Digital Communications

Fundamentals and Applications

Second Edition

Bernard Sklar
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DIGITAL COMMUNICATIONS

Fundamentals and Applications

Second Edition

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Delhi • Chennai • Chandigarh

To the memory of my mother and father,
Ruth and Julius Sklar, my wife Gwen, and our children,
Debra, Sharon, and Dean

—Bernard Sklar

To the memory of my mother Lila Ray, and
mother-in-law, Renuprova Saha

—Pabitra Kumar Ray

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Contents

Preface

This second edition of *Digital Communications: Fundamentals and Applications* represents an update of the original publication. The key features that have been updated are:

- The error-correction coding chapters have been expanded, particularly in the areas of Reed–Solomon codes, turbo codes, and trellis-coded modulation.
- A new chapter on fading channels and how to mitigate the degrading effects of fading has been introduced.
- Explanations and descriptions of essential digital communication concepts have been amplified.
- End-of-chapter problem sets have been expanded. Also, end-of-chapter question sets (and where to find the answers), as well as end-of-chapter CD exercises have been added.
- A compact disc (CD) containing an educational version of the design software SystemView by ELANIX® accompanies the textbook. The CD contains a workbook with over 200 exercises, as well as a concise tutorial on digital signal processing (DSP). CD exercises in the workbook reinforce material in the textbook; concepts can be explored by viewing waveforms with a windows-based PC and by changing parameters to see the effects on the overall system. Some of the exercises provide basic training in using SystemView; others provide additional training in DSP techniques.

The teaching of a one-semester university course proceeds in a very different manner compared with that of a short-course in the same subject. At the university, one has the luxury of time—time to develop the needed skills and mathematical tools, time to practice the ideas with homework exercises. In a short-course, the treatment is almost backwards compared with the university. Because of the time factor, a short-course teacher must "jump in" early with essential concepts and applications. One of the vehicles that I found useful in structuring a short course was to start by handing out a check list. This was not merely an outline of the curriculum. It represented a collection of concepts and nomenclature that are not clearly documented, and are often misunderstood. The short-course students were thus initiated into the course by being challenged. I promised them that once they felt comfortable describing each issue, or answering each question on the list, they would be well on their way toward becoming knowledgeable in the field of digital communications. I have learned that this list of essential concepts is just as valuable for teaching full-semester courses as it is for short courses. Here then is my "check list" for digital communications.

1. What mathematical dilemma is the cause for there being several definitions of bandwidth? (See Section 1.7.2.)
2. Why is the ratio of bit energy-to-noise power spectral density, E_b/N_0, a natural figure-to-merit for digital communication systems? (See Section 3.1.5.)
3. When representing timed events, what dilemma can easily result in confusing the most-significant bit (MSB) and the least-significant bit (LSB)? (See Section 3.2.3.1.)
4. The error performance of digital signaling suffers primarily from two degradation types. a) loss in signal-to-noise ratio, b) distortion resulting in an irreducible bit-error probability. How do they differ? (See Section 3.3.2.)
5. Often times, providing more E_b/N_0 will not mitigate the degradation due to intersymbol interference (ISI). Explain why. (See Section 3.3.2.)
6. At what location in the system is E_b/N_0 defined? (See Section 4.3.2.)
7. Digital modulation schemes fall into one of two classes with opposite behavior characteristics. a) orthogonal signaling, b) phase/amplitude signaling. Describe the behavior of each class. (See Sections 4.8.2 and 9.7.)
8. Why do binary phase shift keying (BPSK) and quaternary phase shift keying (QPSK) manifest the same bit-error-probability relationship? Does the same hold true for M-ary pulse amplitude modulation (M-PAM) and M^2-ary quadrature amplitude modulation (M^2-QAM) bit-error probability? (See Sections 4.8.4 and 9.8.3.1.)
9. In orthogonal signaling, why does error-performance improve with higher dimensional signaling? (See Section 4.8.5.)
10. Why is *free-space loss* a function of wavelength? (See Section 5.3.3.)
11. What is the relationship between received signal to noise (S/N) ratio and carrier to noise (C/N) ratio? (See Section 5.4.)
12. Describe four types of trade-offs that can be accomplished by using an error-correcting code. (See Section 6.3.4.)

13. Why do traditional error-correcting codes yield error-performance degradation at low values of E_b/N_0? (See Section 6.3.4.)
14. Of what use is the *standard array* in understanding a block code, and in evaluating its capability? (See Section 6.6.5.)
15. Why is the Shannon limit of –1.6 dB not a useful goal in the design of real systems? (See Section 8.4.5.2.)
16. What are the consequences of the fact that the Viterbi decoding algorithm does not yield *a posteriori* probabilities? What is a more descriptive name for the Viterbi algorithm? (See Section 8.4.6.)
17. Why do binary and 4-ary orthogonal frequency shift keying (FSK) manifest the same bandwidth-efficiency relationship? (See Section 9.5.1.)
18. Describe the subtle energy and rate transformations of received signals: from data-bits to channel-bits to symbols to chips. (See Section 9.7.7.)
19. Define the following terms: Baud, State, Communications Resource, Chip, Robust Signal. (See Sections 1.1.3 and 7.2.2, Chapter 11, and Sections 12.3.2 and 12.4.2.)
20. In a fading channel, why is signal dispersion independent of fading rapidity? (See Section 15.1.1.1.)

