, which could lead to misinterpretation of the transmitted message at the receiving end. Line coding techniques are used for inserting systematic amplitude variations, power levels, and synchronization points, whereas channel coding techniques are used to insert structured sequences. Both techniques help in combating channel errors. Data can be protected through the following error control strategies:

Forward error correction It uses error-correcting codes that are derived mathematically and inserted systematically at the transmitting end, and are able to correct errors at the receiving end.

Automatic repeat request It uses error-detecting codes with no capabilities of correcting them, (except single bit error correction techniques) but with strategies of retransmissions (sending the same data again) when erroneous data is received.

The channel coding stage systematically adds extra bits to the input data stream, even after the removal of the redundant bits by the source coders to balance the reliable transmission capabilities of the channel. Though sending the extra bits involves extra cost in terms of bandwidth utilization and speed, these bits are used for error correction to enhance the quality of reception. The channel decoder decodes the data in such a way that the effects of noise and interference in the communication channel are minimized. These techniques are discussed in depth in Chapter 6.

Modulator/Demodulator stage The modulation method to be used should be selected based on the channel characteristics. Channel-related issues and the corrections for channel effects are discussed in Chapters 3 and 8. Corrections should take place before demodulation to reduce the probability of errors. The output from the channel coder is fed into the modulator. Since the modulator deals with complex modulation techniques, there are independent I (in-phase) and Q (quadrature-phase) components in the radio; half of the information can be sent on I and the other half on Q. This is one reason why digital radios work well with this type of digital signals. The I and Q components are separate and orthogonal. The modulator block generates a signal suitable for the transmission channel. The blocks in the reverse path do the opposite of those in the forward path. Modulation techniques are basically divided into three types: *pulse modulation*, *carrier modulation*, and *spread spectrum techniques*. The latest modulation technique based on multicarrier transmission, which eliminates most of the problems of wireless channel, is orthogonal frequency division multiplexing (OFDM). An overview of all the modulation techniques is given in Chapter 7.

Intermediate frequency/radio frequency stage After the modulator, the rest of the transmitter looks similar to a typical RF or microwave transmitter. The signal is converted up to an IF and then further upconverted to a higher RF. Any undesirable signal produced by the upconversion is then filtered out. Depending upon the requirements, the power amplifier is selected for amplifying the power to cover the required transmission distance. The receiver RF section provides efficient coupling between the antenna and the rest of the hardware, which utilizes the energy abstracted from the radio wave. It also provides discrimination or selectivity against image and IF signals. Major receivers follow the superheterodyne technique at the RF stage.

> Symbols are the group of bits processed together at the modulation stage and they represent specific amplitude and phase as per the bit pattern.

Additional Comments—Transmitter

Sometimes, training sequences need to be sent for estimation or equalization. This can make synchronization (or finding the symbol clock) easier for the receiver. Symbols are processed in synchronism. The symbol clock is an essential part of the link and represents the frequency and exact timing of the transmission of the individual symbols. At the symbol clock transitions, the transmitted carrier is at the correct *I/Q* (or magnitude/phase) value to represent a specific symbol. Then the values (*I/Q*) of the transmitted carrier are changed to represent another symbol. The interval between these two is the symbol clock period. The reciprocal of this is the symbol clock frequency. The symbol phase is correct when the symbol clock is aligned with the optimum instant(s) to detect the symbols. One essential step after channel coding in the transmitter is filtering, which is required for good bandwidth efficiency. Without filtering, signals would have very fast transitions between states and therefore, very wide frequency spectra—much wider than is needed for the purpose of sending information. A single filter can be shown for simplicity in the block diagram, but in reality, there are two filters, one each for the *I* and *Q* channels. This creates a compact and spectrally efficient signal that can be placed on a carrier. Many times, pulse shaping and windowing techniques of digital signal processing (DSP) make the communication efficient.

Additional Comments—Receiver

The desired receiver characteristics or issues are as follows:

Sensitivity This is expressed in terms of the voltage that must be applied to the receiver input to give a standard output.

Selectivity This characteristic determines the extent to which the receiver is capable of distinguishing between the desired signal and the signal of other frequencies.

Fidelity This represents the variation of the output with the modulation frequency, when the output load impedance is a resistance. At the lower modulation frequencies, it is determined by the low-frequency characteristics of the audio frequency amplifier. At the higher modulation frequencies, the fidelity is affected by the high-frequency characteristics of the audio frequency amplifier.

Noise figure This is a measure of the extent to which the noise appearing in the receiver output in the absence of a signal is greater than the noise that would be present, if the receiver was a perfect one from the point of view of generating the minimum possible noise. It determines the smallest power that may be received without being drowned out by the noise.

Learning about the *wireless medium* is essential to understand the reasoning behind the specific designs for wireless communication protocols or systems. In particular, the design of the physical and medium access protocols is highly affected by the behaviour of the channel that varies substantially in different indoor and outdoor areas. The diversity and complexity of transmission techniques in wireless communications are far more complex than those of wired communications.

The incoming RF signal is first downconverted to IF and demodulated. The ability to demodulate the signal is hampered by factors including atmospheric noise, competing signals, and signal strength variations. The concept of demodulation is explained in Chapter 7. Generally, demodulation involves the following stages:

- Carrier frequency recovery (carrier lock)
- Symbol clock recovery (symbol lock)
- Signal decomposition to *I* and *Q* components
- Determination of *I* and *Q* values for each symbol (slicing)
- Decoding and de-interleaving

Automatic gain control and power control are the important aspects of transceivers and require closed-loop systems.

- Expansion to original bit stream
- Digital-to-analog conversion, if required

Carrier and symbol clock recovery is a complex issue in the receiver. Both the symbol clock frequency and phase (or timing) must be correct in the receiver to successfully demodulate the bits and recover the transmitted information. Offset in frequency or phase will lead to unsuccessful demodulation. Usually, the frequency of a symbol clock is fixed, and both the transmitter and receiver accurately know this frequency. The difficulty is to get them aligned in phase or timing. A variety of techniques is available and most systems employ two or more such techniques. If the signal amplitude varies during modulation, a receiver can measure the variations. The transmitter can send a specific synchronization signal or a predetermined bit sequence such as 10101010101010 to train the receiver's clock. In systems with a pulsed carrier, the symbol clock can be aligned with the power turn-on of the carrier. In the transmitter, it is known where the RF carrier and digital data clock are because they are being generated inside the transmitter itself, whereas in the receiver, this is not known. The receiver can approximate where the carrier is, but has no information about the symbol clock phase or timing. Creating the carrier and symbol clock recovery algorithms is a difficult task in receiver design. This task can be made easier by the channel coding performed in the transmitter.

Mobile telephony, mobile internet services, and wireless local area networks (WLANs) are a few applications that are based on protocol. The lowermost layer of the protocol stack is the physical layer, which is the wireless link along with the standard specifications. These are explained in part 4 of the book (Chapters 10, 11 and 12).

Some of the useful signal processing aspects observed in the wireless link are Fourier series and Fourier transforms of the various functions (observing the signal in the time and frequency domains), sampling theorem, filters, correlation, convolution, and windowing. Various properties of Fourier transforms are applied at various stages, and these fundamentals can be revised by self-study.

1.3 BANDWIDTH

A signal may have one or more frequency content, which can be represented in the frequency domain. Information, which may be in the form of analog or digital signals, can be represented in the time domain (amplitude versus time plot) and the frequency domain (amplitude versus frequency plot, also called the spectrum). A digital signal is the representation of a signal with discrete values at discrete time. It is produced by the sampling of a continuous envelope of information and will carry discrete, well-defined amplitude levels. Binary coded data is one typical case of a digital system; it takes only two values of amplitude levels, one each for logic 0 and logic 1. It will carry the amplitudes decided for logic 0 and 1. When an analog or a digital time domain signal is converted into a frequency domain signal, the significant frequency components of the spectrum decide the bandwidth. Practically, the signal is band-limited by applying certain techniques to meet certain requirements.

There is no universally satisfying definition for the term *bandwidth*, which is used in the following circumstances:

- It is used to characterize a signal, which can be the input signal or the baseband or broadband to be transmitted. Correspondingly, this is called the signal or transmission bandwidth.
- A channel allocated to the user to allow the transmission of maximum frequency content (allowable range of frequencies) is called channel bandwidth. It decides the capacity of the transmission. Channel bandwidth may be decided by the service provider.

Note: It is observed that an analog signal consumes less spectrum compared to its digital counterpart and hence requires lesser bandwidth, because the digital counterpart is the result of sharp transitions.

- While designing wireless system hardware, including transmitter and receiver, the frequency response of the hardware stages must be such that the total system bandwidth supports the channel bandwidth (or the hardware frequency response must be set accordingly).

Bit rate, symbol rate, and baud rate Digital data transfer is measured in bits per second, as mentioned earlier, or in symbols per second units. When the number of bits is represented together at the modulator front end, the bit rate is converted into the symbol rate. To understand and compare the efficiencies of different modulation schemes, it is important to first understand the difference between the bit and symbol rates. The transmission bandwidth due to digital modulation techniques depends on the symbol rate, and not on the bit rate (refer to digital modulation schemes discussed in Chapter 7). The bit rate is the frequency of a system bit stream.

$$\text{Symbol rate} = \frac{\text{Bit rate}}{\text{Number of bits transmitted with each symbol}}$$

Each symbol represents M finite states and k bits of information, where

$$k = \log_2 M \tag{1.1}$$

The symbol rate is measured in symbols per second.

The baud rate refers to the signalling rate at which the data is sent through a channel and is measured in electrical transitions per second. It is the reciprocal of the duration of the shortest signalling element. If there is one signal transition per bit, then the bit rate and the baud rate are identical. If two electrical transitions are required for each bit, as in the case of return-to-zero (RZ), then at a rate of 9600 baud, only 4800 bits per second can be conveyed (refer to Section 7.2 of Chapter 7 for further discussion on signalling). The baud rate decides the bandwidth as it decides the highest frequency occurred.

The spectrum of a signal is the collective representation of all its frequency components along with their amplitude weights.

Example 1.1 Let the symbols be represented by 4, 8, and 16 modulo values at the front end of a modulator. What will be the symbol rates in all the cases if the bit stream is of 256 Mbps?

Solution A modulator modulates symbols rather than bits. For 4 modulo values, 2 bits/symbol are taken in for modulation. For 8 modulo values, 3 bits/symbol are taken in, and for 16 modulo values, 4 bits/symbol are taken in. Hence,

for case 1, Symbol rate = 256 Mbps/2 = 128 Mbps
for case II, Symbol rate = 256 Mbps/3 = 85.33 Mbps
for case III, Symbol rate = 256 Mbps/4 = 64 Mbps

Bandwidth of signal and system A system can be as simple as a low-pass filter or an amplifier or as complicated as an entire satellite communication link. Bandwidth, when referring to a system or a device, usually means the ability to pass, amplify, or somehow process a band of frequencies. However, bandwidth of significant energy for a signal can be subjective. For example, the speech signal bandwidth of maximum energy could be specified as the range between 100 Hz and 6000 Hz, whereas the bandwidth of significant energy for

Note: The line coded signal decides the transmission channel bandwidth in the case of baseband communication, whereas the modulated signal decides the bandwidth in the case of broadband communication.

telephone quality speech could be specified as between 100 Hz and 3000 Hz. The bandwidth for a system is usually defined between the 3 dB points (at higher and lower cut-off frequencies) assuming 0 dB point as the maximum gain, when the system gain is plotted against the range of frequencies.

Pulse degradations are dependent upon the rate of transmission, channel bandwidth, SNR condition, and channel delay.

In strict technical terms, there is no need to differentiate between analog and digital signals, because we just need to look at the spectral content of each signal, the extent of which determines the bandwidth. Typical analog signals, because of their smooth variations, usually have a finite bandwidth, whereas digital signals, due to their discrete nature, usually have unlimited bandwidth. However, it is useful to specify a finite bandwidth for digital signals. To find the most appropriate bandwidth for a digital signal, it is necessary to know the range of frequencies that contains the significant energy of the signal.

We can now make a simple but important observation. When the available bandwidth of a transmission system (medium) is equal to or larger than the bandwidth of a signal that is to be transmitted over the system, and also the actual transmitted signal frequency contents at an instant of time is less than the maximum frequency allowed by the medium, that is,

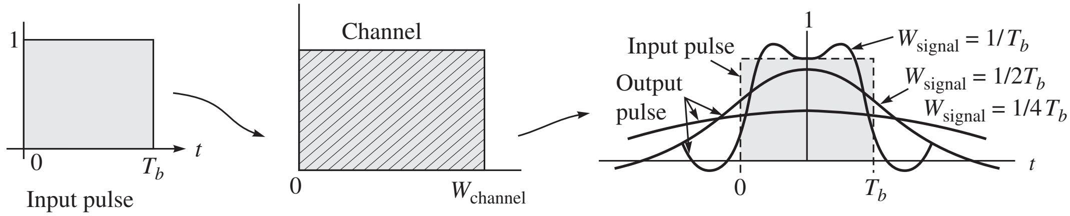
$$W_{\text{signal}} \le W_{\text{channel}} \tag{1.2}$$

then, the entire information content of the signal can be recovered at the receiving end. Conversely, when the transmission system bandwidth is less than the signal bandwidth, some degradation of the signals always occurs because of the loss of frequency components due to its lack of capacity to transfer those frequencies.

Pulse transmission over channel Let us consider digital signals and the bandwidth requirements for pulse transmission. We have to distinguish between the case of an exact reproduction at the receiving end of a transmitted square pulse (which represents a binary digit 1) and a distorted reproduction. An exact reproduction would require a transmission channel with ideally infinite bandwidth, as an ideal square pulse has infinite bandwidth due to extremely high frequency content to retain its sharpness. However, if we only need to detect that a pulse has been sent, we can get by with a finite channel bandwidth. For example, if we were to calculate the effect of an ideal low-pass filter on a square pulse, we would find the output to be a distorted pulse that resembles the original pulse better and better with increasing bandwidth W of the filter. The channel acts as a low-pass filter. Hence, higher harmonic losses are certain. In addition, attenuation also occurs. Ideally, the bandwidth of a binary digital signal with the baud rate same as the bit rate will always be half of that of its bit rate. This is because the consecutive 1 and 0 bits will establish the worst-case condition for transitions, which will decide the highest frequency content, making one cycle of frequency and bandwidths being represented in terms of frequencies normally.

The variation in the bit rate of a channel with fixed bandwidth W_{channel} generates different situations, as shown in Fig. 1.4, because a change in the bit rate will vary the signal bandwidth. Here, the bit rate and signal bandwidth are related mathematically as

$$W_{\text{signal}} = 1/2T_b \tag{1.3}$$



Wireless channel acts as low-pass filter as it spreads the pulse

Fig. 1.4 Response of an ideal pulse transmitted through a channel according to the bit rate over a fixed channel bandwidth

> *SNR–bandwidth trade-off*: Transmission rate increases at the cost of reduced SNR due to the noise distributed over the wide bandwidth and vice versa.

where T_b is the 1-bit interval (time duration). For many purposes, this bandwidth yields a resolution with an acceptable error rate.

