

国外电子与通信教材系列

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数字信号处理

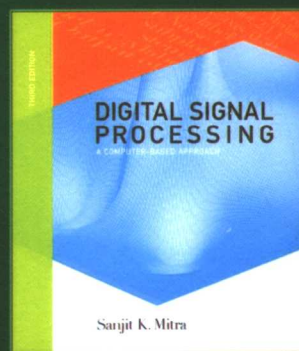
—— 基于计算机的方法 (第三版)

Digital Signal Processing

A Computer-Based Approach, Third Edition

[美] Sanjit K. Mitra 著

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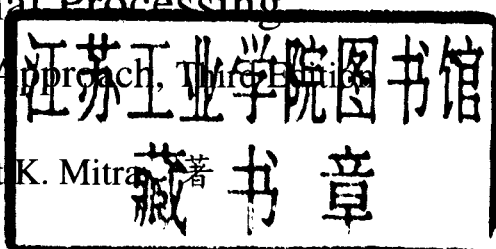
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北京·BEIJING

内 容 简 介

本书是在数字信号领域的经典教材 *Digital Signal Processing: A Computer-Based Approach* (第三版) 的基础上改编而成的, 内容涵盖信号与信号处理、时域中的离散时间信号与系统、变换域中的离散时间信号、变换域中的 LTI 离散时间系统、连续时间信号的数字处理、数字滤波器的结构与与设计等方面。本书的特点是在讲解上述内容的同时, 给出了 MATLAB 程序验证, 并有大量的高质量习题和仿真作业。

本书可作为高等院校电子信息类专业本科生或低年级研究生的教材, 也可供有关技术、科研管理人员使用, 或作为继续教育的参考书。

Sanjit K. Mitra: *Digital Signal Processing: A Computer-Based Approach, Third Edition.*

ISBN 0-07-286546-6

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Reprint adapted edition jointly published by McGraw-Hill Education (Asia) Co. and Publishing House of Electronics Industry. Copyright © 2006.

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版权贸易合同登记号 图字: 01-2006-0282

图书在版编目 (CIP) 数据

数字信号处理: 基于计算机的方法: 第3版 / (美) 米特拉 (Mitra, S. K.) 著; 阎永红改编

北京: 电子工业出版社, 2006.3

(国外电子与通信教材系列)

书名原文: *Digital Signal Processing: A Computer-Based Approach, Third Edition*

ISBN 7-121-02367-9

I. 数... II. ①米... ②阎... III. 数字信号-信号处理-教材-英文 IV. TN911.72

中国版本图书馆 CIP 数据核字 (2006) 第 018303 号

责任编辑: 窦 昊

印 刷: 北京市天竺颖华印刷厂

出版发行: 电子工业出版社

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: 787 × 980 1/16 印张: 31.5 字数: 706 千字

印 次: 2006 年 3 月第 1 次印刷

定 价: 45.00 元

凡购买电子工业出版社的图书, 如有缺损问题, 请向购买书店调换; 若书店售缺, 请与本社发行部联系。联系电话: (010) 68279077。质量投诉请发邮件至 zltz@phei.com.cn, 盗版侵权举报请发邮件至 dbqq@phei.com.cn。

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 and 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。



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 and 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”。

为作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。

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改编者的话

自教育部于2001年出台《关于加强高等学校本科教学工作提高教学质量的若干意见》以来,双语教学工作在许多高校开展起来,尽管对于双语教学的意义、效果的讨论一直没有中断过,但发展现状表明双语教学仍在较为迅速地发展。

不同院校开展双语教学的层次不同,最基本的要求是采用原版教材,主要是学习国外先进的教学思想、科学技术和语言文化;由于文化传统、教学体系的差异,原版教材与我国教学体系和教学习惯的矛盾是存在的。

“数字信号处理”是开展双语教学较早、较为普及的课程,很多学校都选择了Sanjit K. Mitra编写的*Digital Signal Processing: A Computer Based Algorithm* (第二版)的影印版作为课本。Mitra博士是国际著名的信号处理专家、IEEE电路与系统学会的主席和多家国际著名学术刊物的编委,他的教材不仅详细介绍了数字信号处理的经典内容,也介绍了许多数字信号处理的最新进展及应用实例,是理论与实践结合的典范。书中有大量的例题、高质量的练习题和上机仿真作业,并用MATLAB来演示结果。该教材在美国加州大学、哥伦比亚大学、明尼苏达大学等著名大学使用,销量大。教材共计有868页,是按照一年的教学计划编写的。