I hope you find it useful to be challenged in this way. Now, let us describe the purpose of the book in a more methodical way. This second edition is intended to provide a comprehensive coverage of digital communication systems for senior level undergraduates, first year graduate students, and practicing engineers. Though the emphasis is on digital communications, necessary analog fundamentals are included since analog waveforms are used for the radio transmission of digital signals. The key feature of a digital communication system is that it deals with a finite set of discrete messages, in contrast to an analog communication system in which messages are defined on a continuum. The objective at the receiver of the digital system is not to reproduce a waveform with precision; it is instead to determine from a noise-perturbed signal, which of the finite set of waveforms had been sent by the transmitter. In fulfillment of this objective, there has arisen an impressive assortment of signal processing techniques.

The book develops these techniques in the context of a unified structure. The structure, in block diagram form, appears at the beginning of each chapter; blocks in the diagram are emphasized, when appropriate, to correspond to the subject of that chapter. Major purposes of the book are to add organization and structure to a field that has grown and continues to grow rapidly, and to insure awareness of the "big picture" even while delving into the details. Signals and key processing steps are traced from the information source through the transmitter, channel, receiver, and ultimately to the information sink. Signal transformations are organized according to nine functional classes: Formatting and source coding, Baseband signaling, Bandpass signaling, Equalization, Channel coding, Muliplexing and multiple access, Spreading, Encryption, and Synchronization. Throughout the book, emphasis is placed on system goals and the need to trade off basic system parameters such as signal-to-noise ratio, probability of error, and bandwidth expenditure.

ORGANIZATION OF THE BOOK

Chapter 1 introduces the overall digital communication system and the basic signal transformations that are highlighted in subsequent chapters. Some basic ideas of random variables and the *additive white Gaussian noise* (AWGN) model are reviewed. Also, the relationship between power spectral density and autocorrelation, and the basics of signal transmission through linear systems are established. Chapter 2 covers the signal processing step, known as *formatting,* in order to render an information signal compatible with a digital system. Chapter 3 emphasizes *baseband signaling,* the detection of signals in Gaussian noise, and receiver optimization. Chapter 4 deals with *bandpass signaling* and its associated modulation and demodulation/detection techniques. Chapter 5 deals with *link analysis,* an important subject for providing overall system insight; it considers some subtleties that are often missed. Chapters 6, 7, and 8 deal with *channel coding*—a cost-effective way of providing a variety of system performance trade-offs. Chapter 6 emphasizes *linear block codes,* Chapter 7 deals with *convolutional codes,* and Chapter 8 deals with *Reed–Solomon codes* and *concatenated codes* such as *turbo codes.*

Chapter 9 considers various modulation/coding system *trade-offs* dealing with probability of bit-error performance, bandwidth efficiency, and signal-to-noise ratio. It also treats the important area of coded modulation, particularly *trellis-coded modulation.* Chapter 10 deals with *synchronization* for digital systems. It covers phase-locked loop implementation for achieving carrier synchronization. It covers bit synchronization, frame synchronization, and network synchronization, and it introduces some ways of performing synchronization using digital methods.

Chapter 11 treats *multiplexing* and *multiple access*. It explores techniques that are available for utilizing the communication resource efficiently. Chapter 12 introduces *spread spectrum* techniques and their application in such areas as multiple access, ranging, and interference rejection. This technology is important for both military and commercial applications. Chapter 13 deals with *source coding* which is a special class of data formatting. Both formatting and source coding involve digitization of data; the main difference between them is that source coding additionally involves data redundancy reduction. Rather than considering source coding immediately after formatting, it is purposely treated in a later chapter so as not to interrupt the presentation flow of the basic processing steps. Chapter 14 covers basic *encryption/decryption* ideas. It includes some classical concepts, as well as a class of systems called public key cryptosystems, and the widely used E-mail encryption software known as *Pretty Good Privacy* (PGP). Chapter 15 deals with *fading channels.* Here, we deal with applications, such as mobile radios, where characterization of the channel is much more involved than that of a nonfading one. The design of a communication system that will withstand the degradation effects of fading can be much more challenging than the design of its nonfading counterpart. In this chapter, we describe a variety of techniques that can mitigate the effects of fading, and we show some successful designs that have been implemented.