Since a wireless transmission channel with multipath effects has band-pass characteristics similar to that of a low-pass filter, a pulse propagating over the channel will be affected by the spreading of the pulse. The reasons are explained in chapter 3. As the bit interval becomes narrower, more errors are likely to occur. However, an advantage of digital transmission is that the message is preserved. In analog transmission, the signal becomes irreversibly distorted due to the addition of noise. In contrast, in digital transmission, even though the individual pulses become badly distorted during propagation, as long as the distorted signal that is received can be identified with the presence or absence of a pulse, the original message is preserved. There are some techniques for regenerating the digital signal with the help of a pre-decided threshold level.

Signal-to-noise ratio and channel bandwidth The amount of information that a channel can carry reliably depends on the bandwidth of the channel and the magnitude of the noise present in the channel. The amount of noise present in any channel limits the number of distinct amplitude levels that a signal propagating may have. For example, if a varying analog signal has a maximum level of 10 V and the noise level is 5 V, the signal may have only two levels. On the other hand, if the noise level is only 1 mV, the same signal can be divided into approximately $10\,\text{V}/1\,\text{mV} = 10^4$ levels. Figure 1.2(b) illustrates how noise that has been added during transmission can degrade the signal and hence, its resolution at the receiving ends.

The SNR is the standard measure of the noise level in a system. It can be measured at different stages in the wireless link. It is the ratio of power P_s to noise power P_n. Since power is proportional to voltage squared, we can express SNR as

$$SNR = \frac{P_s}{P_n} = \left(\frac{V_s}{V_n}\right)^2 \tag{1.4}$$

where V_s is the signal voltage and V_n is the noise voltage (because of their multitude of random amplitudes, noise voltages are typically given as rms voltages). SNR is usually expressed in decibels (dB).

$$SNR_{dB} = 10\log_{10} SNR = 20\log_{10}\left(\frac{V_s}{V_n}\right) \tag{1.5}$$

Signal power plays a very important role in successful communication. On increasing the signal power, the effect of channel noise reduces and the signal is received more accurately. A larger SNR allows for a longer distance of transmission. An important feature of signal power is that the SNR and bandwidth are exchangeable. Higher the bandwidth, more will be the noise power distributed, which reduces the SNR. This means that to maintain the given data rate and accuracy of the information transmission, we have to trade the SNR for bandwidth. One may reduce the bandwidth if the SNR is to be increased.

It can be shown that the relationship between the bandwidth expansion factor and the SNR is exponential. Consider the SNR–bandwidth trade-off. *SNR*1 is a value with a particular rate of transmission with bandwidth W_1. *SNR*2 is another value with a different rate and bandwidth W_2. Then, for the same channel capacity, it can be derived that

$$SNR2 \approx SNR1^{W_1/W_2} \tag{1.6a}$$

Thus, if $W_2 = 2W_1$, then $SNR2 \approx SNR1^{1/2}$ (1.6b)

That is, *SNR*2 is the square root of *SNR*1.

Example 1.2 Compare the SNR requirements for 1 bit/symbol and 2 bits/symbol transmission systems that have a bit rate of 1 Mbps.

Solution The visualization of this example will be better if we treat one symbol block as one pulse, because the symbol rate decides the transmission bandwidth.

Considering first nulls in the sync shaped frequency domain response of a pulse:

For 1 bit/symbol transmission bandwidth W_1, *baud rate* = 1 Mbps (because it takes in 1 bit per symbol).

For 2 bit/symbol transmission bandwidth W_2, baud rate = 0.5 Mbps. Hence, $W_1/W_2 = 2$.

Now, $SNR2 \approx SNR1^{W_1/W_2} \Rightarrow SNR2 \approx SNR1^2$

Thus, theoretically, the second scheme requires a higher value of SNR compared to the first scheme for the same bit rate to be transmitted.

Note for 1 *bit/symbol transmissions*: We know that for square signals, the spectrum contains odd harmonics of the fundamental, which here equals $1/2T_b$. Thus, the signal's bandwidth is infinite. In practical terms, we use 90 per cent power bandwidth to assess the effective range of frequencies consumed by the signal. The first and third harmonics contain that fraction of the total power, meaning that the effective bandwidth of our baseband signal is $3/2T_b$ or, expressing this quantity in terms of the data rate, $3R_b/2$. Thus, a digital communications signal requires more bandwidth than the data rate: a 1 Mbps baseband system requires a bandwidth of at least 1.5 MHz. However, bandwidth also depends upon the adopted line coding scheme.

Shannon's and Nyquist's equations Channel capacity is decided by the transmission bandwidth and SNR condition, and the relationship is given by Shannon and Nyquist from their independent research. In 1948, Dr Claude Shannon of Bell Telephone Laboratories published a groundbreaking work entitled *The Mathematical Theory of Communication*, in which he described the development of communication systems that transmit data effectively with limits on the exchange of the SNR and bandwidth. The limitations imposed on communication by the transmission with zero errors. We can consider the channel as a pipe through which we send information. Shannon worked on the channel capacity and found the equation for the band-limited signal to be transmitted over additive white Gaussian noise (AWGN) channel as follows:

> Channel capacity is the maximum amount of data that can be pumped through the channel in a fixed period of time and can be measured in terms of bits per second.

$$C = W_{channel} \log_2(1 + SNR) \tag{1.7a}$$

Nyquist had given another formula:

$$C = 2W_{channel} \log_2 M \tag{1.7b}$$

Here, W_{channel} is the channel bandwidth in hertz and *SNR* is the power ratio in general, in which S is the signal power and N is the noise power in watts. Equation (1.7) gives the maximum possible data transmission when 1 bit/symbol is transmitted; $M = 2^k$ are the signalling levels. If more bits per symbols are being transmitted, then the maximum rate of transmission of information in symbols per second is C_s, and for k bits/symbols, we can say $C = kC_s$. Combining Nyquist and Shannon relationships, $k = C$/symbol rate, where C is the maximum bit rate capacity, k is the number of bits per signalling element (symbol), and symbol rate is two times the bandwidth of the signal according to Nyquist relation. There may be 2^k different possible bit combinations to send in the form of symbols.

There is a parameter related to the SNR that is more convenient for determining digital data rates and error rates. It is the ratio of the signal energy per bit to the noise power density (noise power per hertz), E_b/N_o. Consider a signal that contains binary digital data transmitted at a certain bit rate R. Recalling 1 watt = 1 J/s, the energy per bit in a signal is given by

$$E_b = ST_b$$

where S is the signal power and T_b is the time required to send 1 bit. The data rate is just $R = 1/T_b$. For thermal noise,

$$E_b/N_o = S/KTR$$

Example 1.3 A standard 4 kHz telephone channel has an SNR of 25 dB at the input to the receiver. Calculate its information-carrying capacity. In addition, find the capacity of the channel if its bandwidth is doubled while the transmitted signal power remains constant.

Solution SNR = antilog (25/10) = 316
Capacity of the channel in the first case

$$C = 4000\log_2 (1 + 316) = 33.233 \text{ kbps}$$

If the SNR is 316, it means that when the signal power is 316 mW, the noise power is 1 mW. Now, the bandwidth is doubled with no change in the signal power, effectively, the noise power is doubled due to twice the bandwidth. Hence, the SNR drops to half the original value.

Capacity in the second case

$$C = 8000\log_2 (1 + 316/2) = 58.503 \text{ kbps}$$

Thus, the capacity of the channel has increased.

1.4 TYPES OF SIGNALS

Appropriate signal processing can be applied in the transmitter, as well as receiver, if and only if we know the type of the signal. If we are aware of the nature of the signal, we can treat it in the time or frequency domain and can identify the changes applied. In addition, we can decide the approach to deal with the system and performance parameters of the system.

1.4.1 Analog and Digital Signals

Signals are classified in terms of the nature of amplitude. Normally, they are represented in the time domain.

Let us define the analog and digital signals once again in terms of DSP. A signal whose amplitude takes all the values in the specified range over the measuring interval or time, and is continuous in time is called an *analog signal*. Here, the signal can take an infinite number of values, and precision is dependent upon the resolution of the system. If the signal amplitude takes a finite number of values and not all, it is called a *digital signal*. Binary is a special case of digital signal and takes only two values, one each for logic 0 and 1.

1.4.2 Continuous-time and Discrete-time Signals

Signals are classified on the basis of time as continuous-time and discrete-time signals, and are represented in the time domain. A continuous-time signal is specified for every value of time, whatever precise time can be resolved, whereas a discrete-time signal is specified with the gap of measuring instants.

The following are the mathematical representations for signals with peak value A_0:

$$s(t) = A_0 \sin \omega t \qquad \text{(continuous-time signal with time variable } t\text{)}$$

$$s(n) = A_0 \sin \omega n/N \quad \text{(discrete-time signal with index variable } n\text{)}$$

where N is a period of n samples.

A discrete-time signal is represented at discrete instants of time with its natural value or quantized value. The time variable is not continuous and hence, a discrete-time signal can be represented as a sequence of numbers.

From these two types, four different signal categories can be formed:

(a) Continuous-time analog signal (real-time signals)
(b) Continuous-time digital signal (square wave representing a binary signal)
(c) Discrete-time analog signal (with natural value of samples)
(d) Discrete-time digital signal (with quantized value of samples)

A discrete-time signal is represented as a sequence $s(n)$, where n can take in a set of values in the integer range $-\infty$ to $+\infty$. In most cases, the discrete-time signal $s(n)$ is obtained by sampling a continuous-time signal $s(t)$ at periodic interval ΔT_s. So, we can write $s(n) = s(t)|_{t=n\Delta Ts}$.

A *discrete-time system* is one that accepts a set of sequences $s_i[n]$ (i stands for the ith sequence) and produces a set of sequences $r_j[n]$ as output.

1.4.3 Periodic and Aperiodic Signals

A signal is said to be periodic for some positive constant T_0 (or N for a discrete signal), that is, a fixed interval, if it satisfies the following conditions:

$$s(t) = s(t + T_0) \text{ for all } t \qquad \text{(continuous time)} \tag{1.8a}$$

$$s(n) = s(n + N) \text{ for all } n \qquad \text{(discrete time)} \tag{1.8b}$$

The smallest value of T_0 that satisfies this condition is called a *period* in terms of time unit. It is obvious that $s(t)$ will remain the same when it is shifted in time by one period. A periodic signal

> *Note*: For the class of periodic signals, decomposition in sinusoidal components is called a *Fourier series*, whereas for the class of finite energy signal (aperiodic), it is called a *Fourier transform*.

Voice signal is a continuous-time analog, aperiodic, random, and energy signal.

must start at $-\infty$ and continue forever. Moreover, it can be generated by repeating the signal $s(t)$ with the period T_0 infinite number of times. The instants from and to which instants of time the period is measured is immaterial due to periodicity; the shape of $s(t)$ during that period must repeat itself an infinite number of times. The signal that occurs for a finite duration of time is called an *aperiodic signal*. Here, the shape of $s(t)$ is not repeated an infinite number of times. It is a time-limited non-repetitive signal.

Most signals of practical interest can be decomposed into a sum of sinusoidal signal components. The signals in the time domain and the corresponding frequency domain equivalents are given in Table 1.1.

Table 1.1 Time and frequency domain signal equivalents

Time domain signal	**Frequency domain equivalent**
Continuous time, periodic	Discrete spectrum, aperiodic
Continuous time, aperiodic	Continuous spectrum, aperiodic
Discrete time, periodic	Discrete spectrum, periodic
Discrete time, aperiodic	Continuous spectrum, periodic

1.4.4 Deterministic and Probabilistic Signals

A signal can be classified as deterministic if there is no uncertainty with respect to its value at any instant of time. Probabilistic signals, also known as random or non-deterministic signals, cannot be predicted, that is, there is some degree of uncertainty. Deterministic signals can be represented with a mathematical expression, which will be unique. Random signals are generated from random or stochastic processes.

Random functions of time are often referred to as stochastic signals. A stochastic signal may be continuous or discrete in time and may have continuous-valued or discrete-valued amplitudes. Stochastic processes are classes of signals whose fluctuations in time are partially or completely random; examples of such signals are speech, music, image, time-varying channel response, noise, and video. Stochastic signals are completely described in terms of the probability model and theory, but can also be characterized with relatively simple statistics, such as the mean or statistical averages, correlation, and power spectrum. They must deal with the ensemble averages, variance, probability distribution function (PDF), cumulative distribution function (CDF), and so on. Readers can refer to any book on statistical signal modelling to explore these topics further.

1.4.5 Energy and Power Signals

Power is related to signal voltage or current. Here, the continuous-time analog signal is considered, and therefore, we have to deal with integrations in the subsequent formulas. Power signal can be defined as

$$P(t) = \frac{V^2(t)}{R} \quad \text{or} \quad P(t) = i^2(t) \times R \tag{1.9}$$

where R is the resistance across which power is measured.

In a communication system, power is often represented in the normalized form, assuming $R = 1\,\Omega$, though the resistance may have another value in the actual circuit. The actual value of power is obtained by denormalizing the normalized power value. Conventionally, irrespective of whether the signal is of the voltage or current waveform, the normalization convention for power allows us to express the instantaneous power as

$$P(t) = s^2(t) \tag{1.10}$$

Energy dissipated in the time interval $-T/2$ to $T/2$ of a signal with instantaneous power is measured by the following expression:

$$E_S = \int_{\frac{-T}{2}}^{\frac{T}{2}} s^2(t)dt \tag{1.11}$$

The average power dissipated by the signal during the same interval is

$$P_{av} = \frac{E_s}{T} = \frac{1}{T}\int_{\frac{-T}{2}}^{\frac{T}{2}} s^2(t)dt \tag{1.12}$$

The performance of the communication link depends on the energy of the received signal. Higher the energy, more accurate the signal detection. At the same time, power is the rate at which the energy is delivered. This is necessary because voltages, currents, or EM field intensities are related to powers and they need to be designed as per power requirements. The signal $s(t)$ can be converted into the discrete form by sampling, and samples can be written as $s(n)$, where n is the index value. All these formulas can be rewritten by replacing the integration with summation and $s(t)$ with $s(n)$. Similarly, the changes can be applied to energy and power signals as well.

This fundamental knowledge can be used to differentiate between energy and power signals. While analysing the signals, it is often desirable to deal with the waveform energy E_s. We can classify $s(t)$ as an energy signal, if and only if, it has finite but non-zero energy for all time, that is, when $T \rightarrow \infty$.

$$E_s = \int_{-\infty}^{\infty} s^2(t)dt \quad \text{(continuous-time signal)} \tag{1.13a}$$

$$E_s = \sum_{-\infty}^{\infty} |s(n)|^2 \quad \text{(discrete-time signal)} \tag{1.13b}$$

In the real world, transmitted signals have finite energy ($0 < E_s < \infty$). A finite energy signal has zero average power. However, in order to describe periodic signals, which by definition exist for all time and thus have infinite energy, these are called power signals. Even random signals having infinite energy are power signals. If E_s is infinite, the average power P_s may be either finite or infinite. A signal is defined as a power signal only if it has finite but non-zero power for all time t.

$$P_s = \lim_{T \rightarrow \infty} \frac{1}{T}\int_{\frac{-T}{2}}^{\frac{T}{2}} s^2(t)dt \tag{1.14}$$

To study random signals, mathematical models based on the PDF and exhibiting their behaviour are used.

For signal $s(n) = Ae^{j\omega_n}$ has average power A^2.

