国外电子与通信教材系列

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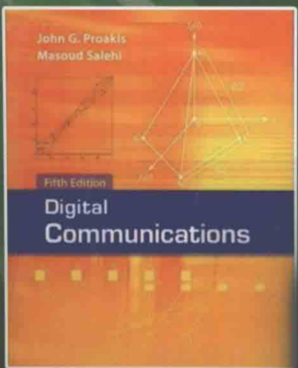
数字通信

（第五版·英文精简版）



Digital Communications

Fifth Edition

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国外电子与通信教材系列

数 字 通 信

（第五版·英文精简版）

Digital Communications（Fifth Edition）

[美] John G. Proakis　Masoud Salehi　著

张力军　等改编

電子工業出版社

Publishing House of Electronics Industry

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内 容 简 介

本书是在《数字通信（第五版）》的基础上，根据国内的实际教学情况进行精简和改编的。主要的精简原则为：保留信号传输理论内容，舍去信息传输理论内容，并以传统而经典的数字传输理论为主，无线通信为辅。改编的部分主要是根据国内实际教学的常用习惯来进行的。精简后的内容主要涵盖：确定与随机信号分析，数字调制方法，AWGN 信道的最佳接收机，载波和符号同步，通过带限信道的数字通信，自适应均衡，多信道和多载波系统，数字通信用扩频信号，衰落信道：信道特征与信号传输，多天线系统。

本书取材新颖，讨论问题系统全面、逐步深入、概念清晰，理论分析严谨、逻辑性强，习题丰富，适合作为信息和通信专业高年级本科生和研究生的教材，也可作为相关专业的教师和科技工作者的参考书。

John G. Proakis, Masoud Salehi
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序

2001年7月间，电子工业出版社的领导同志邀请各高校十几位通信领域方面的老师，商量引进国外教材问题。与会同志对出版社提出的计划十分赞同，大家认为，这对我国通信事业、特别是对高等院校通信学科的教学工作会很有好处。

教材建设是高校教学建设的主要内容之一。编写、出版一本好的教材，意味着开设了一门好的课程，甚至可能预示着一个崭新学科的诞生。20世纪40年代MIT林肯实验室出版的一套28本雷达丛书，对近代电子学科、特别是对雷达技术的推动作用，就是一个很好的例子。

我国领导部门对教材建设一直非常重视。20世纪80年代，在原教委教材编审委员会的领导下，汇集了高等院校几百位富有教学经验的专家，编写、出版了一大批教材；很多院校还根据学校的特点和需要，陆续编写了大量的讲义和参考书。这些教材对高校的教学工作发挥了极好的作用。近年来，随着教学改革不断深入和科学技术的飞速进步，有的教材内容已比较陈旧、落后，难以适应教学的要求，特别是在电子学和通信技术发展神速、可以讲是日新月异的今天，如何适应这种情况，更是一个必须认真考虑的问题。解决这个问题，除了依靠高校的老师和专家撰写新的符合要求的教科书外，引进和出版一些国外优秀电子与通信教材，尤其是有选择地引进一批英文原版教材，是会有好处的。

一年多来，电子工业出版社为此做了很多工作。他们成立了一个“国外电子与通信教材系列”项目组，选派了富有经验的业务骨干负责有关工作，收集了230余种通信教材和参考书的详细资料，调来了100余种原版教材样书，依靠由20余位专家组成的出版委员会，从中精选了40多种，内容丰富，覆盖了电路理论与应用、信号与系统、数字信号处理、微电子、通信系统、电磁场与微波等方面，既可作为通信专业本科生和研究生的教学用书，也可作为有关专业人员的参考材料。此外，这批教材，有的翻译为中文，还有部分教材直接影印出版，以供教师用英语直接授课。希望这些教材的引进和出版对高校通信教学和教材改革能起一定作用。

在这里，我还要感谢参加工作的各位教授、专家、老师与参加翻译、编辑和出版的同志们。各位专家认真负责、严谨细致、不辞辛劳、不怕琐碎和精益求精的态度，充分体现了中国教育工作者和出版工作者的良好美德。

