英文版

《通信与信息科学教育丛书》

Digital Communications

Fourth Edition

John G. Proakis

数字通信（第四版）

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McGraw-Hill
麦格劳－希尔教育出版集团

PHEI
電子工業出版社
PUBLISHING HOUSE OF ELECTRONICS INDUSTRY
http://www.phei.com.cn

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Publishing House of Electronics Industry
北京·BEIJING

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Digital Communications, Fourth Edition
ISBN:0-07-232111-3

图书在版编目(CIP)数据

数字通信/(美)普罗科斯(Proakis,J.G.)著.-北京:电子工业出版社,2001.10
(通信与信息科学教育丛书)
书名原文:Digital Communications(4e)
ISBN 7-5053-7048-0

Ⅰ.数… Ⅱ.普… Ⅲ.数字通信-英文 Ⅳ.TN914.3

中国版本图书馆 CIP 数据核字(2001)第 068982 号

原 书 名:Digital Communications, Fourth Edition
书　　名:**数字通信(第四版)**
著　　者:〔美〕John G. Proakis
责任编辑:段　颖
排版制作:电子工业出版社计算机排版室监制
印 刷 者:北京天竺颖华印刷厂
装 订 者:三河市金马印装有限公司
出版发行:电子工业出版社　http://www.phei.com.cn
北京市海淀区万寿路 173 信箱　邮编 100036
经　　销:各地新华书店
开　　本:850×1 168　1/32　印张:32
版　　次:2001 年 10 月第 4 版　2001 年 10 月第 1 次印刷
书　　号:ISBN 7-5053-7048-0 / TN·1474
印　　数:5 000 册　定价:59.00 元
版权贸易合同登记号　图字:01-2000-4291
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PREFACE

The fourth edition of *Digital Communications* has undergone a minor revision. Several new topics have been added, including serial and parallel concatenated codes, punctured convolutional codes, turbo TCM and turbo equalization, and spatial multiplexing. Since this is an introductory-level text, the treatment of these topics is limited in scope.

The book is designed to serve as a text for a first-year graduate-level course for students in electrical engineering. It is also designed to serve as a text for self-study and as a reference book for the practicing engineer involved in the design of digital communications systems. As a background, I presume that the reader has a thorough understanding of basic calculus and elementary linear systems theory and some prior knowledge of probability and stochastic processes.

Chapter 1 is an introduction to the subject, including a historical perspective and a description of channel characteristics and channel models.

Chapter 2 contains a review of the basic elements of probability and stochastic processes. It deals with a number of probability distribution functions and moments that are used throughout the book. It also includes the derivation of the Chernoff bound, which is useful in obtaining bounds on the performance of digital communications systems.

Chapter 3 treats source coding for discrete and analog sources. Emphasis is placed on scalar and vector quantization techniques, and comparisons are made with basic results from rate-distortion theory.

In **Chapter 4**, the reader is introduced to the representation of digitally modulated signals and to the characterization of narrowband signals and systems. Also treated in this chapter are the spectral characteristics of digitally modulated signals. New material has been added on a linear representation of CPM signals.

Chapter 5 treats the design of modulation and optimum demodulation and detection methods for digital communications over an additive white Gaussian noise channel. Emphasis is placed on the evaluation of the error rate performance for the various digital signaling techniques and on the channel bandwidth requirements of the corresponding signals.

Chapter 6 is devoted to carrier phase estimation and time synchronization methods based on the maximum-likelihood criterion. Both decision-directed and non-decision-directed methods are described.

Chapter 7 treats the topics of channel capacity for several different channel models and random coding.

Chapter 8 treats linear block and convolutional codes. The new topics added to the chapter include serial and parallel interleaved concatenated block and convolutional codes, punctured and rate-compatible convolutional codes, the soft-output Viterbi algorithm (SOVA), and turbo TCM.

Chapter 9 is focused on signal design for bandlimited channels. This chapter includes the topics of partial response signals and run-length-limited codes for spectral shaping.