The classification of energy and power signals is mutually exclusive. An energy signal has finite energy but zero average power (e.g., deterministic and aperiodic signals) and can be generated in a laboratory. A power signal has finite

Ramp signal is neither the energy signal nor the power signal.

average power but infinite energy (e.g., periodic and probabilistic signals). It is impossible to generate a true power signal in practice, because such a signal has infinite duration and infinite energy.

From the theory of linear systems, Parseval's theorem states that the Fourier transform preserves energy and power. However, the energy (or power) in the complex envelope is not equal to the corresponding energy (or power) in the corresponding band-pass signal.

1.5 TYPES OF COMMUNICATION SYSTEMS

There are two possible options in many scenarios while dealing with communications between two hardware ends—a transmitter and a receiver.

- The input (or baseband) signals may be analog or digital
- The channels may be wired (guided) or wireless (unguided)
- The transmissions may be analog or digital
- The number of bits sent at a time may be serial (one bit at a time) or parallel (more bits at a time, i.e., symbols)
- The communication may be baseband or passband (general terms for broadband and wideband)
- The mode of communication may be synchronous or asynchronous
- The information may be real time or non-real time (stored data)
- The direction of transmission may be unidirectional or bidirectional

Out of the two possibilities, only one can exist at a time. To have a combination of both possibilities, either conversions or convergence in the system is required. As there are two possibilities in the input signals and two possibilities in the transmissions, according to the binary theory, four combinations of communication systems are possible. The systems may be analysed by using a qualitative approach first and then a quantitative approach. Moreover, we must analyse the ideal system and then the actual system, with noise.

In general, communication systems can be of four different types: analog input–analog transmission, analog input–digital transmission, digital input–digital transmission, and digital input–analog transmission. The different types of systems and the corresponding modulation schemes are described here for a proper visualization.

Analog input–analog transmission Wireless communication commercially started with amplitude modulation (AM) radio broadcasting in the range 550–1600 kHz. Thereafter, frequency modulation (FM) transmissions also started commercially in the range 88–108 MHz. In both these systems, the input was in the analog form of audio signal. These broadcast systems still exist. When analog television standards were framed, quadrature AM was selected for video information and FM for audio information for combined audio and video transmission, resulting in vestigial sideband communication. These standards are still followed to maintain compatibility with the older televisions and follow the very high frequency (VHF) and ultra-high frequency (UHF) ranges. In local loops of wired telephone lines, the analog baseband signal is transmitted without modification in the signal.

The PCM scheme serves various stages in the communication link—analog-to-digital converter in the source coder as well as modulator in the baseband communication link.

In the near future, commercial systems based on the analog input–analog transmission may become obsolete. The transient period of revolution has already started with digital broadcast systems employing A-D-A conversion stages with the digital audio broadcasting (DAB) and digital video broadcasting (DVB) standards. High definition radio (HD radio) and digital radio mondiale (DRM) systems have also come up. All these systems follow the OFDM modulation scheme, which is suitable for long-distance communication, and hence for broadcasting.

Analog input–digital transmission The pulse code modulation (PCM) scheme, which exists for analog-to-digital conversion (ADC), is considered in the source coding stage of the wireless communication link, though it is the method for analog input–digital transmission. Thus, PCM forms the basis for the source coding stage of the wireless link for real-time input signals such as voice, image, and video. It is discussed in detail in chapter 5. Digital transmission in its baseband form is suitable only for transmissions on the wired lines. To achieve this, ADC is required, which can be achieved through the PCM scheme. PCM signals of 64 kbps bit rate are transmitted over the telephone trunk lines or over the integrated services digital network (ISDN) or broadband ISDN (B-ISDN) channels. Another method for analog input–digital transmission is delta modulation (DM), but because of its practical limitations related to slope overload and sampling rate, it is not standardized in commercial systems. PCM signals can also be converted into frames for transmissions over wired links of computer networks. Differential pulse code modulation (DPCM) and adaptive DPCM (ADPCM) are the modified and bandwidth-efficient versions of PCM.

Digital input–digital transmission When it is necessary to send digital information in its baseband form, the binary form of transmission may not always be suitable, as it may not be compatible with the transmission channel. In addition, the binary form of transmission adds a DC voltage level to the final transmission, which takes more energy in the signal. Therefore, it is required to convert the form of transmission by changing the bit representation format or voltage levels for shaping the signal power, and also incorporating the synchronization points in the signal. In short, the signal can be shaped as per the desired spectrum characteristics for digital baseband communication. *Non-return-to-zero* (NRZ), RZ, Manchester, differential Manchester, and bipolar are some methods that have a final digital form of transmission. These methods are normally suitable for wired line or computer networks; however, they are incorporated in wireless links as well. These methods are also called *digital signalling* as they are a suitable form for ISDN lines. It is also called *line coding*. Line coding can be applied to the digital baseband in wireless communication before the modulation stage. Refer to Chapter 7 for further discussion on this topic.

Digital input–analog transmission This type of transmission is mainly used in the systems that use a modem (modulator-demodulator), either over wired lines or wireless links. Here, the modulation scheme converts the input digital signal into the analog form using the carrier wave. The final wireless communication is always in the analog form. If wireless transmission can be used and the carrier frequency after modulation does not fall in the RF range, it is necessary to use an RF upconversion. If wired communication is used, only a data modem can be used without upconversion. Amplitude shift keying (ASK), frequency shift keying (FSK), M-ary phase shift keying (M-PSK), M-ary quadrature amplitude modulation (M-QAM), minimum shift keying (MSK), spread spectrum modulation (SSM), and OFDM fall into this category. The details of these modulation schemes are provided in Chapter 7.

1.6 WIRED VERSUS WIRELESS MEDIA

The existing systems are not all wireless; a few are wired. The fundamentals of both types of media are described here, which will answer questions regarding the differences between the two systems and the kind of conversions required for the converged system.

The electrical signals in an open wire line, such as a twisted pair, travel at the velocity of light, which is determined by the expression

$$v = 1/sqrt(\varepsilon\mu) \tag{1.15}$$

Analog input–digital transmission and digital input–digital transmission techniques are cascaded in practice to achieve the required form of transmission signal.

where ε and μ are the permittivity of free space (capacitance per unit length measured in farads/metre) and the permeability of free space (inductance per unit length measured in henries/metre), respectively. In free space, $v = 3 \times 10^8$ m/s, given that $\varepsilon = 9.854 \times 10^{-12}$ F/m and $\mu = 4\pi \times 10^{-7}$ H/m. The signal travels as an EM wave just outside the wires (radiation). It differs from a free space EM wave (such as the one launched by a television, radio, or mobile antenna, which spreads out in all directions) only in that it is bound to and guided by the wires of the transmission line.

Note: Metallic wired media follows the conduction theory and undergoes radiation losses, whereas fibre and wireless media follow the theory of dielectric material as per their natures and do not have radiations.

The following wired media are mainly popular:

(a) *Twisted pair* wirelines, unshielded twisted pair (UTP), and shielded twisted pair (STP), for conventional landline telephone systems, 10Base-T Ethernet cabling, and so on

(b) *Coaxial cable* for closed circuit televisions (CCTV) and cable television network, Ethernet 10Base2, 10Base5 cabling, and so on and transmission lines

(c) *Optical fibres* for long-distance communications, B-ISDN, fibre distributed data interface (FDDI), local area network (LAN), synchronous optical network (SONET), and so on

Twisted pair and coaxial cables provide a reliable, guided link that conduct an electrical signal associated with the transmission of information from one fixed terminal to another. The wires act as filters (due to lumped resistance and capacitance) that limit the maximum transmitted data rate of the channel because of band-limiting frequency response characteristics. A twisted pair wire line can typically support a 250 kbps bit rate, whereas a coaxial cable may typically support 300 Mbps. The signal passing through a wire radiates EM waves outside the wire to some extent, which can cause interference to the nearby radio signals or to other wired transmissions as a noise. These characteristics may differ from one wired medium to another. Laying additional cables in general can double the bandwidth of the wired medium.

An optical fibre is a dielectric guided medium that passes the data through itself as light waves. The carrier frequency range is of the order of 10^{14} Hz. Ideally, optical fibres have infinite bandwidth, but in practice, due to the limitations of sources and detectors and the dispersion effect, the bit rate up to Tbps (terabits per second) is achieved over high-grade

WIRED MEDIA AS TRANSMISSION LINE

When do two connecting wires become a transmission line? It is when the capacitance between the wires and the inductance of the wires acts as *distributed* instead of *lumped*. This begins to happen when the wire approaches the dimensions of a wavelength (wavelength λ and frequency f are related by $\lambda = v/f$). At sufficiently high frequencies, when the length of the connecting wires between any two devices is in the order of the wavelength or larger, the voltages and currents between these two devices act as waves that can travel back and forth on the wires. Hence, a signal sent out by one device propagates as a wave towards the receiving device and the wave is reflected unless the receiving device is properly terminated or matched. If there is a mismatch, the reflected wave can interfere with the incident wave, making communication unreliable or even impossible. Proper termination of the wired link is important when networking computers, printers, and other peripherals, which must be properly matched to avoid reflections. Transmission lines are used to carry the signal from the transmitter front end to the antenna site.

Dispersion effect is due to group delay of the multiple EM waves of the same light source propagating through the fibre and results in pulse spreading. The similar effect in a wireless channel is delay spread.

optical fibres. Optical fibres exhibit *pulse spreading effect* due to dispersion and hence, bit errors may occur. A dielectric medium allows more than one frequency to pass through it and this is the case in optical fibres in the form of *wavelength division multiplexing* (WDM). A wireless medium (which is also dielectric in nature) supports more than one frequency at a time. All links undergo the effect of white noise.

Compared to wired media, the wireless medium is unreliable; though ideally infinite, it has a low bandwidth, effectively due to the delay spread and intersymbol interference (ISI) effects. However, it supports mobility due to its *tetherless* nature. Different signals through wired media are physically conducted through different wires, but all wireless transmissions share the same medium—air—in the form of unguided EM waves released through an antenna of supporting bandwidth. Thus, it is the frequency of operation and the legality of access to the band that differentiates the variety of wireless services. Wireless networks operate in the following bands:

- 1 GHz–Cellular
- 2.4 GHz–Personal communication systems (PCS)
- 5 GHz–WLANs
- 28–60 GHz–Local multipoint distribution service (LMDS) and point to point (P2P) base station connections
- 300 GHz–Satellite ranges, infrared (IR) frequencies for optical line of sight (LoS) communication or laser communication

These bands are either licensed, such as the cellular bands, or unlicensed, such as the industrial, scientific, and medical (ISM) bands or U-NII bands used for PCS.

PROBLEM OF ELECTROMAGNETIC WAVE PENETRATION THROUGH DIFFERENT MATERIALS

As the frequency of operation and data rates increase, two simultaneous problems arise—the hardware implementation cost increases and the ability of a radio signal to penetrate walls decreases. For frequencies up to a few gigahertz, the signal penetrates through walls, allowing indoor applications with minimal wireless infrastructure inside a building. At higher frequencies, a signal generated outdoors does not penetrate into buildings and a signal generated indoors stays confined to a room. This phenomenon imposes restrictions on the selection of a suitable band for wireless application, though electronic cost has become less significant with time. Concrete and building structures are partially transparent to wavelengths of microwave range and attenuate the signal, and hence, the signal is weaker inside the buildings than outside. However, radio waves and microwaves cannot penetrate a lift (which is essentially a metal box), because the metal is a conductor and EM radiation can penetrate only a small distance into a conductor.

Capacity enhancement Wired media provide an easy means to increase capacity; we can use more wires, as and when required, if it is affordable. In contrast, in the case of the wireless medium, bandwidth is a limited resource, which imposes severe restrictions on the effective capacity. Limited bands are available for operation, and it is not possible to obtain new bands or duplicate the medium to accommodate more number of users in a system. Therefore, researchers have developed numerous techniques to increase the capacity of wireless systems to support more users with a fixed bandwidth. One such method for wireless cellular systems is *frequency reuse*, which is comparable to laying new wires in wired systems. If two cells are at a sufficient distance, then there will be no interference, even when the same frequency is used for communication in these two cells. The theory is explained in

Chapter 2. One may even reduce the size of the cells to overcome the demand of the population. In a wireless system, reducing the size of the cells by half allows twice as many users as in one cell. However, reducing the size of the cells increases the cost and complexity of the infrastructure that interconnects the cells. Multiplexing and multiple access schemes also help to accommodate more users. Capacity issues with multiple users are discussed in detail in Chapters 2 and 9 for various technologies implemented over cellular infrastructure and wireless scenario.

The capacity can be highly improved using smart antenna systems. Single input, multiple output (SIMO), multiple input, single output (MISO), and multiple input, multiple output (MIMO) systems are explained in Chapter 8. Compared to single input, single output systems, capacity increment by 300–400 per cent is possible in cellular environments with such techniques through exploiting the concepts of diversity and multipath, and then combining them. Even OFDM can support multiple users with multicarrier communication in the cellular environment.

1.7 TYPES OF WIRELESS SYSTEMS

There are three types of wireless communication systems:

(a) *Wireless broadcast systems*: The user is always at the receiver end.
(b) *Wireless networks*: Multiple users can exchange data independently being a transmitter or a receiver and share the common resources as per requirement.
(c) *Wireless navigation systems*: This is required for location-based services with the help of the global positioning system (GPS).

Modulation schemes are selected according to the suitability of the system. Wireless link requirement and protocol structures are also different.

Wireless broadcast systems These kinds of systems do not require the cellular structure or device identification numbers (except some special systems with encrypted data). Transmissions occur through a single transmitter and are of sufficiently high-power amplification. Within the predefined range, anybody can receive transmissions with the help of a receiver. These communications are mainly based on frequency tuning. Examples of such systems are AM/FM radio, television, direct-to-home (DTH), DAB, DVB, and mobile television systems.

Wireless networks These types of systems are mainly based on cellular infrastructure or ad hoc connections (forming two different types of wireless networks). Examples include mobile telephone networks, WLANs, and metropolitan area networks (MANs) for broadband access, and wireless sensor networks (in distributed configuration), which are based on cell support. For cell-based systems, at least one transceiver per cell is required, in the form of either a base station or an access point. They are low-power transmitters when compared to broadcast systems. The transmitters (or transceivers) of different cells may be interlinked to form a path between the destination and source devices. These communications are based on frequency tuning plus identification number or address. Ad hoc networks do not always require a cellular infrastructure as they are self-configurable networks. Examples of such systems include Wi-fi, Bluetooth, WiMAX, and wireless sensor networks (in centralized configuration). These networks are discussed in Chapters 11 and 12.

> Conversational cellular networks supporting data services as well have to get licensed frequency bands as they are managed by service providers, whereas ad hoc networks are self-configurable and use ISM band.

Wireless navigation systems These services are used for various applications, such as providing turn–by-turn voice-based or onscreen driving directions, automatic rerouting in case of a missed turn, real-time traffic monitoring and upgrade, alerting to slow down, and locating and navigating restaurants, Wi-fi hotspots, and maps. It is a self-correcting closed-loop system working on mobile devices. Navigation services are supported by wireless internet services.