It is assumed that the reader is familiar with Fourier methods and convolution. Appendix A reviews these techniques, emphasizing those properties that are particularly useful in the study of communication theory. It also assumed that the reader has a knowledge of basic probability and has some familiarity with random variables. Appendix B builds on these disciplines for a short treatment on statistical decision theory with emphasis on hypothesis testing—so important in the understanding of detection theory. A new section, Appendix E, has been added to serve as a short tutorial on s-domain, z-domain, and digital filtering.

If the book is used for a two-term course, a simple partitioning is suggested; the first seven chapters can be taught in the first term, and the last eight chapters in the second term. If the book is used for a one-term introductory course, it is suggested that the course material be selected from the following chapters: 1, 2, 3, 4, 5, 6, 7, 9, 10, 12.

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It is difficult to write a technical book without contributions from others. I have received an abundance of such assistance, for which I am deeply grateful. For their generous help, I want to thank Dr. Andrew Viterbi, Dr. Chuck Wheatley, Dr. Ed Tiedeman, Dr. Joe Odenwalder, and Serge Willinegger of Qualcomm. I also want to thank Dr. Dariush Divsalar of Jet Propulsion Laboratory (JPL), Dr. Bob Bogusch of Mission Research, Dr. Tom Stanley of the Federal Communications Commission, Professor Larry Milstein of the University of California, San Diego, Professor Ray Pickholtz of George Washington University, Professor Daniel Costello of Notre Dame University, Professor Ted Rappaport of Virginia Polytechnic Institute, Phil Kossin of Lincom, Les Brown of Motorola, as well as Dr. Bob Price and Frank Amoroso.

I also want to acknowledge those people who played a big part in helping me with the first edition of the book. They are: Dr. Maurice King, Don Martin and Ned Feldman of The Aerospace Corporation, Dr. Marv Simon of JPL, Dr. Bill Lindsey of Lincom, Professor Wayne Stark of the University of Michigan, as well as Dr. Jim Omura, Dr. Adam Lender, and Dr. Todd Citron.

I want to thank Dr. Maurice King for contributing Chapter 10 on Synchronization, and Professor Fred Harris of San Diego State University for contributing Chapter 13 on Source Coding. Also, thanks to Michelle Landry for writing the sections on Pretty Good Privacy in Chapter 14, and to Andrew Guidi for contributing end-of-chapter problems in Chapter 15.

I am particularly indebted to my friends and colleagues Fred Harris, Professor Dan Bukofzer of California State University at Fresno, and Dr. Maury Schiff of Elanix, who put up with my incessant argumentative discussions anytime that I called on them. I also want to thank my very best teachers—they are my students at the University of California, Los Angeles, as well as those students all over the world who attended my short courses. Their questions motivated me and provoked me to write this second edition. I hope that I have answered all their questions with clarity.

I offer special thanks for technical clarifications that my son, Dean Sklar, suggested; he took on the difficult role of being his father's chief critic and "devil's advocate." I am particularly indebted to Professor Bob Stewart of the University of

Strathclyde, Glasgow, who contributed countless hours of work in writing and preparing the CD and in authoring Appendix E. I thank Rose Kernan, my editor, for watching over me and this project, and I thank Bernard Goodwin, Publisher at Prentice Hall, for indulging me and believing in me. His recommendations were invaluable. Finally, I am extremely grateful to my wife, Gwen, for her encouragement, devotion, and valuable advice. She protected me from the "slings and arrows" of everyday life, making it possible for me to complete this second edition.

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Interest in optimum receiver design for attaining minimum error probability of binary message and estimation of signal parameter in fading channel has necessitated their study in Digital Communication. Keeping this in mind, I have added a new chapter on 'Optimum Detection and Estimation' which highlights the basic theories and applications in the field of detection and estimation has therefore been added to the adapted version. Many new problems and end-of-chapter exercises have also been included. Finally, every effort has been made to retain the lucid and well-illustrated style of presentation in the original book.

This work involved many hours of effort that was frequently interrupted due to my engagements in various academic commitments. It would have taken a longer time to finish the work had I not been driven by the ardent wish of and constant insistence by my wife Sharmistha for completion of the work. I thank her.

I am further thankful to Thomas Mathew Rajesh of Pearson Education who has not only given support to the work, but also forgiven my failures to submit the manuscript on time in spite of repeated reminders.