Chapter 10 treats the problem of demodulation and detection of signals corrupted by intersymbol interference. The emphasis is on optimum and suboptimum equalization methods and their performance. New topics added to the chapter include Tomlinson-Harashima precoding, reduced complexity maximum-likelihood detectors, and turbo equalization.

Chapter 11 treats adaptive channel equalization. The LMS and recursive least-squares algorithms are described, together with their performance characteristics. This chapter also includes a treatment of blind equalization algorithms. New topics added include the tap-leakage algorithm and methods for accelerating the initial convergence of the LMS algorithm.

Chapter 12 treats multichannel and multicarrier modulation. The latter subject is particularly appropriate in view of several important applications that have been developed over the past two decades.

Chapter 13 is devoted to spread spectrum signals and systems. The benefits of coding in the design of spread spectrum signals is emphasized throughout this chapter.

Chapter 14 treats communication through fading channels. Several channel fading statistical models are considered, with emphasis placed on Rayleigh fading and Nakagami fading. Trellis coding for fading channels is also included in this chapter. New material added includes a brief treatment of fading and multipath characteristics of mobile radio channels, receiver structures for fading multipath channels with intersymbol interference, and spatial multiplexing using multiple transmit and receive antennas.

Chapter 15 treats multiuser communications. The emphasis is on code-division multiple access (CDMA), signal detection and random access methods, such as ALOHA and carrier-sense multiple access (CSMA).

With 15 chapters and a variety of topics, the instructor has the flexibility to design either a one- or two-semester course. Chapters 3 through 6 provide a basic treatment of digital modulation/demodulation and detection methods. Channel coding, treated in Chapters 7 and 8, can be included along with modulation and demodulation in a one-semester course. The topics of channel equalization, fading channels, spread spectrum, and multiuser communications can be covered in a second-semester course.

Throughout my professional career, I have had the opportunity to work with and learn from a number of people whom I should like to publicly acknowledge. These include Dr. R. Price, P.R. Drouilhet, Jr., and Dr. P.E. Green, Jr., who introduced me to various aspects of digital communications through fading multipath channels and multichannel signal transmission during my employment at the MIT Lincoln Laboratory. I am also indebted to Professor D.W. Tufts, who supervised my Ph.D. dissertation at Harvard University and who introduced me to the problems of signal design and equalization for band-limited channels. Over the years, I have had the pleasure of working on a variety of research projects in collaboration with colleagues at GTE and Stein Associates, including Dr. S. Stein, Dr. B. Barrow, Dr. A.A. Giordano, Dr. A.H. Levesque, Dr. R. Greenspan, Dr. D. Freeman, P.H.Anderson, D. Gooding, and J. Lindholm. At Northeastern University, I have had the benefit of collaborating with Dr. M. Salehi, Dr. M. Stojanovic, and Dr. D. Brady. Dr. T. Schonhoff provided the graphs illustrating the spectral characteristics of CPFSK, and H. Gibbons provided the data for the graphs in Chapter 14 that show the performance of PSK and DPSK with diversity. The assistance of these colleagues is greatly appreciated.

McGraw-Hill and I would like to thank the following reviewers of this edition for their valuable suggestions: William E. Ryan, *University of Arizona*; Tan Wong, *University of Florida*; and Raymond Pickholtz, *George Washington University*.

Finally, I wish to express my appreciation to Gloria Doukakis, for typing the manuscript of this edition, and to Apostolos Rizos for preparing the Solutions Manual.

CONTENTS

1

Introduction

In this book, we present the basic principles that underlie the analysis and design of digital communication systems. The subject of digital communications involves the transmission of information in digital form from a source that generates the information to one or more destinations. Of particular importance in the analysis and design of communication systems are the characteristics of the physical channels through which the information is transmitted. The characteristics of the channel generally affect the design of the basic building blocks of the communication system. Below, we describe the elements of a communication system and their functions.