1.8 CELLULAR NETWORKS

Cell is the basic region with a base station tower and a transceiver set having radiating power for the coverage of the basic region. A set of frequencies is allocated to the cell for communication. Multiple cells together form a cellular network. There are three types of cellular networks: cellular voice networks, cellular data networks, and cellular satellite networks. Voice networks are for conversational services, data networks are for internet access through wireless broadband services, and satellite networks are for international support to the other two networks, navigation—GPS, and so on. All the three networks support mobility. Cellular design based on some theoretical aspects is very useful in practice today, without which the existing land mobile communication would be near impossible. It forms the basis for cellular telephony. Cellular networks enable calls to be routed to and from mobile phones, even when their users are moving from one cell to another. They also enable other essential operations such as access to the network, billing, and security. To support such varied operations, a cellular network comprises many elements, each having its own function to perform.

The most important part of a cellular network is the base station with antennas and its associated equipment. To provide seamless connectivity, the system needs to have elements of central control. It also needs to link in with the public-switched telephone network (PSTN) to enable calls to be made to and from the wire-based phones, or between the networks served by different service providers.

Cellular division of an area is very useful to manage coverage, mobile device location, and handover of the services from one cell to another to have seamless connectivity during pedestrian or vehicular mobility.

Various cellular systems are available, such as the global system for mobile communication (GSM) and universal mobile telecommunication system (UMTS), and each system has its own cellular standards. For example, GSM has its own well-defined structure with which the manufacturers' products can be standardized, whereas UMTS has its own structure, standards, and protocols. Despite the differences between the different cellular systems, the basic concepts are very similar. Cellular basics and various cellular networks are explained in Chapter 2.

1.9 EXISTING TECHNOLOGIES

There is an increasing demand for broadband or wideband wireless communication systems because of the need for high-speed communications (mobile internet, wireless video transmissions etc.). At the same time, the telecommunications industry faces the problem of providing telephone services to rural areas, where the customer base is small, but the cost of installing a wired phone network is very high. One method of reducing the high infrastructure cost due to a wired system is to use a fixed wireless radio network. The disadvantage with this is that to enable the rural and urban areas to communicate, large cell sizes are required for obtaining sufficient coverage. It results in problems caused by the large signal path loss and long delay times in multipath signal propagation due to long distances. If we design more number of cells for the rural area, it would be inefficient and expensive due to the low population density. Hence, a modulation technique that covers a longer distance while eliminating the problems of a wireless channel should be introduced in the system.

Other aspects that researchers are currently working on include multipath delay compensation, speed of communication or high bit rate communication, and efficient use of available spectrum for accommodating more users and applications.

Leading Techniques of Modern Era

Several techniques play a leading role in the modernization of digital phone systems, land mobile communication, and wireless internet, with the aim of improving cell capacity, multipath immunity, security, and flexibility. Modern techniques include wideband code division multiple access (WCDMA). The latest development is the emergence of the multicarrier modulation (MCM) or multiple access technique, namely OFDM or orthogonal FDMA (OFDMA). Both these techniques could be applied to provide a fixed wireless system for rural areas. However, each technique has different properties, making it more suited for specific applications. The combinations of both these schemes are also considered to overcome the limitations and to exploit the advantages of both the systems.

The *WCDMA* technique combines two major phone technologies: code division multiple access (CDMA) and GSM. There are several key advantages of WCDMA, some of which are as follows:

- Each transmitter is assigned an identification code; hence, data from multiple transmitters can be carried over the same frequency in the same geographical area.
- It uses power control and adjusts the strength of the signal, eliminating the problem of far-off users being dominated by near users with higher signal strength.
- It is more suitable for densely populated regions and capacity enhancement as compared to CDMA.

The *OFDM* technique is for multi-user access and allows many users to simultaneously transmit in an allocated band by subdividing the available bandwidth into many narrow bandwidth carriers (described in Chapter 7). Information is allocated to several carriers in which the data is to be transmitted, so that the bits on each subcarrier are much longer, drastically reducing the ISI. Thus, it provides the concept of multicarrier modulation (multiple carriers for one digital baseband signal) rather than the conventional single-carrier modulation. The transmission is generated in such a way that the carriers used are orthogonal to one another and non-interfering with each other, thus allowing them to be packed together much closer than in standard frequency division multiplexing (FDM). This leads to OFDM providing a high spectral efficiency.

Broadcast Technologies

The main broadcast technologies are DAB and DVB, which are based on OFDM that forms the single-frequency network concept. Therefore, high-speed, high-quality communication has now become possible. Most of the applications are audio- and video-based entertainment; however, some data services are also supported.

Digital audio broadcasting is a digital radio broadcasting standard that is designed to replace the analog FM and AM radio transmissions. The development of terrestrial DAB (T-DAB) was carried out in the EUREKA 147 consortium formed by broadcasting companies, network operators, consumer electronics industries, and research institutes. The development started officially in 1987, and in 1995, the European Telecommunication Standard Institute (ETSI) standardized DAB. European Telecommunication Standard (ETS) 300–401 became the first standard to include OFDM. In 1997, the second edition of ETS 300–401 was

> Approximately 80 per cent of the world's cellular systems are based on GSM technology; most of the remaining 20 per cent are based on CDMA technology.

> The bands that are allocated for public DAB services are abbreviated as terrestrial DAB (T-DAB).

released, and the commercial employment of DAB started in 1998. Later, DAB included satellite as well as hybrid satellite or terrestrial broadcasting options. DAB is more robust against noise and multipath fading. It is based on wide-bandwidth broadcast technology and single-frequency network concept; that is, all the transmitters use the same transmission frequency with a very large coverage area.

Technically, there are two main ways of delivering mobile television in today's scenario: via two-way cellular network and via one-way dedicated broadcast network. Some examples of mobile television technologies include DVB-H, satellite digital multimedia broadcast (S-DMB), T-DMB, TDTV (based on TD-CDMA technology from IPWireless), China mobile multimedia broadcasting (CMMB), 1seg (one segment), which is based on Japan's integrated service digital broadcasting (ISDB-T), MediaFLO, general packet radio service (GPRS), and third generation (3G). DVB is a set of standards that defines digital broadcasting using existing satellite cable and terrestrial infrastructures. The DVB project consists of over 220 organizations in more than 29 countries worldwide. DVB standards are published by the Joint Technical Committee (JTC) of the ETSI, European Committee for Electro technical standardization (CENELEC), and European Broadcasting Union (EBU). DVB mostly uses moving picture experts group (MPEG) standards for the compression of audio and video signals. On the basis of distribution, there are four different standards:

- DVB-S is based on satellites
- DVB-C is based on the cable network in houses
- DVB-T is based on terrestrial transmission
- DVB-H is for audio/video streaming (H stands for hand-held) to broadcast television content to mobile devices such as personal digital assistants (PDAs) and mobile phones

Cellular Technologies

Let us have a look at some of the cellular technologies.

GSM and upgradations Currently, the GSM technology is being applied to wireless telephone systems even in rural areas. GSM900, GSM1800, and GSM1900 are the three main specifications of this technology. GSM uses frequency division multiple access (FDMA) and time division multiple access (TDMA) with frequency reuse, which has limited frequency channels to communicate. Since GSM has a high symbol rate, it leads to problems with multipath, causing ISI. Hence, there was a need for a scheme that has no ISI effects at high-speed communications. Enhanced data rate for GSM evolution (EDGE) was introduced for higher bit rate solution. Many service providers compete with each other in providing the maximum possible coverage for mobile telephony. They also try to introduce advanced services to the subscribers in order to acquire the market. Hence, EDGE technology with its high-speed support received a good response and made GSM very popular in parallel data service support.

General packet radio service is the protocol by which packet radio is made possible, and hence data services are added in the GSM system with minor modifications in the infrastructure. It is designed to have wireless web access through mobile telephony service providers.

CDMA and upgradations In CDMA systems, all users transmit in the same frequency band using specialized separate orthogonal codes as a basis of channelization (discussed in Chapter 9). The transmitted information is spread over the spectrum by multiplying it with a wide-bandwidth pseudo-random sequence. Both the base station and the mobile station know these random codes, which are used to modulate the data sent, allowing it to descramble the received signal. The use of CDMA technology started in 1990 with the IS-95 standard, which then developed to IS-95A and IS-95B with further improvements in the voice quality, bit rate, and data services. The next development was CDMA2000. It is now a challenge to cover the global wireless communication using CDMA techniques, and hence, International Mobile Telecommunications-2000 (IMT-2000) has taken up the UMTS project. Using WCDMA, standards are developed for the system even for indoor and outdoor communication. A CDMA high data rate system has been developed by Qualcomm, now called 3G 1X EV-DO, which has improved throughput and made significant enhancements in the downlink structure of CDMA2000.

Long-term evolution (LTE) This new revolutionary technology is partially commercialized. It is based on subcarrier block transmissions using OFDMA in the downlink and single-carrier FDMA (SC-FDMA) in the uplink transmissions. The research work is still going on in the LTE standard. It is an emerging high-speed wireless technology, described as the fourth-generation (4G) technology, which is based on cellular division.

All these cellular technologies are discussed in Chapter 11.

Ad hoc Networks

Ad hoc networks are the data networks established temporarily without using any infrastructure. However, they take cellular division support for some configurations in LAN and MAN. The number of users in such systems may be limited. Mostly, ad hoc networks are established for personal use or for use within a limited domain, such as an office or a plant. Due to the temporary nature of these networks and their use in personal domain for communication among personal devices, they use ISM band frequencies as their carrier frequency. ISM bands are explained in Section 1.11. Bluetooth, ultra-wideband (UWB), and ZigBee IEEE 802.15.4 (wireless sensor network) are some of the protocols for an ad hoc scenario. Wi-fi IEEE a/b/g/n is an ad hoc network with multiple configurations. The Wi-fi configuration based on the access point is similar to that of a cell because the access point acts as a base station and it has its own coverage area.

Ad hoc networks dealing with internet access follow internet protocol (IP)-based protocols. These networks allow mobility. A central challenge in the design of mobile ad hoc networks is the development of dynamic routing protocols that can efficiently find routes between two communication nodes. A mobile ad hoc networking (MANET) working group has been formed within the internet engineering task force (IETF) to develop a routing framework for IP-based protocols in ad hoc networks. Another challenge is the proper design of medium access control (MAC) protocols for multihop ad hoc networks. WiMAX IEEE 802.16x can be considered as an ad hoc network with multihop. Currently, OFDM is used as a physical layer standard in IEEE 802.11a/g/n and 802.16x protocols, HIPERLAN protocols, and so on. IEEE 802.16x are the protocols for IP-based metropolitan area broadband access networks.

Concept of Convergence in Personal Networking and Broadband Access

The aforementioned technologies can help maintain wireless connections with mobility and ensure that information is made available whenever the user requires it. However, the nature of resource and information sharing differs according to user requirements, and hence, a convergence of the technologies is required.

Convergence leads to heterogeneous networks.

Newly designed mobile devices can support many technologies in one device along with conventional mobile telephony services. The wireless technologies that are coexisting with second-generation (2G) GSM include UWB, Wi-fi, Bluetooth, and various 3G technologies, such as WCDMA and wireless access protocol (WAP). These technologies are working synergistically to meet the unique needs of the users. Apart from this, many systems require interworking among them to pass on the data to the appropriate destination.

Some examples of the convergence are as follows:

- Multiple WLANs can be connected to a WiMAX tower
- Sensor network can collect data through data aggregation techniques and send the collected data through Wi-fi to far-distance sites using broadband services
- A GSM operator can provide faster Internet services with speedy access protocols such as WAP and high speed packet access (HSPA)

A typical example of convergence using an integrated network scenario is shown in Fig. 1.5.

All these technologies and their development phases are categorized in *generations* as per the similarity in the system capabilities, bit rate support, and so on.

Table 1.2 summarizes some of the present wireless digital communication-based systems that are already in practice. Table 1.3 gives a comparison chart for the existing and upcoming technologies for wireless networking.

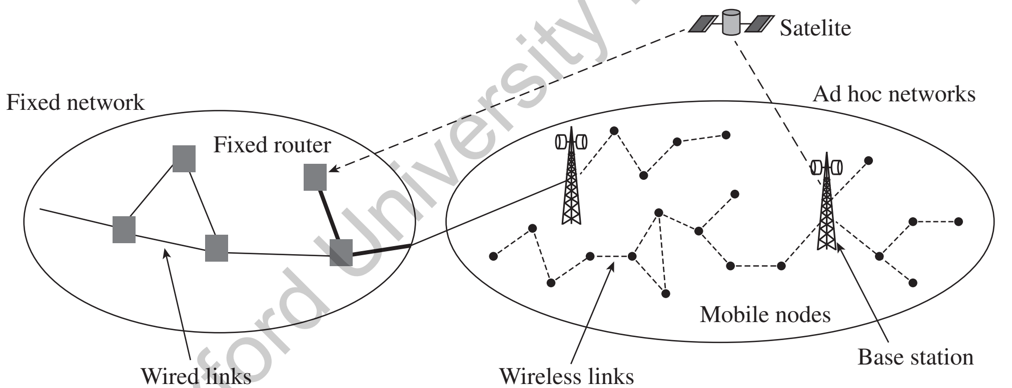


Fig. 1.5 Heterogeneous networks—integrated network combining different kinds of wireless and wired networks

Table 1.2 Summary of applications based on existing wireless digital communication

Application	**Existing standard or technology used**
Mobile telephony (digital cellular telephony)	GSM, CDMA (IS-95 to CDMA 2000), WCDMA-UMTS
WLAN/MAM/WAN	IEEE 802.11(Wi-fi), 802.16(WiMAX), etc.
Personal area communication	Bluetooth
Digital audio broadcast, HD Radio, DRM	DAB
Digital video broadcast, DTH through satellite	DVB
Mobile satellite communication, global communication	Iridium, UMTS, GPS
Mobile internet access	GPRS, Mobile IPv6, WAP, LTE
Wireless local loops	DECT, CorDECT, CDMA, GSM
Mobile ad hoc networks	All WLAN/WMAN standards and Bluetooth, sensor N/w

Table 1.3 Comparison of most-recent wireless networking technologies

	EDGE	**CDMA 2000/1 x EV-DO**	**Bluetooth**	**Wi-fi**	**Wi-fi**	**Wi-fi**	**WiMAX**	**WiMAX**	**WCDMA/ UMTS**	**UWB**	**LTE**
Standard	2.5G	3G	802.15.1	802.11a	802.11b	802.11g	802.16d	802.16e	3G	802.15.3a	4G
Usage	WWAN	WWAN	WPAN	WLAN	WLAN	WLAN	WMAN Fixed	WMAN Portable	WWAN	WPAN	WMAN/ WWAN
Throughput	Up to 384 Kbps	Up to 2.4 Mbps (typical 300–600 kbps)	Up to 720 kbps	Up to 54 Mbps	Up to 11 Mbps	Up to 54 Mbps	Up to 75Mbps (20 MHz BW)	Up to 30Mbps (10 MHz BW)	Up to 2Mbps (Up to 10 Mbps with HSDPA technology)	110–480Mbps	Typically 2–20 Mbps (RB through-put) up to 101.8 for 20 MHz carrier and 162.9 for 2 × 2 MIMO
Range	Typically 1–5 miles	Typically 1–5 miles	Up to 30 feet	Up to 300 feet	Up to 300 feet	Up to 300 feet	Typically 4–6 miles	Typically 1–3 miles	Typically 1–5 miles	Up to 30 feet	Typically 3–18 miles
Frequency	1900 MHz	400, 800, 900, 1700, 1800, 1900, 2100 MHz	2.4 GHz	5 GHz	2.4 GHz	2.4 GHz	Sub 11 GHz	2–6 GHz	1800, 1900, 2100 MHz	7.5 GHz	Multiple bands such as 700, 800, 900, 1700, 1800 MHz 2.1, 2.6 GHz etc. (Different bands for FDD and TDD modes)*

*TDD–Time Division Duplex
FDD–Frequency Division Duplex

1.10 EVOLUTION OF WIRELESS SYSTEMS

In general, the communication link requires a transmitter, a channel, and a receiver to transfer data. Here, the real-time signals and data must be modified in accordance with the channel characteristics and in a suitable detectable format, so that they can be communicated reliably through the media. Wired or wireless media can be chosen for transmission, but at the transmitter and receiver ends, a large amount of signal processing is required; hence, hardware designs need to pay more attention to the portability of the devices and should ensure good quality of reception at the same time. In the present scenario, we have a combination of systems that may have wireless infrastructure with an extensive wired support. However, the future scenario is going to be *wireless everywhere* providing the facility of mobility to the user. Hence, the following points need to be taken into consideration:

Mobility and speed of communication There is a trade off in the systems between the mobility of the user and the speed of communication achieved, as shown in Figure 1.6(b). Figure 1.6(a) represents the mobility versus data rate for various systems. It is an approximate and relative representation. It can be seen that the mobility and bit rate are increasing with the generations. The following can be observed from the figure:

- GSM provides the best mobility but very low data rate support, whereas EDGE achieves higher bit rate but compromises vehicular mobility.