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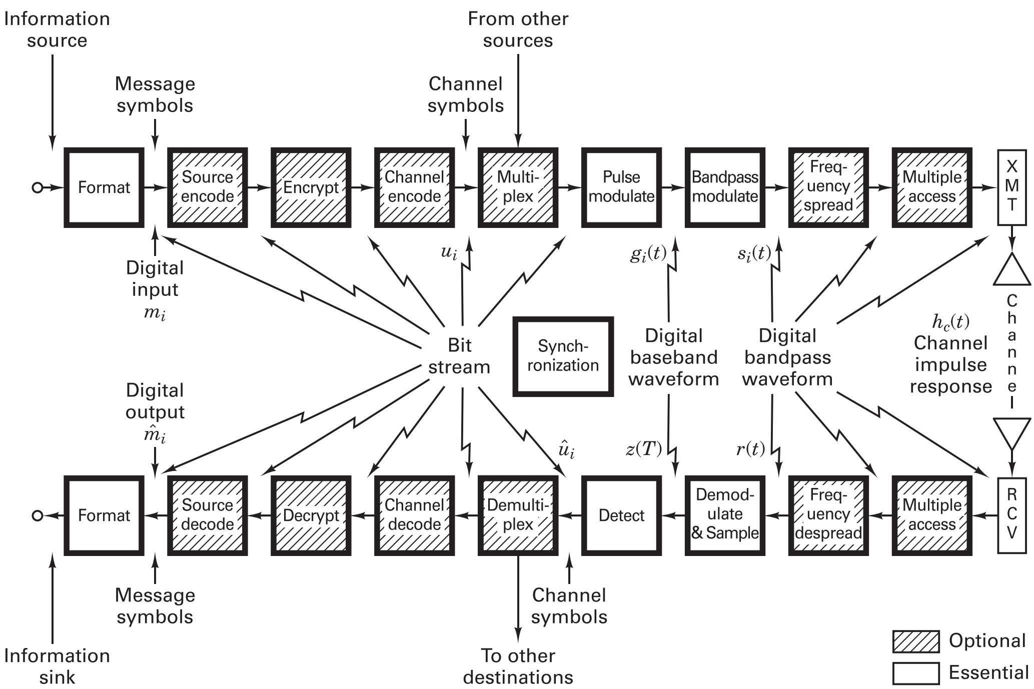
CHAPTER 1

Signals and Spectra



This book presents the ideas and techniques fundamental to digital communication systems. Emphasis is placed on system design goals and on the need for trade-offs among basic system parameters such as signal-to-noise ratio (SNR), probability of error, and bandwidth expenditure. We shall deal with the transmission of information (voice, video, or data) over a path (channel) that may consist of wires, waveguides, or space.

Digital communication systems are becoming increasingly attractive because of the ever-growing demand for data communication and because digital transmission offers data processing options and flexibilities not available with analog transmission. In this book, a digital system is often treated in the context of a satellite communications link. Sometimes the treatment is in the context of a mobile radio system, in which case signal transmission typically suffers from a phenomenon called *fading.* In general, the task of characterizing and mitigating the degradation effects of a fading channel is more challenging than performing similar tasks for a nonfading channel.

The principal feature of a digital communication system (DCS) is that during a finite interval of time, it sends a waveform from a finite set of possible waveforms, in contrast to an analog communication system, which sends a waveform from an infinite variety of waveform shapes with theoretically infinite resolution. In a DCS, the objective at the receiver is *not* to reproduce a transmitted waveform with precision; instead, the objective is to determine from a noise-perturbed signal which waveform from the finite set of waveforms was sent by the transmitter. An important measure of system performance in a DCS is the probability of error (P_E).

1.1 DIGITAL COMMUNICATION SIGNAL PROCESSING

1.1.1 Why Digital?

Why are communication systems, military and commercial alike, "going digital"? There are many reasons. The primary advantage is the ease with which digital signals, compared with analog signals, are regenerated. Figure 1.1 illustrates an ideal binary digital pulse propagating along a transmission line. The shape of the waveform is affected by two basic mechanisms: (1) as all transmission lines and circuits have some nonideal frequency transfer function, there is a distorting effect on the ideal pulse; and (2) unwanted electrical noise or other interference further distorts the pulse waveform. Both of these mechanisms cause the pulse shape to degrade as a function of line length, as shown in Figure 1.1. During the time that the transmitted pulse can still be reliably identified (before it is degraded to an ambiguous state), the pulse is amplified by a digital amplifier that recovers its original ideal shape. The pulse is thus "reborn" or regenerated. Circuits that perform this function at regular intervals along a transmission system are called *regenerative repeaters.*

Digital circuits are less subject to distortion and interference than are analog circuits. Because binary digital circuits operate in one of two states—fully on or fully off—to be meaningful, a disturbance must be large enough to change the circuit operating point from one state to the other. Such two-state operation facilitates signal regeneration and thus prevents noise and other disturbances from accumulating in transmission. Analog signals, however, are *not* two-state signals; they can take an *infinite variety* of shapes. With analog circuits, even a small disturbance can render the reproduced waveform unacceptably distorted. Once the analog signal is distorted, the distortion cannot be removed by amplification. Because accumulated noise is irrevocably bound to analog signals, they cannot be perfectly regenerated. With digital techniques, extremely low error rates producing