随着我国经济建设的发展和科学技术的不断进步，对高校教学工作会不断提出新的要求和希望。我想，无论如何，要做好引进国外教材的工作，一定要联系我国的实际。教材和学术专著不同，既要注意科学性、学术性，也要重视可读性，要深入浅出，便于读者自学；引进的教材要适应高校教学改革的需要，针对目前一些教材内容较为陈旧的问题，有目的地引进一些先进的和正在发展中的交叉学科的参考书；要与国内出版的教材相配套，安排好出版英文原版教材和翻译教材的比例。我们努力使这套教材能尽量满足上述要求，希望它们能放在学生们的课桌上，发挥一定的作用。

最后，预祝“国外电子与通信教材系列”项目取得成功，为我国电子与通信教学和通信产业的发展培土施肥。也恳切希望读者能对这些书籍的不足之处、特别是翻译中存在的问题，提出意见和建议，以便再版时更正。

吴佑寿

中国工程院院士、清华大学教授

“国外电子与通信教材系列”出版委员会主任

出版说明

进入21世纪以来，我国信息产业在生产和科研方面都大大加快了发展速度，并已成为国民经济发展的支柱产业之一。但是，与世界上其他信息产业发达的国家相比，我国在技术开发、教育培训等方面都还存在着较大的差距。特别是在加入WTO后的今天，我国信息产业面临着国外竞争对手的严峻挑战。

作为我国信息产业的专业科技出版社，我们始终关注着全球电子信息技术的发展方向，始终把引进国外优秀电子与通信信息技术教材和专业书籍放在我们工作的重要位置上。在2000年至2001年间，我社先后从世界著名出版公司引进出版了40余种教材，形成了一套“国外计算机科学教材系列”，在全国高校以及科研部门中受到了欢迎和好评，得到了计算机领域的广大教师与科研工作者的充分肯定。

引进和出版一些国外优秀电子与通信教材，尤其是有选择地引进一批英文原版教材，将有助于我国信息产业培养具有国际竞争能力的技术人才，也将有助于我国国内在电子与通信教学工作中掌握和跟踪国际发展水平。根据国内信息产业的现状、教育部《关于“十五”期间普通高等教育教材建设与改革的意见》的指示精神以及高等院校老师们反映的各种意见，我们决定引进“国外电子与通信教材系列”，并随后开展了大量准备工作。此次引进的国外电子与通信教材均来自国际著名出版商，其中影印教材约占一半。教材内容涉及的学科方向包括电路理论与应用、信号与系统、数字信号处理、微电子、通信系统、电磁场与微波等，其中既有本科专业课程教材，也有研究生课程教材，以适应不同院系、不同专业、不同层次的师生对教材的需求，广大师生可自由选择和自由组合使用。我们还将与国外出版商一起，陆续推出一些教材的教学支持资料，为授课教师提供帮助。

此外，“国外电子与通信教材系列”的引进和出版工作得到了教育部高等教育司的大力支持和帮助，其中的部分引进教材已通过“教育部高等学校电子信息科学与工程类专业教学指导委员会”的审核，并得到教育部高等教育司的批准，纳入了“教育部高等教育司推荐——国外优秀信息科学与技术系列教学用书”。

为做好该系列教材的翻译工作，我们聘请了清华大学、北京大学、北京邮电大学、南京邮电大学、东南大学、西安交通大学、天津大学、西安电子科技大学、电子科技大学、中山大学、哈尔滨工业大学、西南交通大学等著名高校的教授和骨干教师参与教材的翻译和审校工作。许多教授在国内电子与通信专业领域享有较高的声望，具有丰富的教学经验，他们的渊博学识从根本上保证了教材的翻译质量和专业学术方面的严格与准确。我们在此对他们的辛勤工作与贡献表示衷心的感谢。此外，对于编辑的选择，我们达到了专业对口；对于从英文原书中发现的错误，我们通过与作者联络、从网上下载勘误表等方式，逐一进行了修订；同时，我们对审校、排版、印制质量进行了严格把关。