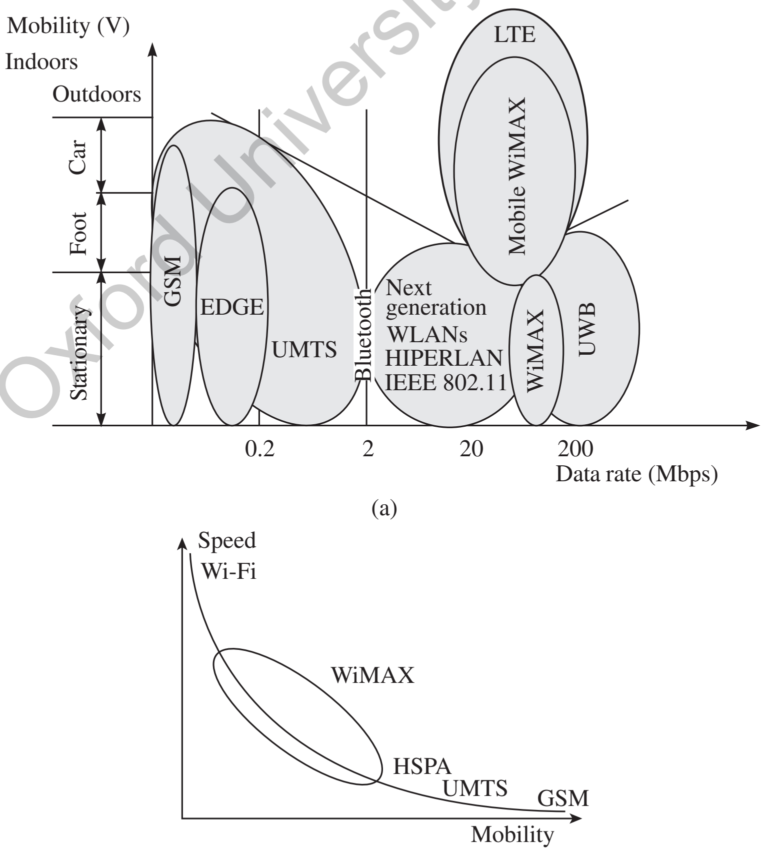


Fig. 1.6 Mobility versus data rate for various systems

The cellular infrastructure for UMTS and LTE follow almost similar architecture to that of GSM with the required upgradations in the interfaces and system components.

- Considerable mobility is achieved with UMTS with a little compromise on the bit rate.
- IEEE 802.16e mobile wireless broadband access system and LTE are found to have vehicular mobility with a higher bit rate.
- Bluetooth and Wi-fi do not require high mobility conditions, as they are small area networks and are mostly operative in personal domains. Such low-power systems perform well in terms of data rate.
- UWB gives the highest data rate.

Wireless communication versus mobile communication There exists a very thin line of difference between wireless communication and mobile communication. Basically, in wireless communication, the focus is on the main link (transmitter + channel + receiver) and its fundamentals for communication, including various blocks of processing the information signal described in Chapters 5 to 8. Here, it is necessary to know the various methods of modifying the data or real signals, modulation schemes, channel characteristics, receiving methods, and so on. Cellular theory provides the systematic platform to have the infrastructure for developing wireless communication links for multiple users without interference. Using the cell concept, users can be identified uniquely even in the mobility mode. In mobile communications, the main focus is on cell-based wireless multi-user telecommunication systems, for which standards and protocols are developed. Here, the user is assumed to be either in steady or in mobility mode.

Wireless communication aims for an optimized wireless link whereas mobile communication aims for an optimized mobile system including architecture and protocols.

Growth in hardware Wireless communications were initially developed for military purpose. Gradually, the development in computers, DSP, and chip technology enabled rapid progress in the development of portable, sophisticated wireless units, such as mobile phones as well as laptops and palmtops based on Centrino technology. DSP has become indispensable for existing wireless systems. Today's wireless communication systems are mostly based on processors, VLSI/ASIC/FPGA chips, microstrip RF circuits, and PC interface. Figure 1.7 shows that faster DSP processors (compared in terms of multi-instructions per second—MIPS) are incorporated in systems to support higher bit rate. MIPS is the measure to compute the speed of a DSP processor.

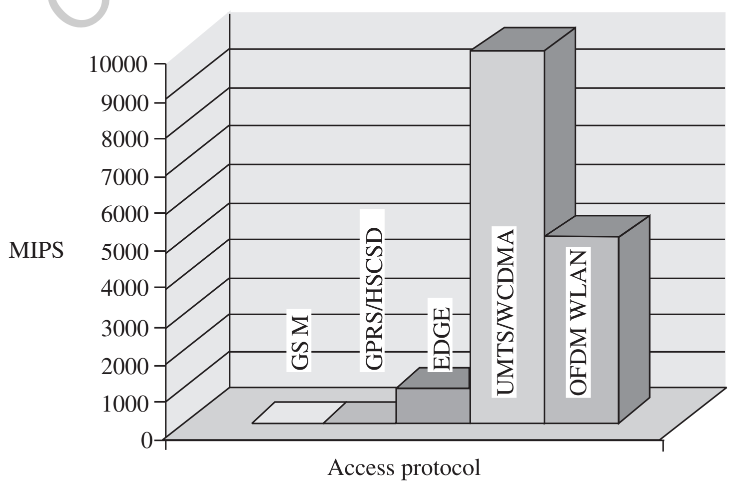


Fig. 1.7 Processing power requirement for wireless protocols and standards according to complexity of hardware

Frequency planning is required to serve millions of users in terms of different services without interference and using the limited spectrum efficiently.

Frequency planning This is necessary to have frequency planning for various wireless systems to coexist. Wireless channel is an unguided dielectric media and hence, the frequency ranges it can support are ideally infinite. Still, due to many reasons, the full available spectrum cannot be utilized. The RF and the above range utilized for wireless communication are systematically shared; different ranges are used for different applications. Various frequency ranges from the satellites provide global coverage to the cellular system, covering 50–70 km. In contrast, LANs and personal area networks (PANs) provide a maximum range of a few to hundred metres. Hence, the carrier frequency requirement also varies. If the systems are to coexist, they would obtain a crowded frequency spectrum, since there are many factors that want their share of limited frequency resource. Therefore, it is extremely important to use spectrally efficient signal strategies. The current trend to achieve high spectral efficiency is to use adaptivity on all four dimensions: time, frequency, power, and phase. The cellular theory in Chapter 2 and the multiplexing and multiple access techniques in Chapter 9 provide the best techniques of frequency planning.

Note: In short, the requirements of wireless communication include high speed/high bit rate, high spectrally efficiency, zero ISI/ICI, convergence, anywhere and anytime, global coverage, multimedia support, wireless, and digital communication systems.

Latest techniques such as WCDMA, OFDM, Hybrid OFDM, and MIMO will fulfil most of these requirements. Moreover, new approaches, such as software defined radio and cognitive radio, are coming up with a fixed set of hardware (Processor, FPGA, etc.) but with programmable software support to perform signal processing tasks, providing options such as different channel coding or different modulation scheme selection.

1.10.1 First to Fourth Generation Wireless Systems

There is no specific measure to calculate the years of generations in wireless communication. Rather, the generations are measured on the basis of the considerable innovations in the standards and applications. Analog systems are considered as the start-up and hence they are known as the first-generation (1G) systems. The systems of other generations are illustrated in Fig. 1.8.

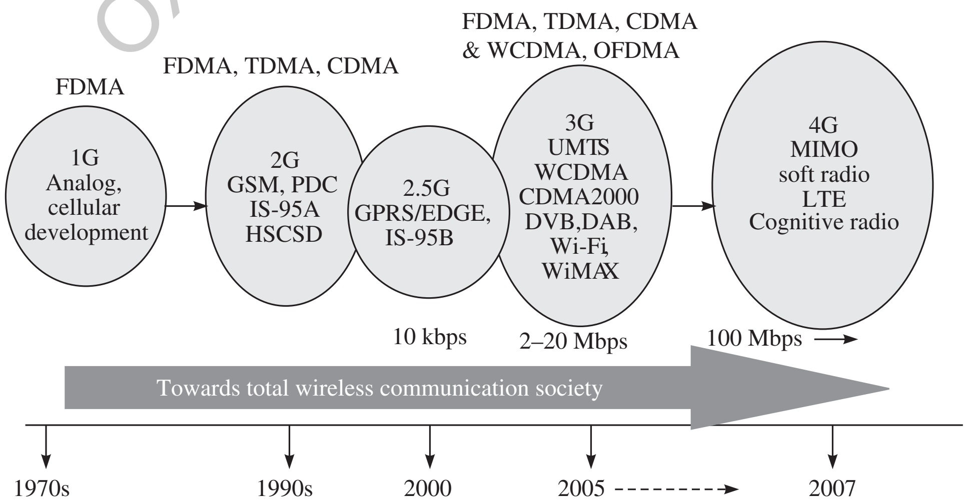


Fig. 1.8 Generations in wireless communication

4G systems are targeted with the bit rate of the order of 100 Mbps and more and that is possible due to multicarrier technique.

As mentioned, it is very difficult to distinguish the systems on the basis of generations. For simplicity, complete analog systems mainly dealing with audio (except television with analog video) are classified as 1G systems, including analog mobile phone systems (AMPS). Partially analog and digital are classified as 2G systems. In these systems, audio and images were able to communicate, and the bit rate was very low, about 10–50 kbps. Fully digital systems with audio, image, and video are classified as 3G systems. There was a tremendous rise in the bit rate, of the order of 2–20 Mbps and even up to 54 Mbps in Wi-fi and WiMAX. In 4G systems, high-speed, fully digital, anywhere, anytime, and converged wireless communication is expected with total multimedia. The expected bit rate may reach up to 100 Mbps or more in wireless environment. With evolution in WiMAX standards, UWB, and LTE, development in the 4G systems have started.

The following can be stated as the major differences in the generations:

1G—Cell structure, analog communication

2G—Cell structure, digital communication, convolution coding, power control

3G—Hierarchical cell structure, turbo coding, Hybrid Automatic Repeat Request (HARQ)

4G—Smart antenna, adaptive systems over above scenario

Why does a wireless channel face the problems of high bit rate? The channel faces the problem of delay spread due to multipath fading, meaning that the channels are time dispersive; this is discussed in detail in Chapter 3. Spreading results in merging of two consecutive pulses. If the bit rate is too high, the bit duration is low; hence, due to the merging, it is very difficult to identify the two separate pulses. This limits the bit rate of the system. Higher-order M-PSK, diversity mitigation techniques such as MIMO, or multicarrier techniques such as OFDM can eliminate the problem of higher bit rate.

The 2G technology for mobile communication originated during the 1990s, before which the conventional telephony based on wired lines was being used. A few military wireless applications and AM, FM, television, radar, and satellite communication systems were the only wireless systems implemented and known to the people. The revolution started with two new systems: the Internet based on wired lines and the cellular-based GSM that depend on wireless channels mainly for voice communication. In 2000, data transmission in the GSM was enhanced, resulting in GPRS, which could use any number of time slots among the total eight slots available for sending data. The technology exists with a data rate of 14.4–64 kbps. Another high-speed data enhancement was made in GSM, called EDGE, in which the modulation scheme is changed from *Gaussian minimum shift keying* (GMSK) to 8-PSK and the transmission data rate can be up to 500 kbps. The GSM system initially was focused on voice services with circuit switching, whereas the current 2.5G technology is focused on circuit-switched voice service and packet-switched data services.

The major challenges before the implementation of 3G were as follows:

- There was slow production of mobile phones and services.
- Wireless Internet for exponentially growing users was difficult to implement until IPv6 was implemented. (Refer to any book on computer networks for IPv6, which is the protocol for IP layer and includes IP addresses for mobile networks as well.)
- Global roaming with a single number as proposed was yet to be standardized.
- Low-cost flexible mobile devices with all desirable features were yet to evolve.

All these challenges were overcome by the scientists and engineers. The 3G systems were successfully developed, solving major problems. Now, we are into the 4G technologies, moving towards the fifth generation.

The 3G technology is optimally focused on using a single interface number and an advanced core network.

AIMS OF 3G SYSTEMS

- Anywhere and anytime mobile communication with low-cost and flexible hand-held devices
- Wireless data access, particularly with wireless Internet connection, which was motivated by the exponential growth of Internet access
- High data rate of 2 Mbps or more compared to the previous 2G systems offering 10–50 kbps
- High-speed multimedia or broadband services causing shift from voice-oriented services to Internet access (both data and voice), video, graphics, and other multimedia services
- Global roaming support and global communication
- Use of spectrum around 2 GHz and higher whereas spectrum allocation for 2G was 800/900 MHz

The 2G technology offered a quiet satisfactory voice communication, but with growing data traffic, the 3G technology has mainly targeted data services, particularly the Internet traffic. The main service component of the 3G technology is quality and reliable data traffic. The journey from 2G to 3G proceeded with an intermediate halt on 2.5G, which provides reliable services with minimal investment. The UMTS is a typical 3G system that uses WCDMA technology as mentioned previously and has the following aims:

- Data services up to 2 Mbps in rural or urban environment
- Voice over a packet-switched IP-based network
- Good spectral efficiency and low delay
- Complete mobility to the user
- Typical applications :
  - Speech—teleconferencing and voice mail
  - Message—short message service, email, etc.
  - Switched data—low-speed LAN, Internet, etc.
  - Medium multimedia—e-commerce, LAN, and Internet public messaging
  - High multimedia—video clips, online shopping, and fast LAN and Internet
- High interactive multimedia, for example, video telephony and video conferencing

Some important UMTS applications and their requirements are listed in Table 1.4.