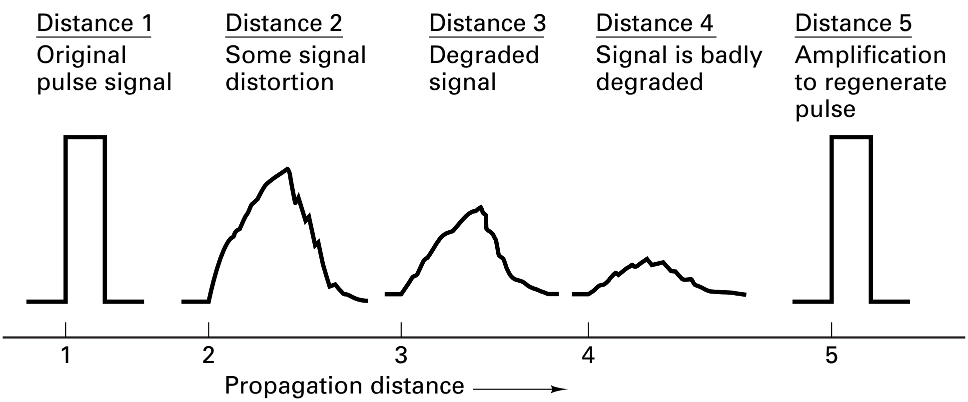


Figure 1.1 Pulse degradation and regeneration.

high signal fidelity are possible through error detection and correction but similar procedures are not available with analog.

There are other important advantages to digital communications. Digital circuits are *more reliable* and can be produced at a lower cost than analog circuits. Also, digital hardware lends itself to *more flexible* implementation than analog hardware [e.g., microprocessors, digital switching, and large-scale integrated (LSI) circuits]. The combining of digital signals using time-division multiplexing (TDM) is *simpler* than the combining of analog signals using frequency-division multiplexing (FDM). Different types of digital signals (data, telegraph, telephone, television) can be treated as identical signals in transmission and switching—*a bit is a bit.* Also, for convenient switching, digital messages can be handled in autonomous groups called *packets.* Digital techniques lend themselves naturally to signal processing functions that protect against interference and jamming, or that provide encryption and privacy. (Such techniques are discussed in Chapters 12 and 14, respectively.) Also, much data communication is from computer to computer, or from digital instruments or terminal to computer. Such digital terminations are naturally best served by digital communication links.

What are the costs associated with the beneficial attributes of digital communication systems? Digital systems tend to be very signal-processing intensive compared with analog. Also, digital systems need to allocate a significant share of their resources to the task of synchronization at various levels. (See Chapter 10.) With analog systems, on the other hand, synchronization often is accomplished more easily. One disadvantage of a digital communication system is *nongraceful degradation.* When the signal-to-noise ratio drops below a certain threshold, the quality of service can change suddenly from very good to very poor. In contrast, most analog communication systems degrade more gracefully.

1.1.2 Typical Block Diagram and Transformations

The functional block diagram shown in Figure 1.2 illustrates the signal flow and the signal-processing steps through a typical digital communication system (DCS). This figure can serve as a kind of road map, guiding the reader through the chapters of this book. The upper blocks—format, source encode, encrypt, channel encode, multiplex, pulse modulate, bandpass modulate, frequency spread, and multiple access—denote signal transformations from the source to the transmitter (XMT). The lower blocks denote signal transformations from the receiver (RCV) to the sink, essentially reversing the signal processing steps performed by the upper blocks. The *modulate* and *demodulate/detect* blocks together are called a *modem.* The term "modem" often encompasses several of the signal processing steps shown in Figure 1.2; when this is the case, the modem can be thought of as the "brains" of the system. The transmitter and receiver can be thought of as the "muscles" of the system. For wireless applications, the transmitter consists of a frequency up-conversion stage to a radio frequency (RF), a high-power amplifier, and an antenna. The receiver portion consists of an antenna and a low-noise amplifier (LNA). Frequency down-conversion is performed in the front end of the receiver and/or the demodulator.