1.1 ELEMENTS OF A DIGITAL COMMUNICATION SYSTEM

Figure 1.1–1 illustrates the functional diagram and the basic elements of a digital communication system. The source output may be either an analog signal, such as an audio or video signal, or a digital signal, such as the output of a teletype machine, that is discrete in time and has a finite number of output characters. In a digital communication system, the messages produced by the source are converted into a sequence of binary digits. Ideally, we should like to represent the source output (message) by as few binary digits as possible. In other words, we seek an efficient representation of the source output that results in little or no redundancy. The process of efficiently converting the output of either an analog or digital source into a sequence of binary digits is called *source encoding* or *data compression*.

The sequence of binary digits from the source encoder, which we call the *information sequence*, is passed to the *channel encoder*. The purpose of the channel encoder is to introduce, in a controlled manner, some redundancy in the binary information sequence that can be used at the receiver to overcome the effects of noise and interference encountered in the transmission of the signal

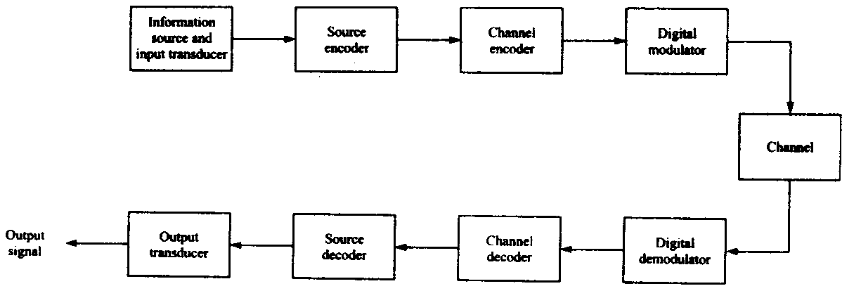

FIGURE 1.1–1
Basic elements of a digital communication system.

through the channel. Thus, the added redundancy serves to increase the reliability of the received data and improves the fidelity of the received signal. In effect, redundancy in the information sequence aids the receiver in decoding the desired information sequence. For example, a (trivial) form of encoding of the binary information sequence is simply to repeat each binary digit m times, where m is some positive integer. More sophisticated (nontrivial) encoding involves taking k information bits at a time and mapping each k-bit sequence into a unique n-bit sequence, called a *code word*. The amount of redundancy introduced by encoding the data in this manner is measured by the ratio n/k. The reciprocal of this ratio, namely k/n, is called the rate of the code or, simply, the *code rate*.

The binary sequence at the output of the channel encoder is passed to the *digital modulator*, which serves as the interface to the communication channel. Since nearly all the communication channels encountered in practice are capable of transmitting electrical signals (waveforms), the primary purpose of the digital modulator is to map the binary information sequence into signal waveforms. To elaborate on this point, let us suppose that the coded information sequence is to be transmitted one bit at a time at some uniform rate R bits per second (bits/s). The digital modulator may simply map the binary digit 0 into a waveform $s_0(t)$ and the binary digit 1 into a waveform $s_1(t)$. In this manner, each bit from the channel encoder is transmitted separately. We call this *binary modulation*. Alternatively, the modulator may transmit b coded information bits at a time by using $M = 2^b$ distinct waveforms $s_i(t)$, $i = 0, 1, \ldots, M - 1$, one waveform for each of the 2^b possible b-bit sequences. We call this *M-ary modulation* ($M > 2$). Note that a new b-bit sequence enters the modulator every b/R seconds. Hence, when the channel bit rate R is fixed, the amount of time available to transmit one of the M waveforms corresponding to a b-bit sequence is b times the time period in a system that uses binary modulation.

The *communication channel* is the physical medium that is used to send the signal from the transmitter to the receiver. In wireless transmission, the channel may be the atmosphere (free space). On the other hand, telephone channels usually employ a variety of physical media, including wire lines, optical fiber cables, and wireless (microwave radio). Whatever the physical medium used for transmission of the information, the essential feature is that the transmitted signal is corrupted in a random manner by a variety of possible mechanisms, such as additive *thermal noise* generated by electronic devices; man-made noise, e.g., automobile ignition noise; and atmospheric noise, e.g., electrical lightning discharges during thunderstorms.