Table 1.4 Important UMTS applications and their requirements

Applications or services	Data rate required	Quality of service required	Time critical data
Messaging (email, etc.)	Low (1–10 kbps)	High	No
Voice	Low (4–20 kbps)	Low (BER < 1e-3)	Yes
Web browsing	As high as possible (>10–100 kbps)	High (BER < 1e-9)	Depends on the material; generally not time critical
Videoconferencing	High (100 kbps–2 Mbps)	Medium	Yes
Video surveillance	Medium (50–300 kbps)	Medium	No
High-quality audio	High (100–300 kbps)	Medium	Yes
Database access	High (>30 kbps)	Very High	No

1.10.2 Beyond Third Generation

During the past 20 years, wireless networks have evolved from the analog, single-medium (voice), and low data rate (few kbps) systems to the digital, multimedia, and high data rate (10–100 Mbps) systems of today.

The International Telecommunication Union (ITU) in July 2003 had made the following requirements for a 4G system:

- At a standstill condition, the transmission data rate should be 1 Gbps.
- At a moving condition, the transmission data rate should be 100 Mbps.

With these high-speed data systems, it is possible to provide users many advanced applications, such as video streaming. A potential 4G system could be used in the family of OFDM, because OFDM can have a transmission data rate of 54–70 Mbps, which is much higher than what a CDMA system can provide. A comprehensive, integrated broadband mobile communication will step forward into all-mobile 4G service and communication. The 4G technology is developed to provide high-speed transmission, next-generation Internet support (IPv6, VOIP, and mobile IP), high capacity, seamless integrated services and coverage, utilization of higher frequency, low mobile cost, efficient spectrum use, quality of service and end-to-end IP system. In short, the 4G requirements are as follows:

- High-speed data communication
- Best quality voice
- Multimedia on mobile
- LAN and intranet or Internet on mobile

1.11 LICENSED AND UNLICENSED BANDS FOR EXISTING WIRELESS SYSTEMS

Wireless channel is shared by a number of users, and the frequency ranges are provided systematically to the users, services, or applications for reliable communication (refer to Chapters 2 and 9). A few frequencies are allocated to the cellular mobile operators, such as Airtel, Hutch, or Idea, who pay heavy charges for using the allocated ranges. Even satellite channels are paid channels because of this reason. Mobile operators cannot invest in huge private infrastructure, such as satellites; moreover, they have to follow government rules. Hence, they have to get the licensed bands for communication. Mobile communications based on GSM and CDMA are made over licensed bands.

Presently, some technologies limited to the user's area without the need for huge or global infrastructure are developed. Some applications of these technologies are PAN, based on Bluetooth, UWB, and WLAN, based on Wi-fi, which are small area communication systems. The frequency range of operation is 2.4–5.6 GHz. Actually, these bands are international bands for scientists and medical officers. As the systems are not concerned with other such systems at far distances, independent communication is possible. For example, in Bluetooth applications, one device with Bluetooth support will search for other active Bluetooth devices within an area of 10 metres. The list of devices will be displayed on the screen and the required device can be selected from the list for communication. Even if any other Bluetooth device is active beyond this range, it will not be listed or connected with the device. These communications are called *unlicensed band communications*. Since they are based on spread spectrum or OFDM technology, secure communication is possible. In spread spectrum techniques, orthogonal codes are present, whereas in OFDM, orthogonal carriers are present.

> Ad hoc networks are operated in the unlicensed band, whereas infrastructure-based cellular network operators need to pay for the licensed frequencies.

1.11.1 Spectral Policies

There is a rapidly increasing growth of wireless services as well as development of new technologies. Consequently, the demands on the use of the RF spectrum are rapidly increasing for both the federal government and non-federal users. The spectrum is heavily occupied in the 0.8–11 GHz range for land mobile systems and 3–30 GHz for television and satellite ranges. An approximate representation is given in Fig. 1.9. The spectrum is shared among many service providers. The services include defence and military applications too, and therefore, the spectrum must be managed with certain policies. There is a continuous revision in the spectrum management policies to satisfy domestic and international uses to cope with the latest development and usage scenario. The US, the UK, and many other countries have their own body to manage such concerns and to take specific actions to improve the spectrum management. Policies vary from country to country.

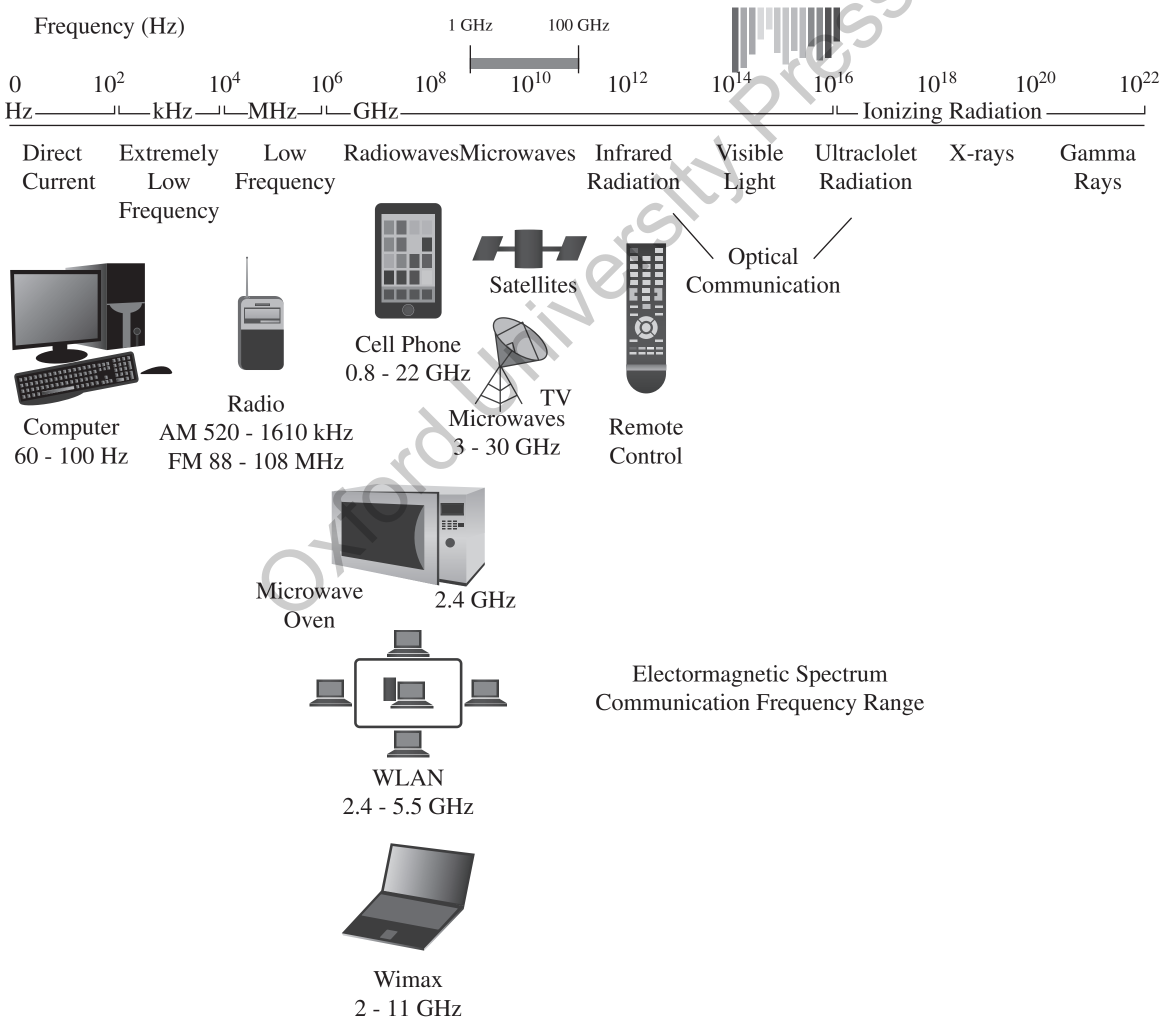


Fig. 1.9 Ready reference for spectrum occupancy

1.12 CLASSIFICATION OF WIRELESS SYSTEMS BASED ON RANGE

Based on range, wireless systems may be categorized into short range systems and long range systems.

1.12.1 Short-range Wireless

A number of wireless technologies have been developed for communication over very short distances. These are referred to as 'short-range wireless communication'. In these cases, signals travel from a few centimetres to several metres. Examples of short-range wireless communications are Bluetooth, infrared, near field communication, UWB, and ZigBee.

Bluetooth The Bluetooth technology is used to transfer data wirelessly over short distances from fixed and mobile devices. It operates at frequencies between 2.402 and 2.480 GHz including guard bands 2 MHz wide at the bottom end and 3.5 MHz wide at the top. The frequency band ranging between 2.402 and 2.480 GHz is a globally licenced band. It employs a radio technology called frequency-hopping spread spectrum. The devices based on Bluetooth technology divides the transmitted data into packets which is further transmitted on packet basis on one of several designated Bluetooth channels. Each channel has a bandwidth of 1 MHz. It usually performs 800 hops per second with Adaptive Frequency-Hopping (AFH)-enabled technology. The Bluetooth uses 2 MHz bandwidth with 24 channels.

The Bluetooth technology exists in many products including mobiles, tablets, media players, gaming equipment, and automobiles. This technology is especially useful when data is to be transferred within near regions.

Infrared Data Association (IrDA) It is very popular technology for short range communication for efficient data transfer system. The most popular applications of inferred data communication includes mobile phone, televisions, laptop computers, and so on. In fact, the remote controls of televisions merely transmit the controlling signals using infrared technology with the help of infrared light-emitting diode (LED) installed at the tip of infrared remote control. Here, the host device, that is, the television, is always ready for receiving the infrared signal sent from its remote. However, it may not be the case for other devices such as laptop computers. In laptop computers, the establishment of link between two computers for data transmission is completed when they not only recognize each other but also allow accepting the infrared signal sent by counterpart laptop computers. A similar mechanism is used by mobile phones also for exchange of data between each other.

Ultra-wideband technology The need of errorless communication has always been the mother for development of new technologies for communication of data between devices. Researchers have also disclosed in literatures that if the bandwidth of any communication systems is increased, the probability of errorless communication is also increased.

The UWB communication technology has been developed for efficient communication of data over a wide range of frequency bands, consuming very low power for a short distance. This technology is employed for the range of few metres only; however, its bandwidth of communication is quite high.

The UWB broadcasts digital pulses that are timed very precisely on a carrier signal across a very wide spectrum at the same time. The transmitter and receiver for UWB communication must be coordinated to send and receive pulses with an accuracy of trillionths of a second due to very wide bandwidth of signal.

ZigBee technology This technology is also utilized for short range communication. This technology was named after the fact that bees pass messages to each other during dancing process, employing multiple loops while flying.

The ZigBee technology is based on IEEE 802.15.4 short-range communications standard. This technology has been designed for low-power, low-cost applications, but on the other hand, it also provides low-throughput. The main merit of this technology is that it supports for mesh topologies, which provide the basis for the fact that ZigBee devices relay messages for each other through multiple wireless hops as the bees do for passing the messages between each other.

1.12.2 Long-range Wireless Technologies

Here, signals in medium-range wireless communication travel up to 100 metres or so, whereas signals in wide-area, wireless communication can travel from several kilometres to several thousand kilometres.

There are many long-range wireless technologies including WLANs, cellular systems, wireless metropolitan area networks, and satellite communication systems.

WLANs A WLAN is a very popular technology in which mobile users are connected with each other for passing information by forming a LAN through a wireless (radio) connection.

Cellular systems This technology can cover wide range of communication between two or more devices efficiently. As illustrated in Fig. 1.10, a basic cellular system consists of three parts:

- Mobile units
- Cell site (it contains a control unit, a radio cabinet, antenna, data terminal, and power planet.)
- Mobile Telecommunication Switching Office (MTSO: It is the central coordinating element for all cell sites and contains the cellular processor and a cellular switch.)

Wireless metropolitan area networks A wireless metropolitan area network (WMAN) is popularly known as wireless local loop (WLL). In India, it was introduced during the year 2000. This technology has, in fact, introduced the Indian community to use wireless services over a wide distance, which was quite more than conventional cordless phones.

This technology employs the IEEE 802.16 standard and can transfer the data with transfer speeds of 1 to 10 Mbps within a range of 4 to 10 kilometres. This technology is not being used by larger population after the induction of mobile phones in India.

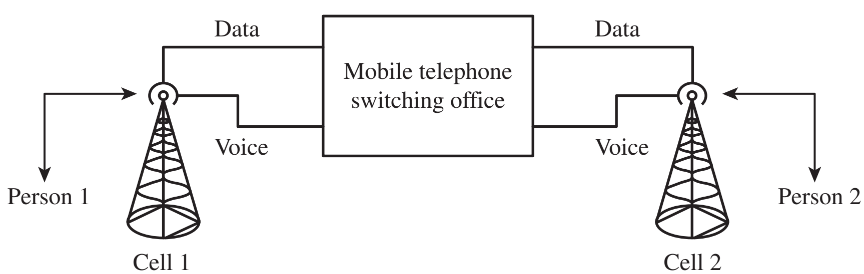


Fig. 1.10 Basic Cellular System

Satellite Communication Systems

In satellite communication system, the data transmission is performed with the help of satellites. Here, the sender and the receiver may be far from each other and the satellite becomes a medium of conveying the amount of information between them.

1.13 CLASSIFICATION OF WIRELESS CHANNEL BASED ON ENVIRONMENT

The communication channels, which act as interfaces between the transmitter and the receiver, may be a pair of wires, wireless system, or any other systems through which the data pass. These channels may carry the information in the form of electrical signals, optical signals, acoustic signals, etc.

The noise in any communication channel mainly distorts the signal. The effect of noise can be minimized by increasing the power in the transmitted signal. This mechanism may become impractical in certain environments where it is not possible to replace the battery very often. The Shannon's theorem results also relate the channel capacity to the available transmitted power and channel bandwidth.

In the following section, some of the important communication channels are being presented in brief based on environment.

Wire line channels In this case, the communication of data takes place in the wire line. In fact, the telephone network makes extensive use of wire lines for voice signal transmission, as well as data and video transmission. The wire line channels may be broadly divided into two categories, namely twisted-pair wire lines and coaxial cable wire lines.

The above said wire line channels are basically guided electromagnetic channels that provide relatively modest bandwidths. Telephone wires are generally used to connect a consumer to a central office, and it has a bandwidth of several hundred kilohertz (kHz). On the other hand, coaxial cable has a usable bandwidth of several megahertz (MHz) and is usually used for transmission of video and other data.

Signals transmitted through such channels are distorted in both amplitude and phase, which may be further corrupted by additive noise. Twisted-pair wire line channels are also prone to crosstalk interference from physically adjacent channels. Because wire line channels carry a large percentage of our daily communications around the country and the world, therefore, much research has been performed on the characterization of their transmission properties and on methods for mitigating the amplitude and phase distortion encountered in signal transmission.

Fibre optic channels The optical fibres offer the communications system designer a channel bandwidth which is several orders of magnitude larger than the coaxial cable channels. During the past decade, the optical fibre cables have been developed which have relatively low signal attenuation. In addition, highly reliable photonic devices have been developed for signal generation and signal detection. These technological advances have resulted in a rapid deployment of optical fibre channels both in domestic telecommunication systems as well as for transatlantic and transpacific communications. With the large bandwidth available on fibre optic channels, it is now possible for the telephone companies to offer subscribers a wide array of telecommunication services including voice, data, facsimile, video, etc.