今后，我们将进一步加强同各高校教师的密切关系，努力引进更多的国外优秀教材和教学参考书，为我国电子与通信教材达到世界先进水平而努力。由于我们对国内外电子与通信教育的发展仍存在一些认识上的不足，在选题、翻译、出版等方面的工作中还有许多需要改进的地方，恳请广大师生和读者提出批评及建议。

电子工业出版社

教材出版委员会

PREFACE
前 言

John G. Proakis和Masoud Salehi所著的《数字通信（第五版）》是著者多年教学和科研的总结，是一本比较全面、系统、深入论述数字通信理论的经典力作，在学术界有很大的影响，同时也是一本优秀的研究生教材，多年来，国内外许多高等院校普遍采用本书作为信息和通信专业的研究生教材。《数字通信（第五版）》有16章，共800多页/1100多页（中译版/英文版），内容十分丰富。麦格劳-希尔（亚洲）教育出版公司和电子工业出版社考虑到国内高等院校相关专业教学实际情况和读者的需求，计划在《数字通信（第五版）》的基础上适当地改编出版本书的精简版，要求精简版的篇幅大约为完全版的一半，以更好地适应国内教学和读者的需求。

由于精简的篇幅相当大，确定精简原则并制定一个周密的精简方案是非常重要的。为此，有必要先对《数字通信（第五版）》的结构和内容进行分析，全书的内容大致可分为三大部分：第一部分为数字传输理论（第1、2、3、4、5、9、10等章，约占全书篇幅44%），主要内容是论述通信信号、数字调制、同步和自适应均衡等；第二部分为信息传输理论（第6、7、8章，约占全书篇幅24%），主要论述信息论基础、信源编码和信道纠错编码等；第三部分为无线通信基础（第11、12、13、14、15、16等章，约占全书篇幅32%），主要论述衰落信道、扩频、多载波、多天线、多用户通信等。这三部分内容不是截然分隔，而是相互交叉、紧密联系的完整的理论体系。例如，第一部分是传统的数字通信理论基础，当然也是无线通信的基础。第三部分在讨论无线通信的论题时，更多地应用了第二部分的信息论与编码、信道纠错编码的理论知识。

考虑精简的要求和实际的教学情况，并结合多年的教学经验，确定以下的精简原则：保留信号传输理论内容（上述第一和第三两个部分），舍去信息传输理论内容（上述第二部分），并以传统而经典的数字传输理论为主，无线通信为辅。第一部分基本完整地保留数字传输理论基础体系，主要精简其中比较深入的高级论题的内容。第二部分全部精简。第三部分无线通信基础除了精简与第二部分有关的内容外，其余均保留。

精简方案如下：

整章精简的有第6、7、8、14、16五章。主要鉴于以下的考虑：要求精简的幅度很大同时也不可能大幅度改写原著各章的内容，许多院校都开设了《信息论与编码》和《纠错编码》等课程，课时有限等。

整章保留的有第3、5、11、13四章。

部分精简的有第1、2、4、9、10、12、15七章。部分精简的内容主要基于以下的考虑：非基本教学内容或后续专业课有更深入的介绍，深入的高级论题，与信息论及编码有关的内

容，篇幅限制等。

在第 2 章中，除了精简部分小节内容外，还对 2.1 节（带通与低通信号的表示）和 2.9 节（带通和低通随机过程）的内容进行了改写。2.1 节的名称改为“带通信号与系统的表示”，本节包含 2.1.1 节（带通信号的表示法）和 2.1.2 节（带通系统对带通信号的响应）。2.9 节的名称改为“带通平稳随机过程”。