At the receiving end of a digital communication system, the *digital demodulator* processes the channel-corrupted transmitted waveform and reduces the waveforms to a sequence of numbers that represent estimates of the transmitted data symbols (binary or M-ary). This sequence of numbers is passed to the channel decoder, which attempts to reconstruct the original information sequence from knowledge of the code used by the channel encoder and the redundancy contained in the received data.

A measure of how well the demodulator and decoder perform is the frequency with which errors occur in the decoded sequence. More precisely, the average probability of a bit-error at the output of the decoder is a measure of the performance of the demodulator–decoder combination. In general, the probability of error is a function of the code characteristics, the types of waveforms used to transmit the information over the channel, the transmitter power, the characteristics of the channel (i.e., the amount of noise, the nature of the interference), and the method of demodulation and decoding. These items and their effect on performance will be discussed in detail in subsequent chapters.

As a final step, when an analog output is desired, the source decoder accepts the output sequence from the channel decoder and, from knowledge of the source encoding method used, attempts to reconstruct the original signal from the source. Because of channel decoding errors and possible distortion introduced by the source encoder, and perhaps, the source decoder, the signal at the output of the source decoder is an approximation to the original source output. The difference or some function of the difference between the original signal and the reconstructed signal is a measure of the distortion introduced by the digital communication system.

1.2
COMMUNICATION CHANNELS AND THEIR CHARACTERISTICS

As indicated in the preceding discussion, the communication channel provides the connection between the transmitter and the receiver. The physical channel may be a pair of wires that carry the electrical signal, or an optical fiber that carries the information on a modulated light beam, or an underwater ocean channel in which the information is transmitted acoustically, or free space over which the information-bearing signal is radiated by use of an antenna. Other media that can be characterized as communication channels are data storage media, such as magnetic tape, magnetic disks, and optical disks.

One common problem in signal transmission through any channel is additive noise. In general, additive noise is generated internally by components such as resistors and solid-state devices used to implement the communication system. This is sometimes called *thermal noise*. Other sources of noise and interference may arise externally to the system, such as interference from other users of the channel. When such noise and interference occupy the same frequency band as the desired signal, their effect can be minimized by the proper design of the transmitted signal and its demodulator at the receiver. Other types of signal degradations that may be encountered in transmission over the channel are signal attenuation, amplitude and phase distortion, and multipath distortion.

The effects of noise may be minimized by increasing the power in the transmitted signal. However, equipment and other practical constraints limit the power level in the transmitted signal. Another basic limitation is the available channel bandwidth. A bandwidth constraint is usually due to the physical limitations of the medium and the electronic components used to implement the transmitter and the receiver. These two limitations constrain the amount of data that can be transmitted reliably over any communication channel as we shall observe in later chapters. Below, we describe some of the important characteristics of several communication channels.

Wireline channels. The telephone network makes extensive use of wire lines for voice signal transmission, as well as data and video transmission. Twisted-pair wire lines and coaxial cable are basically guided electromagnetic channels that provide relatively modest bandwidths. Telephone wire generally used to connect a customer to a central office has a bandwidth of several hundred kilohertz (kHz). On the other hand, coaxial cable has a usable bandwidth of several megahertz (MHz). Figure 1.2–1 illustrates the frequency range of guided electromagnetic channels, which include waveguides and optical fibers.

Signals transmitted through such channels are distorted in both amplitude and phase and further corrupted by additive noise. Twisted-pair wireline channels are also prone to crosstalk interference from physically adjacent channels. Because wireline channels carry a large percentage of our daily communications around the country and the world, 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. In Chapter 9, we describe methods for designing optimum transmitted signals and their demodulation; in Chapters 10 and 11, we consider the design of channel equalizers that compensate for amplitude and phase distortion on these channels.

Fiber-optic channels. Optical fibers offer the communication system designer a channel bandwidth that is several orders of magnitude larger than coaxial cable channels. During the past two decades, optical fiber cables have been developed that have a relatively low signal attenuation, and highly reliable photonic devices have been developed for signal generation and signal detection. These technological advances have resulted in a rapid deployment of optical fiber channels, both