The transmitter or modulator in a fibre optic communication system is a light source, either a LED or a laser. The information is then transmitted by varying (modulating) the intensity of the light source in accordance with the message signal. This light wave propagates through the fibre and is amplified periodically (in the case of digital transmission, it is detected and regenerated by repeaters) along the transmission path to compensate for signal attenuation. At the receiver end, the light intensity is duly detected by a photodiode whose output is an electrical signal that varies in direct proportion to the power of the light impinging on the photodiode.

In India, it is envisioned that optical fibre channels will replace nearly all wire line channels in the telephone network in the next few years.

Wireless electromagnetic channels In radio communication systems, electromagnetic energy is coupled to the propagation medium by an antenna that serves as the radiator for the energy to be transmitted or radiated. The physical size and the configuration of the antenna depend primarily on the frequency of operation, that is, wavelength.

The mode of propagation of electromagnetic waves in the atmosphere and in free space may be subdivided into three categories, namely ground-wave propagation, sky-wave propagation, and LoS propagation.

In the VLF and ELF frequency bands, where the wavelengths exceed 10 km, the earth and the ionosphere act as a waveguide medium for electromagnetic wave propagation. In these frequency ranges, communication signals practically propagate around the globe in a circular manner. For this reason, these frequency bands are primarily used to provide navigational aids from shore to ships, around the world. The channel bandwidths available in these frequency bands are relatively small (usually from 1–10% of the centre frequency), and hence, the information that is transmitted through these channels is relatively slow speed and generally confined to digital transmission. A dominant type of noise is generated at these frequency levels from thunderstorm activity around the globe, especially in tropical regions. Interference results due to many users of these frequency bands.

Ground-wave propagation is the dominant mode of propagation for frequencies in the MF band (0.3–3 MHz). This is the frequency band used for AM broadcasting and maritime radio broadcasting. In AM broadcast, the range with ground-wave propagation of even the more powerful radio stations is limited to about 100 miles. Atmospheric noise, man-made noise, and thermal noise from electronic components at the receiver are dominant disturbances for signal transmission of MF.

When transmitted signals are being reflected (bent or refracted) from the ionosphere, which consists of several layers of charged particles ranging in altitude from 30–250 miles above the surface of the earth, it results in Sky-wave propagation. During the daytime hours, the heating of the lower atmosphere by the sun causes the formation of the lower layers at altitudes below 75 miles. As a result, these lower layers, especially the D-layer, absorb frequencies below 2 MHz, thus severely limiting sky-wave propagation of AM radio broadcast. However, during the night-time hours, the electron density in the lower layers of the ionosphere gets dropped sharply and the frequency absorption that occurs during the daytime is reduced significantly. As a result, the powerful AM radio broadcast stations, therefore, can propagate over large distances via sky-wave over the F-layer of the ionosphere, which ranges from 90–250 miles above the surface of the earth.

However, signal multipath fading is a commonly occurring problem with electromagnetic wave propagation via sky wave in the HF frequency range. The signal multipath fading generally occurs when the transmitted signal arrives at the receiver via multiple propagation paths through different delays that results in ISI in a digital communication system. Moreover, the signal components arriving via different propagation paths may add destructively, resulting in a phenomenon called signal fading, which most people have experienced when listening to a distant radio station at night, when sky wave is the dominant propagation mode. It has also to be noted that additive noise at HF is a combination of atmospheric noise and thermal voice. Further to quote that the sky-wave ionospheric propagation ceases to exist at frequencies above approximately 30 MHz, which is the end of the HF band. However, even it is possible to have ionospheric scatter propagation at frequencies in the range of 30–60 MHz, resulting from signal scattering from the lower ionosphere.

It is also possible to communicate over distances of several hundred miles by the use of tropospheric scattering at frequencies in the bandwidth of 40–300 MHz. The troposcattering process results due to signal scattering from the particles in the atmosphere at altitudes of 10 miles or less. The general demerit of such communication is that the ionospheric and tropospheric scatterings involve large signal propagation losses and require a large amount of transmitter power and relatively large antennas. The frequencies ranging above 30 MHz propagate through the ionosphere with relatively little loss and make satellite and extra-terrestrial communications possible for communication purpose. Hence, at frequencies in the VHF band and higher, the dominant mode of electromagnetic propagation is always LoS propagation. For terrestrial communication systems, it may be concluded that the transmitter and receiver antennas must be in direct LoS with relatively little or no obstruction. For this reason, the television stations transmitting in the VHF and UHF frequency bands, mount their antennas on high towers and hills in order to achieve a broad coverage area.

In general, the coverage area for LoS propagation is mainly limited by the curvature of the earth. Consider the case when the transmitting antenna is mounted at a height *h* feet above the surface of the earth, the distance to the radio horizon, assuming no physical obstructions such as mountains, is approximated at $d = \sqrt{2(h_{\text{antenna}})}$ miles where height of the antenna is mentioned in feet. As an example, a TV antenna mounted on a tower of 1000 feet in height provides coverage of approximately 50 miles. In another example, microwave radio relay systems used extensively for telephone and video transmission at frequencies above 1 GHz have antennas mounted on tall towers or on the top of tall buildings or hills.

At frequencies above the EHF band, the infrared and visible light regions of the electromagnetic spectrum may be used to provide LoS optical communication in free space. Nowadays, these frequency bands are being used in experimental communication systems such as satellite-to-satellite links.

Wireless underwater communication The communication below the water surface is as important as the communication over the earth surface. The requirement for underwater (UW) wireless communication exists in applications including remote control in off-shore oil industry, pollution monitoring in environmental systems, speech transmission between divers, processing of the ocean floor for objects detection, collection of scientific data recorded at ocean-bottom stations and by unmanned underwater vehicles, and security of national territory in the sea area. The known depth of sea is approximately 11,000 metres, whereas the submarines and even whales cannot go below the depth of 500 metres in the sea. Therefore, considering the fact that sea area is approximately 70% of earth surface a lot of ecological and mining survey is required in the sea world. This may be only archived with the help of UW communication.

The wireless underwater communications can be established by transmission of acoustic waves in the water channel. The radio waves are of little use in such channel because they are severely attenuated due to characteristics of sea water with smaller wavelength of signal. The optical wave communications in sea water suffer from severe scattering, and hence, optical wave communications need high precision in pointing the laser beams. Underwater acoustic communication channels are very complex in nature due to several parameters including multipath, salinity of water, existence of water columns, etc. They have limited bandwidth and often cause severe signal dispersion in time and frequency.

UW System Requirements

The required data is collected by submerged acoustic instrument such as hydrophones, seismometers, sensors, current-meters, chemical sensors, in addition to other sensors. Data rates on the order of one to several tens of kbps are required for these applications. The reliability requirements are not as stringent as for the command signals, and a probability of bit error of $10^{-3} - 10^{-4}$ is acceptable for many of the applications.

Factors Influencing Acoustic UW Communication

Path loss The path loss in UW communication is mainly caused by three phenomena:

- Geometric spreading loss
- Attenuation
- Noise
- Multipath distortion

The transmission loss for a signal of frequency f [kHz] over a transmission distance d [m] can be expressed in [dB] as:

$$10 \log TL\,(d, f) = ks \cdot 10 \log (d) + d \cdot (f) + Atm \tag{1.16}$$

where k is the spreading factor, which describes the geometry of propagation, (f) [dB/m] is the absorption coefficient, and Atm [dB] is the so-called transmission anomaly that accounts for factors other than absorption including multipath propagation, refraction, diffraction, and scattering; typically, spreading loss depends only on propagation range, and therefore, it is frequency independent.

Geometric spreading In general, the geometric spreading occurs due to reflections of signal from various geometries present in the sea area.

Spherical geometric spreading It occurs when acoustic waves spread spherically outwards from a source in an unbounded medium that characterizes deep water communications.

Cylindrical geometric spreading This types of geometric spreading occurs when acoustic waves spread horizontally due to a medium that has parallel upper and lower bounds with characteristics of shallow water communications. The spreading factor, k, is equal to 1 for cylindrical and 2 for spherical spreading. In practice, a spreading factor of $k = 1.5$ is often considered.

Attenuation The attenuation below sea surface may be mainly concerned to absorption caused by conversion of energy of the propagating acoustic wave into heat (also referred to as absorption loss). The absorption coefficient for frequencies above a few hundred Hertz can be expressed empirically using Thorp's formula,which defines $\alpha(f)$ [dB/m] as a function of f [kHz].

$$\alpha(f) = (0.11 f^2/ f^2 + 1 + 44(f^2/ f^2 + 4100) + 2.75 \cdot 10^{-4}\ f^2 + 0.003) \cdot 10^{-3} \quad (1.17)$$

For lower frequencies, the absorption coefficient can be expressed as:

$$\alpha(f) = (0.002 + 0.11(f^2/f^2 + 1) + 0.011f^2) \cdot 10^{-3} \quad (1.18)$$

Noise The noise in UW communication may be natural or man-made. The latter is mainly caused by machinery noise (pumps, reduction gears, and power plants) and shipping activities, whereas the former is produced by biological, seismic activities and hydrodynamics (waves, currents, tides, rain, and wind) in addition to noises made by fish and others. The contributions of the major noise sources can be expressed through empirical formulae, which provide power spectral densities of each source relative to the frequency f [kHz] in [dB Hz].

$$10 \log N_{\text{turbulence}}(f) = 17 - 30 \log f$$
$$10 \log N_{\text{shipping}}(f) = 40 + 20(s - 5) + 26 \log f - 60 \log(f + 0.03)$$
$$10 \log N_{\text{wind}}(f) = 50 + 7.5w1/2 + 20 \log f - 40 \log(f + 0.4)$$
$$10 \log N_{\text{thermal}}(f) = -15 + 20 \log f$$

Where N_t, N_s, N_w, and Nth stand for turbulence, shipping, wind, and thermal noises, respectively. The total noise power spectral density for a given frequency f [kHz] is then given as:

$$N(f) = N_{\text{turbulence}}(f) + N_{\text{shipping}}(f) + N_{\text{wind}}(f) + N_{\text{thermal}}(f) \quad (1.19)$$

In shallow water, the noise is difficult to model or predict when compared to the deep water case due to availability of sea surface, and it demonstrates greater variability in both time and location. Here, the three major noise sources in shallow water environments are identified as wind noise, biological noise (especially noise created by snapping shrimp whose noise signature has a high amplitude and wide bandwidth), and shipping noise.

Multipath propagation The multipath propagation arises from either wave reflections from the surface, bottom, and other objects, or wave refraction caused by sound speed variations with depth (acoustic waves always bend towards regions where the propagation speed is lower). Multipath propagation can severely deteriorate the acoustic signal, which may result in multipath fading due to ISI. The multipath geometry below the sea water depends on the link configuration. The vertical channels typically have little time dispersion, whereas horizontal channels may show long multipath spreads in addition to distortion caused by big moving objects including fishes. The extent of spreading is highly dependent on depth and distance between the transmitter and the receiver. The channel impulse response for a time-varying multipath underwater acoustic channel can be expressed as:

$$C('I, t) = {}_p\Sigma Ap(t)\delta('I - 'I_p(t)) \quad (1.20)$$

Where $A_p(t)$ and $'I_p(t)$ denote time-varying path amplitude and time-varying path delay, respectively. This expression can be used in simulation studies and in developing receiver algorithms.

Doppler Frequency Spread

It is the range of frequency which the users may experience in terms of variation in frequency while they are travelling towards or away from each other. It is denoted as B_d. The Doppler spread can be represented in time by the inverse of coherence time of the channel, which is given by

$$\Delta t_c \approx 1/B_d \tag{1.21}$$

In fact, the Doppler spread occurs as a result of Doppler shifts caused by motion at the source, receiver, and channel boundaries in addition to other parameters. When a channel experiences a Doppler spread with a bandwidth B and if a transmitted signal has symbol duration of T, then there will be BT uncorrelated samples of its complex envelope. If BT is much less than unity, the channel is called to be under spread, and Doppler spread effect scan be basically ignored. If Doppler spread is greater than unity, it is said to be overspread.

The Doppler spread can be significant in UWA (Under Water Acoustic) channels, thus causing degradation in the performance of digital communications. The ISI occurs at the receiver with high data rate transmission due to multipath fading phenomena. The Doppler spreading also generates different effects on signals including a simple frequency translation, which is relatively easy for a receiver to compensate for and a continuous spreading of frequencies that creates a non-shifted signal.

MORE SOLVED EXAMPLES

Example 1.4 Identify the type of signal (energy or power) shown in Fig. 1.11 and calculate the suitable measure.

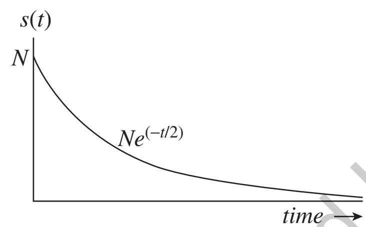


Fig. 1.11 Exponential function for Example 1.4

Solution The signal shown in Fig. 1.11 is an exponentially decaying signal that approaches zero as t approaches infinity. It is not a periodic signal. Hence, it is an energy signal. The suitable measure is energy E_s. From Eq. (1.13a), for a continuous signal

$$E_s = \int_{-\infty}^{\infty} s^2(t)dt = \int_{0}^{\infty}(Ne^{-t/2})^2 dt = \int_{0}^{\infty} N^2 e^{-t} dt = N^2$$

(*Note*: Readers can try to identify from various other functions whether it is a power signal or an energy signal.)

Example 1.5 Show that the frequency spectra of the square wave shown in Fig. 1.12, is sinc shaped. [Hint: sinc function is of the form $\sin(x)/x$.]

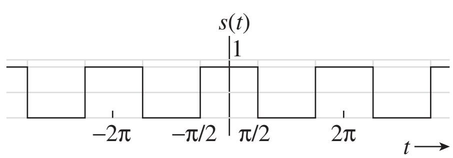


Fig. 1.12 Square wave for Example 1.5

Solution It is better to represent the square wave $s(t)$ in its exponential Fourier series form to get the frequency domain coefficients and the symmetrical form of the spectra.

The signal $s(t)$ can be written in its exponential series form as

$$s(t) = \sum_{n=-\infty}^{\infty} c_n e^{jn2\pi f_0 t}$$

where $f_0 = 1/T_0$, and T_0 is the duration 2π.

$$C_0 = \frac{1}{T_0}\int_{T_0} s(t)dt = \frac{1}{2}$$

$$c_n = \frac{1}{T_0}\int_{T_0} s(t)e^{-jn2\pi f_0 t}dt \; n \neq 0$$

$$= \frac{1}{T_0}\int_{-T_0/4}^{T_0/4} e^{-jn2\pi f_0 t}dt$$

$$= \frac{1}{-jn2\pi f_0 T_0}[e^{-jn2\pi f_0 T_0/4} - e^{jn2\pi f_0 T_0/4}]$$

Rearranging the terms as per the definition of sine wave in terms of the exponential form and substituting $f_0 T_0 = 1$, we get

$$c_n = \frac{1}{n\pi}\sin\left(\frac{n\pi}{2}\right)$$

This is the mathematical representation of a sinc function.