2.1 节的改写方法主要参照《数字通信（第四版）》中 4.1 节（带通信号与系统的表示）。改写主要基于以下的考虑：

（1）《数字通信（第五版）》中 2.1 节介绍的傅里叶变换知识在先修课程“信号与系统”中已学过，可以省去。

（2）《数字通信（第五版）》中 2.1 节介绍的带通系统输出复包络有－1/2 系数[见式（2-1-30）]，这种表示方法不如《数字通信（第四版）》的表示方法[见式（4-1-36）]好，当然这与两版的带通系统的表示方法不同有关。根据教学经验，《数字通信（第四版）》的表示方法比较好，读者更容易理解、记忆并掌握带通信号与系统的等效低通分析方法及其相关的概念。其实，《数字通信（第四版）》的表示方法早在 S.斯坦与 J. J. 琼斯著《现代通信原理》（科学出版社，1979 年，英文原版 S. Stein, J. J. Jones, Modern Communication Principles with Application to Digital Signaling, McGraw-Hill Inc.,1967 年）中就已论述了。这已成为传统而经典的表示方法和习惯，被普遍采用。此外，还明确地给出了《数字通信（第四版）》已有的复互相关系数定义式。

2.9 节的改写方法主要参照《数字通信（第四版）》中的 4.1.4 节（带通平稳随机过程的表示法）。这主要基于以下的考虑：该节主要论述窄带高斯噪声的数学表示和统计特性，这在国内许多本科通信原理教材和课程中都有详细的论述，并与《数字通信（第四版）》论述一致，是一种传统的经典论述方式。因此采用《数字通信（第四版）》的论述和表示方法对国内大多数读者比较熟悉也容易接受和掌握。

本书是在《数字通信（第五版）》（张力军、张宗橙、宋荣芳、曹士坷等译）的基础上进行的，参与本书改编工作的还有张宗橙、宋荣方、曹士柯、曹轩宇、张晓辉、杨文、张海江、张杰、马平、周国平、孟云飞、周克琴。完全版的内容精简后，再进行整合，对章节、公式、图表等编号进行必要的调整，最终完成的精简版整体架构仍保持与完全版一致，共有 11 章，其中数字传输理论基础内容约占 70%，无线通信基础内容约占 30%。

本书（中、英文精简版）由电子工业出版社和麦格劳-希尔（亚洲）教育出版公司委托张力军负责内容精选和改编，两家出版社为本书的出版和提高出版质量做出了很大的努力，在此表示诚挚的谢意。限于编译者的专业水平和教学经验，精简版难免有疏漏和不当之处，敬请读者不吝指正。

张力军
于南京邮电大学

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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 computer, 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 through the channel. Thus, the added redundancy serves to increase the reliability of the received data and improves

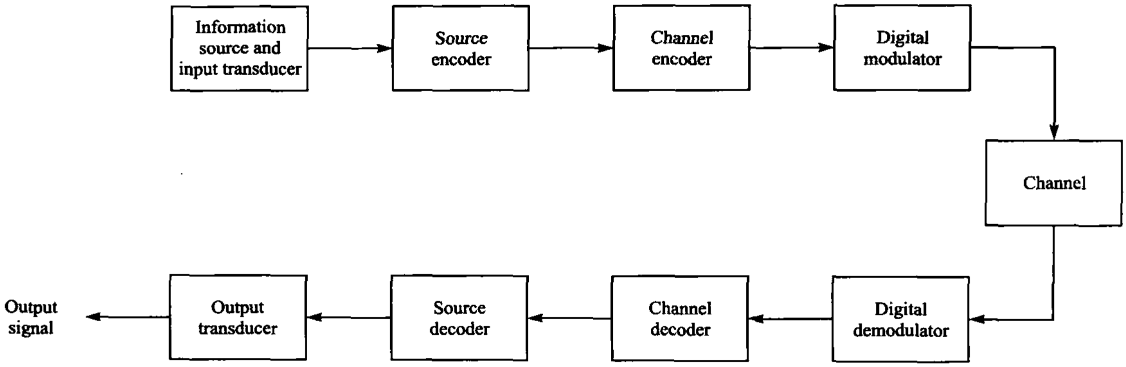

FIGURE 1.1–1
Basic elements of a digital communication system.

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