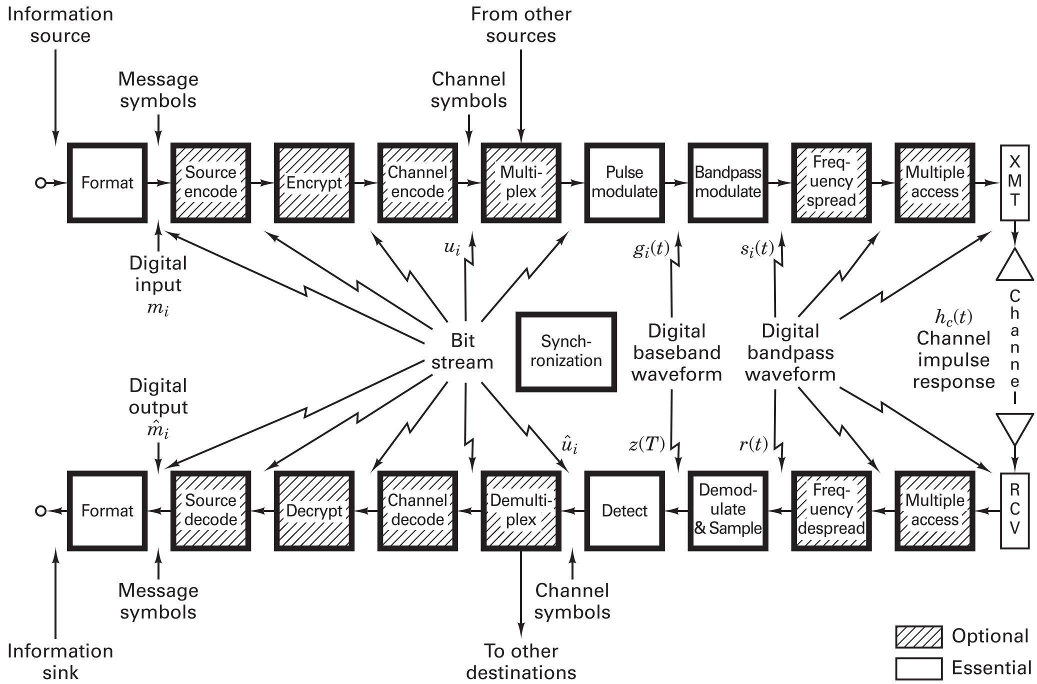


Figure 1.2 Block diagram of a typical digital communication system.

Figure 1.2 illustrates a kind of reciprocity between the blocks in the upper transmitter part of the figure and those in the lower receiver part. The signal processing steps that take place in the transmitter are, for the most part, reversed in the receiver. In Figure 1.2, the input information source is converted to binary digits (*bits*); the bits are then grouped to form *digital messages* or *message symbols.* Each such symbol (m_i, where $i = 1, \ldots, M$) can be regarded as a member of a *finite alphabet* set containing M members. Thus, for $M = 2$, the message symbol m_i is binary (meaning that it constitutes just a single bit). Even though binary symbols fall within the general definition of M-ary, nevertheless the name M-ary is usually applied to those cases where $M > 2$; hence, such symbols are each made up of a sequence of two or more bits. (Compare such a finite alphabet in a DCS with an analog system, where the message waveform is typically a member of an infinite set of possible waveforms.) For systems that use *channel coding* (error correction coding), a sequence of message symbols becomes transformed to a sequence of *channel symbols* (code symbols), where each channel symbol is denoted u_i. Because a message symbol or a channel symbol can consist of a single bit or a grouping of bits, a sequence of such symbols is also described as a *bit stream,* as shown in Figure 1.2.

Consider the key signal processing blocks shown in Figure 1.2; only formatting, modulation, demodulation/detection, and synchronization are essential for a DCS. *Formatting* transforms the source information into bits, thus assuring compatibility between the information and the signal processing within the DCS. From this point in the figure up to the pulse-modulation block, the information remains in the form of a *bit stream.* Modulation is the process by which message symbols or channel symbols (when channel coding is used) are converted to *waveforms* that are compatible with the requirements imposed by the transmission channel. *Pulse modulation* is an essential step because each symbol to be transmitted must first be transformed from a binary representation (voltage levels representing binary ones and zeros) to a *baseband* waveform. The term baseband refers to a signal whose spectrum extends from (or near) dc up to some finite value, usually less than a few megahertz. The pulse-modulation block usually includes filtering for minimizing the transmission bandwidth. When pulse modulation is applied to binary symbols, the resulting binary waveform is called a pulse-code-modulation (PCM) waveform. There are several types of PCM waveforms (described in Chapter 2); in telephone applications, these waveforms are often called *line codes.* When pulse modulation is applied to nonbinary symbols, the resulting waveform is called an M-ary pulse-modulation waveform. There are several types of such waveforms, and they too are described in Chapter 2, where the one called *pulse-amplitude modulation* (PAM) is emphasized. After pulse modulation, each message symbol or channel symbol takes the form of a baseband waveform $g_i(t)$, where $i = 1, \ldots, M$. In any electronic implementation, the bit stream, prior to pulse-modulation, is represented with voltage levels. One might wonder why there is a separate block for pulse modulation when in fact different voltage levels for binary ones and zeros can be viewed as impulses or as ideal rectangular pulses, each pulse occupying one bit time. There are two important differences between such voltage levels and the baseband waveforms used for modulation. First, the pulse-modulation block allows for a variety of binary and M-ary pulse-waveform types. Section 2.8.2 describes the different useful attributes of these types of waveforms. Second, the filtering within the pulse-modulation block yields pulses that occupy more than just one-bit time. Filtering yields pulses that are spread in time, thus the pulses are "smeared" into neighboring bit-times. This filtering is sometimes referred to as pulse shaping; it is used to contain the transmission bandwidth within some desired spectral region.