Now, for $\quad n = 1, c_n = \frac{1}{\pi}\sin\left(\frac{\pi}{2}\right) = \frac{1}{\pi}$

for $n = 2,\ c_n = \frac{1}{2\pi}\sin\left(\frac{2\pi}{2}\right) = \frac{1}{2\pi}\sin\pi = 0$

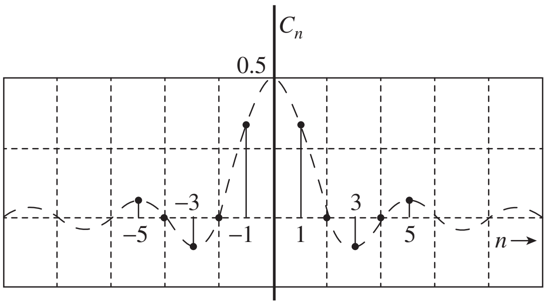


Fig. 1.13 Result of Example 1.5

and so on. On plotting the discrete components and their values and joining them, the since shape is obtained (Fig. 1.13).

Example 1.6 Find the Fourier transform of the unit impulse signal $\delta(t)$.

Solution $s(t) = \delta(t)$, where $\delta(t)$ occurs at a time instant $t = 0$

or $s(f) = \int_{-\infty}^{\infty} \delta(t)e^{-j2\pi ft}dt = e^{-j2\pi f\cdot 0} = 1$

or $\delta(t) \Longleftrightarrow 1$

It occupies the whole spectrum. This property can be observed in AWGN noise (Chapter 4).
(*Note*: The various useful Fourier transforms for communication systems are for rectangular pulse function, sinc function, and sinusoidal wave. Readers may go through them as a self-study.)

Example 1.7 Establish the relationship of E_b/N_o with the channel capacity.

Solution As per Shannon–Hartley theorem,
Channel capacity $C = W\log_2(1 + S/N)$
Signal power = Bit energy $E_b \times$ bit rate C
Noise power = Noise spectral density $N_o \times$ bandwidth W
Substituting the values, we get

$$E_b/N_o = [2^{C/W} - 1]/(C/W)$$

SUMMARY

- The bit rate defines the rate at which information is passed, whereas the signalling rate defines the baud rate. The symbol rate is the number of symbols per second; each symbol represents n bits and has M signal states, where $M = 2^n$. This is called M-ary signalling. Baud is synonymous to symbols per second or pulse per second.
- Using the transforms, any time domain signal can be analysed into its frequency components. For every signal, the signal defines the spectrum and the spectrum defines the signal; that is, they are unique and opposite conversions ideally (especially for linear systems) but may not be so practically.
- Bandwidth gives important information about useful frequency components.
- The SNR and bandwidth are exchangeable and are to be balanced always to decide the channel capacity.
- Voice, video, and other real-time signals are energy signals.
- The final RF transmission is always in the analog form, but the baseband signal inputted to the modulation stage decides whether the wireless communication link is an analog link or a digital link.
- A wireless link transmitter employs source coding, channel coding, modulation, and upconversion, and the opposite blocks are at the receiver side.
- With different combinations of coding and modulation schemes, different responses of the wireless systems can be observed. Hence, the selection of an optimum set-up of the protocols and standards is a matter of balancing the requirements.
- Line coding is applied to digital baseband for obtaining the desired spectral characteristics.
- OFDM and CDMA are the important modulation techniques for the latest wireless systems and for next-generation networks.
- GSM is the first digital wireless system, which was then upgraded to EDGE and is supported by the GPRS packet radio protocol.
- UMTS targets worldwide mobile communication with a unique user number.
- WPAN, WLAN, and WMAN are three major networks with different sizes and are based on the IP protocol.
- LTE is a 4G network based on OFDMA.
- The major systems in the broadcast technologies are DAB and DVB.
- Unlicensed (ISM) band communications are allowed only for personal area communication systems such as Bluetooth and are operated at 2.4–5.6 GHz. Infrastructure-based mobile networks use license bands, in which frequencies are planned out for the coexistence of the systems.

EXERCISES

Multiple-choice Questions

1.1 If the transmission bandwidth is W and the available channel bandwidth is W_{channel}, what should be the condition that will allow fruitful reception?
(a) $W = W_{\text{channel}}$
(b) $W < W_{\text{channel}}$
(c) $W > W_{\text{channel}}$
(d) All of these

1.2 If the bit rate of a data is 1 Mbps, what should be the bandwidth occupied by the rectangular wave?
(a) 1 MHz
(b) 0.1 MHz
(c) 0.5 MHz
(d) 2 MHz

1.3 Real audio/video signal is a/an
(a) energy signal
(b) power signal
(c) deterministic signal
(d) periodic signal

1.4 Unit ramp signal is
(a) an energy signal
(b) a power signal
(c) a periodic signal
(d) none of these

1.5 Which of the following measures cannot be effective in reducing noise?
(a) Decrease in signalling rate
(b) Increase in channel bandwidth
(c) Increase in transmitter power
(d) Use of redundancy

1.6 The channel capacity C of a band-limited Gaussian channel is defined as
(a) $W_{\text{channel}}\log_2(1 + SNR)$
(b) $(1/W_{\text{channel}})\log_2(1 + SNR)$
(c) $W_{\text{channel}}\log_2(SNR)$
(d) $(1/W_{\text{channel}})\log_2(SNR)$

1.7 In communication receivers, fidelity is provided by the
(a) mixer stage
(b) audio stage
(c) IF stage
(d) detector stage

1.8 If a receiver has poor IF selectivity, it will, therefore, also have poor
(a) sensitivity
(b) double spotting
(c) diversity reception
(d) blocking

1.9 Noise figure is used as a figure of merit of a/an
(a) oscillator
(b) modulator
(c) amplifier
(d) isolator

1.10 The selectivity of most receivers is determined largely by the
(a) sensitivity
(b) antenna direction
(c) characteristics of IF section
(d) all of these

1.11 Which one of the following is not a useful quantity for comparing the noise performance of receivers?
(a) Noise figure
(b) Equivalent noise resistance
(c) Input noise voltage
(d) Noise temperature

1.12 Which of the following communication systems is mainly suitable for wireless digital communication?
(a) Analog input–analog transmission
(b) Analog input–digital transmission
(c) Digital input–digital transmission
(d) Digital input–analog transmission

1.13 Which of the following is the scheme for creating a digital database of real signals?
(a) Pulse code modulation
(b) Manchester coding
(c) Binary conversion
(d) Pulse amplitude modulation

1.14 Which of the following systems is a 3G system?
(a) Analog cellular system
(b) EDGE
(c) FM
(d) UMTS

1.15 The capacity of a wireline system can be increased by
(a) TDMA
(b) random access
(c) increasing the number of wires
(d) all of these

1.16 The protocol for a Wi-fi system is
(a) IEEE 802.16d
(b) IEEE 802.15.3
(c) IEEE 802.11a
(d) IEEE 802.15.1

1.17 Which of the following is a system in which long haul communication is involved?
(a) Mobile satellite communication system
(b) GSM system
(c) WiMAX system
(d) Bluetooth system

1.18 The systems that utilizes the ISM band for communication are
(a) GPRS and EDGE
(b) Bluetooth and Wi-fi
(c) GPRS and Bluetooth
(d) Bluetooth and WiMAX

Review Questions

1.1 How are the communication systems classified in general?

1.2 How are the wireless systems classified? State the major changes in the classified wireless systems.

1.3 Presently, what are the systems in which partly wired links and partly wireless communication are incorporated? Can you find the types of cables used in different wired systems?

1.4 Prepare a list of all existing communication systems used in everyday life. Out of these, find which are wired and which are wireless and then prepare a list of the existing wireless systems and the associated standards along with their modulation schemes, bit rate, frequency range of communication, special features, and so on.

1.5 Write short notes on the following terms:

(a) Information (b) Transmitter
(c) Types of channels (d) Types of noise
(e) Receiver (f) Modulation
(g) Carrier (h) Bandwidth
(i) SNR

1.6 The bandwidth of a channel is 250 KHz. What kind of information signals can be transmitted over it? Why should the system bandwidth be higher than the signal bandwidth?

1.7 What are the various commercial ranges for various wireless applications? Some commercial ranges are used for multi-applications. Which factors are considered to derive reliable communication in these situations?

1.8 With reference to Fig. 1.3, find the theoretical range of bit interval for which the bit occurrence can be detected and establish the relation with the system bandwidth.

1.9 What is the relationship between the fundamental frequency and the period of a signal?

1.10 Shannon and Nyquist formulas of channel capacity place an upper limit on the bit rate of a channel. Are they related? How?

1.11 What are the key factors that affect the channel capacity? Explain how the capacity is affected.

1.12 Explain the SNR–bandwidth trade-off.

1.13 Prove that the relationship between the SNR and the bandwidth expansion factor is non-linear.

1.14 Are the signal spectrum and the signal bandwidth the same? Why?

1.15 List out the various types of signals for communication described in the chapter and draw their waveforms. In which category will the audio, image, and video signals fall?

1.16 Find the Fourier transforms of the well-known functions square, triangular, exponential, and ramp.

1.17 Identify the wireless devices that incorporate various modern processors.

1.18 Represent an EM wave equation with its amplitude, frequency, and phase, assuming that the wave is travelling in any one direction.

1.19 When will a signal be a scalar or a vector? How can scalars and vectors be represented in mathematical form?

1.20 Compare AM, FM, and PM techniques of modulation. What are the drawbacks of these techniques that are eliminated using digital modulation techniques?

1.21 Why is line coding more important for wired line communication?

1.22 Why is the receiver a critical part of a complete wireless link?

1.23 Differentiate between the following terms:

(a) Analog and digital EM signals
(b) Analog and digital communication systems
(c) Guided and unguided media

1.24 List out the requirements of 4G, and from the analysis of the existing standards, find the points at which we are lacking.

or

Which are the areas that should be concentrated upon by the scientists and engineers to have a reliable *anywhere, anytime* communication scenario?

1.25 Develop the requirements of a wireless digital communication transmitter and a receiver in the form of blocks and link them to form a basic link diagram.

1.26 List the basic requirements of UMTS and LTE systems.

1.27 Compare wired and wireless communication and find why a higher bit rate is a problem in the wireless link but not in the wired link. When does a wired link have the problem of a higher bit rate?

1.28 How can we increase the user accommodation capacity on wired and wireless links?

1.29 How do licensed and unlicensed band communications differ?

1.30 Discuss the major changes that took place in the communication systems from the first to the fourth generations in general. Also, discuss separately the changes in the 1G to 4G wireless systems.

1.31 How can you say that wireless digital communication exhibits interdisciplinary approach?

Numerical Problems

1.1 If the bit rate is to be maintained at 10 Mbps, what modifications should be made in a system to cope with SNR variations between 10 dB and 20 dB?

1.2 If square pulses, each of duration 0.05 μs, are to be transmitted at a carrier frequency 100 MHz, what will be the shape of the spectrum? According to this spectrum, find the following:
(a) Null to null (significant energy) bandwidth
(b) Fractional power containment bandwidth
(c) Bounded power spectral density
(d) Absolute bandwidth

Hint: *Fractional power containment bandwidth*: According to Federal Communications Commission (FCC) rules, the occupied bandwidth is the band that levels exactly 0.5 per cent of the signal power above the upper band limit and exactly 0.5 per cent of the signal power below the lower band limit. Thus, 99 per cent of the signal power is inside the occupied band.

Bounded power spectral density: Typical attenuation level might be 35 dB or 50 dB.

Absolute bandwidth: It is the interval between the frequencies beyond which the spectrum is zero. However, for all realizable waveforms, absolute bandwidth is infinite.

1.3 The energies of signals $g_1(t)$ and $g_2(t)$ are E_{g1} and E_{g2}, respectively.
(a) Show that, in general, the energy of signal $g_1(t) + g_2(t)$ is not $E_{g1} + E_{g2}$.
(b) Under what condition is the energy of $g_1(t) + g_2(t)$ equal to $E_{g1} + E_{g2}$?
(c) Can the energy of signal $g_1(t)+ g_2(t)$ be zero? If so, under what condition(s) will it happen?

1.4 Determine the energy spectral density of the square pulse $s(t) = rect(t/T)$, where $rect(t/T)$ equals 1 for $-T/2 \le t \le T/2$ and equals 0 elsewhere. Calculate the normalized energy E_s in the pulse.

1.5 The input x and output y of a certain non-linear channel are related as $y = x + 0.22x^3$. The input signal $x(t)$ is a sum of two modulated signals as follows:

$$x(t) = x_1(t)\cos\omega_1 t + x_2(t)\cos\omega_2 t$$

where $X_1(\omega)$ and $X_2(\omega)$ are shown in Fig. 1.14.

Here $\omega_1 = 2\pi(100\times10^3)$ and $\omega_2 = 2\pi(110\times10^3)$
(a) Sketch the spectra of the input signal $x(t)$ and the output signal $y(t)$.
(b) Can the signals $x_1(t)$ and $x_2(t)$ be recovered without distortion and interference from the output $y(t)$?

1.6 Show that an arbitrary function $s(n)$ can be represented by the sum of an even function $s_e(n)$ and an odd function $s_o(n)$.

$$s(n) = s_e(n) + s_o(n)$$

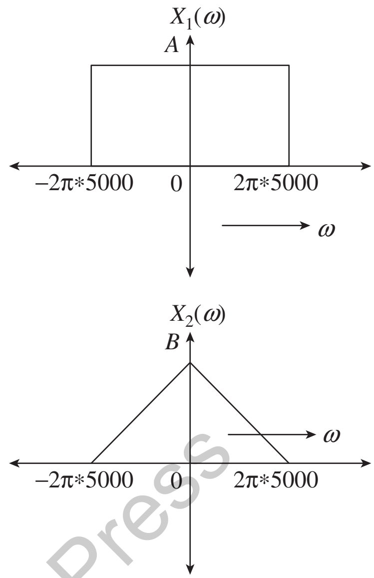


Fig. 1.14 Spectra for Problem 5

1.7 In a multilevel signalling, if the number of discrete signal or voltage levels is 8 in a modem and the bandwidth is 4 kHz, find the channel capacity. If the data rate is increased by increasing the number of signalling elements, for a given bandwidth, what will be the expected changes? Comment on it.

1.8 The bandwidth of a channel is 2 MHz and the SNR is 25 dB. Using Shannon's formula, find the channel capacity. If we assume that we can achieve this limit based on Nyquist's formula, find the number of signalling levels required.

1.9 A system with digital signalling is operated at 4800 bits per second. If the signal element encodes a 4-bit word, what is the minimum required bandwidth?

1.10 For the signal shown in Fig. 1.15, find the type of signal and suitable measure to analyze it.

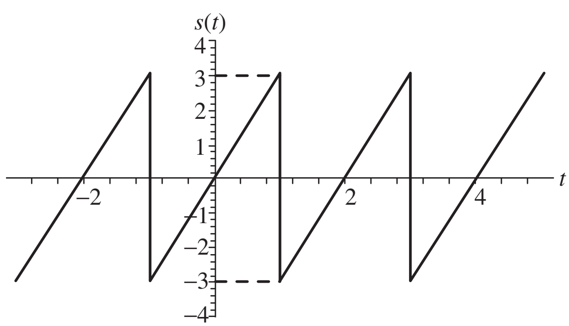


Fig. 1.15 Signal for Problem 10