For an application involving RF transmission, the next important step is *bandpass modulation;* it is required whenever the transmission medium will not support the propagation of pulse-like waveforms. For such cases, the medium requires a bandpass waveform $s_i(t)$, where $i = 1, \ldots, M$. The term *bandpass* is used to indicate that the baseband waveform $g_i(t)$ is frequency translated by a carrier wave to a frequency that is much larger than the spectral content of $g_i(t)$. As $s_i(t)$ propagates over the channel, it is impacted by the channel characteristics, which can be described in terms of the channel's *impulse response* $h_c(t)$ (see Section 1.6.1). Also, at various points along the signal route, additive random noise distorts the received signal $r(t)$, so that its reception must be termed a corrupted version of the signal $s_i(t)$ that was launched at the transmitter. The received signal $r(t)$ can be expressed as

$$r(t) = s_i(t) * h_c(t) + n(t) \qquad i = 1, \ldots, M \tag{1.1}$$

where $*$ represents a convolution operation (see Appendix A), and $n(t)$ represents a noise process (see Section 1.5.5).

In the reverse direction, the receiver front end and/or the demodulator provides frequency down-conversion for each bandpass waveform $r(t)$. The demodulator restores $r(t)$ to an optimally shaped baseband pulse $z(t)$ in preparation for detection. Typically, there can be several filters associated with the receiver and demodulator—filtering to remove unwanted high frequency terms (in the frequency down-conversion of bandpass waveforms), and filtering for pulse shaping. Equalization can be described as a filtering option that is used in or after the demodulator to reverse any degrading effects on the signal that were caused by the channel. Equalization becomes essential whenever the impulse response of the channel, $h_c(t)$, is so poor that the received signal is badly distorted. An equalizer is implemented to compensate for (i.e., remove or diminish) any signal distortion caused by a nonideal $h_c(t)$. Finally, the sampling step transforms the shaped pulse $z(t)$ to a sample $z(T)$, and the detection step transforms $z(T)$ to an estimate of the channel symbol $\hat{u}_i$ or an estimate of the message symbol $\hat{m}_i$ (if there is no channel coding). Some authors use the terms "demodulation" and "detection" interchangeably. However, in this book, *demodulation* is defined as recovery of a waveform (baseband pulse), and *detection* is defined as decision-making regarding the digital meaning of that waveform.

The other signal processing steps within the modem are design options for specific system needs. *Source coding* produces analog-to-digital (A/D) conversion (for analog sources) *and* removes redundant (unneeded) information. Note that a typical DCS would either use the *source coding* option (for both digitizing and compressing the source information), or it would use the simpler *formatting* transformation (for digitizing alone). A system would not use both source coding and formatting, because the former already includes the essential step of digitizing the information. Encryption, which is used to provide communication privacy, prevents unauthorized users from understanding messages and from injecting false messages into the system. *Channel coding,* for a given data rate, can reduce the probability of error, P_E, or reduce the required signal-to-noise ratio to achieve a desired P_E at the expense of transmission bandwidth or decoder complexity. *Multiplexing* and *multiple-access procedures* combine signals that might have different characteristics or might originate from different sources, so that they can share a portion of the communications resource (e.g., spectrum, time). Frequency spreading can produce a signal that is relatively invulnerable to interference (both natural and intentional) and can be used to enhance the privacy of the communicators. It is also a valuable technique used for multiple access.

The signal processing blocks shown in Figure 1.2 represent a typical arrangement; however, these blocks are sometimes implemented in a different order. For example, multiplexing can take place prior to channel encoding, *or* prior to modulation, *or*—with a two-step modulation process (subcarrier and carrier)—it can be performed between the two modulation steps. Similarly, frequency spreading can take place at various locations along the upper portion of Figure 1.2; its precise location depends on the particular technique used. Synchronization and its key element, a clock signal, is involved in the control of all signal processing within the

DCS. For simplicity, the synchronization block in Figure 1.2 is drawn without any connecting lines, when in fact it actually plays a role in regulating the operation of almost every block shown in the figure.

Figure 1.3 shows the basic signal processing functions, which may be viewed as transformations, classified into the following nine groups:

1. Formatting and source coding
2. Baseband signaling
3. Bandpass signaling
4. Equalization
5. Channel coding
6. Multiplexing and multiple access
7. Spreading
8. Encryption
9. Synchronization

Although this organization has some inherent overlap, it provides a useful structure for the book. Beginning with Chapter 2, the nine basic transformations are considered individually. In Chapter 2, the basic formatting techniques for transforming the source information into message symbols are discussed, as well as the selection of baseband pulse waveforms and pulse filtering for making the message symbols compatible with baseband transmission. The reverse steps of demodulation,equalization, sampling, and detection are described in Chapter 3. Formatting and source coding are similar processes, in that they both involve data digitization. However, the term "source coding" has taken on the connotation of data compression in addition to digitization; it is treated later (in Chapter 13), as a special case of formatting.

In Figure 1.3, the *Baseband Signaling* block contains a list of binary choices under the heading of PCM waveforms or line codes. In this block, a nonbinary category of waveforms called *M*-ary pulse modulation is also listed. Another transformation in Figure 1.3, labeled *Bandpass Signaling* is partitioned into two basic blocks, coherent and noncoherent. Demodulation is typically accomplished with the aid of *reference* waveforms. When the references used are a measure of all the signal attributes (particularly phase), the process is termed *coherent;* when phase information is not used, the process is termed *noncoherent.* Both techniques are detailed in Chapter 4.

Chapter 5 is devoted to *link analysis.* Of the many specifications, analyses, and tabulations that support a developing communication system, link analysis stands out in its ability to provide overall system insight. In Chapter 5 we bring together all the link fundamentals that are essential for the analysis of most communication systems.

Channel coding deals with the techniques used to enhance digital signals so that they are less vulnerable to such channel impairments as noise, fading, and jamming. In Figure 1.3 channel coding is partitioned into two blocks, waveform coding and structured sequences. *Waveform coding* involves the use of new waveforms, yielding improved detection performance over that of the original waveforms. *Structured sequences* involve the use of redundant bits to determine whether or

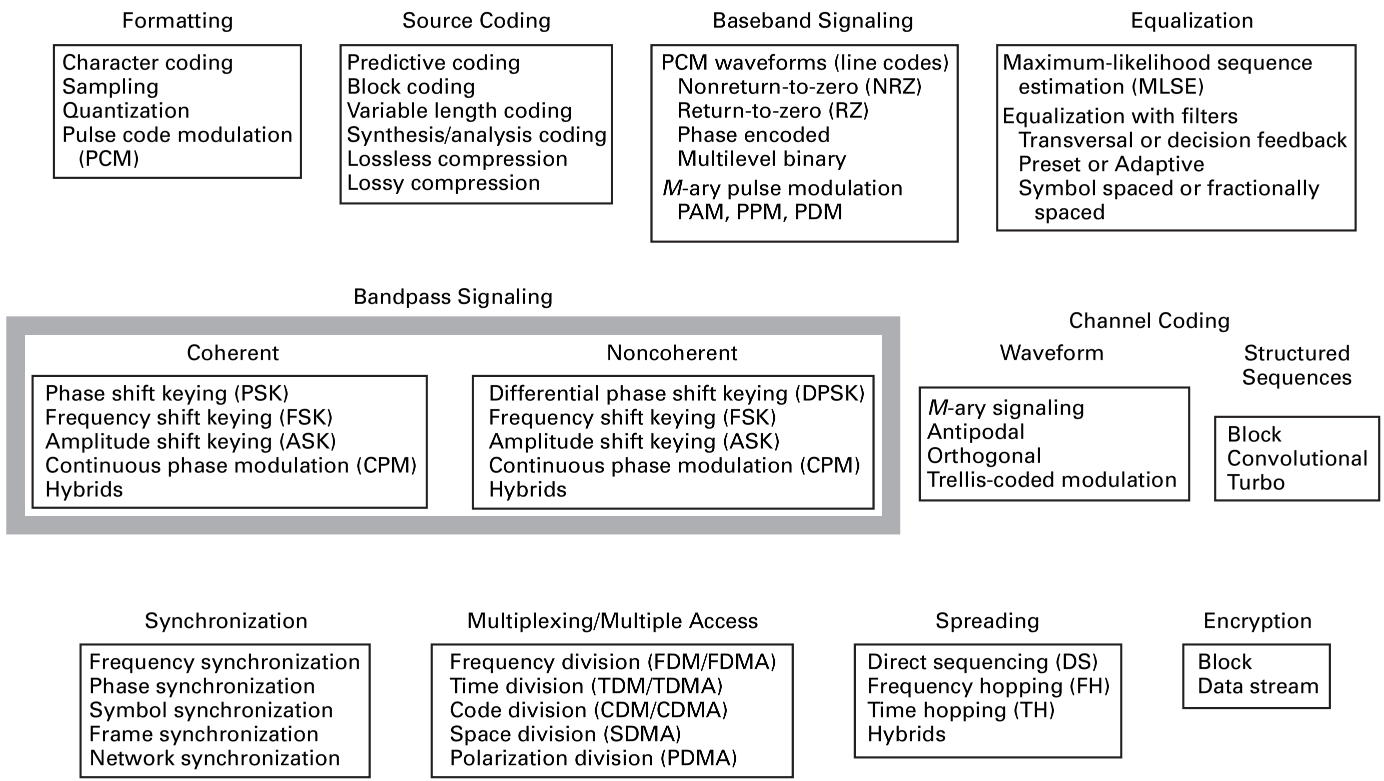


Figure 1.3 Basic digital communication transformations.