原版教材的内容叙述非常详尽,实践性强,使得原版书通常很厚,容量是国内自编教材的几倍。国内高校的“数字信号处理”课程基本是一个学期完成的,因此在实际的教学过程中,教材的1/3内容基本不涉及。由于国内双语教学的开展还没有系统化,学生的英文水平也有较大差异,造成了学生的学习负担加重。因此,对原版教材进行改编,以适应国内课程内容及教学改革的要求,都是一件有意义的工作。

该书第三版的出版,为原版教材改编提供了契机,电子工业出版社积极联系原书作者和版权单位,并委托我在第三版教材的基础上提出详细的改编计划,在原书作者授权同意的前提下,对本书的内容进行了改编。

在改编教材的过程中,主要考虑是在保证教学内容的前提下,压缩教材的篇幅,减轻学生的学习负担,同时兼顾到不同院校的具体教学大纲,尽可能使选择的内容覆盖中文数字信号处理的内容。教材的编排仍沿用原版教材的体系,剪掉了需要后续课程知识的实例以及随机信号处理的相关内容,并把抽样理论统一并入第4章。在模拟信号数字化处理中,去掉了相关的模拟芯片的理论分析,简化了线性卷积的计算和格型滤波器的结构等内容,在第8章引入了信号流图的表示。滤波器设计方面,采用频率变换的方法来完成带阻、带通滤波器的设计,第11章仅介绍DFT的算法。改编过程中,我们将原出版商提供的勘误信息直接体现在了改编本中。原书将示例程序等放在随书光盘中提供给读者,而我们的改编本则将原光盘中相关的资源放在电子工业出版社的网站(www.phei.com.cn)上供读者下载,以降低教材的价格。

由于本人初次尝试原版教材的改编工作,加上个人能力的有限,在改编中存在这样、那样的不足,恳请提出批评意见。

阎永红
2006.2.15

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1 Signals and Signal Processing

Signals play an important role in our daily life. Examples of signals that we encounter frequently are speech, music, picture, and video signals. A signal is a function of independent variables such as time, distance, position, temperature, and pressure. For example, speech and music signals represent air pressure as a function of time at a point in space. A black-and-white picture is a representation of light intensity as a function of two spatial coordinates. The video signal in television consists of a sequence of images, called frames, and is a function of three variables: two spatial coordinates and time.

Most signals we encounter are generated by natural means. However, a signal can also be generated synthetically or by computer simulation. A signal carries information, and the objective of signal processing is to extract useful information carried by the signal. The method of information extraction depends on the type of signal and the nature of the information being carried by the signal. Thus, roughly speaking, signal processing is concerned with the mathematical representation of the signal and the algorithmic operation carried out on it to extract the information present. The representation of the signal can be in terms of basis functions in the domain of the original independent variable(s), or it can be in terms of basis functions in a transformed domain. Likewise, the information extraction process may be carried out in the original domain of the signal or in a transformed domain. This book is concerned with discrete-time representation of signals and their discrete-time processing.

This chapter provides an overview of signals and signal processing methods. The mathematical characterization of the signal is first discussed along with a classification of signals. Next, some typical signals are discussed in detail, and the type of information carried by them is described. Then, a review of some commonly used signal processing operations is provided and illustrated through examples. A brief review of some typical signal processing applications is discussed next. Finally, the advantages and disadvantages of digital processing of signals are discussed.

1.1 Characterization and Classification of Signals

Depending on the nature of the independent variables and the value of the function defining the signal, various types of signals can be defined. For example, independent variables can be continuous or discrete. Likewise, the signal can either be a continuous or a discrete function of the independent variables. Moreover, the signal can be either a real-valued function or a complex-valued function.

A signal can be generated by a single source or by multiple sources. In the former case, it is a scalar signal, and in the latter case, it is a vector signal, often called a multichannel signal.

A one-dimensional (1-D) signal is a function of a single independent variable. A two-dimensional (2-D) signal is a function of two independent variables. A multidimensional (M -D) signal is a function of more than one variable. The speech signal is an example of a 1-D signal where the independent variable is time. An image signal, such as a photograph, is an example of a 2-D signal where the two independent variables are the two spatial variables. Each frame of a black-and-white video signal is a 2-D image signal that is a function of two discrete spatial variables, with each frame occurring sequentially at discrete instants of time. Hence, the black-and-white video signal can be considered an example of a three-dimensional (3-D) signal where the three independent variables are the two spatial variables and time. A color video signal

is a three-channel signal composed of three 3-D signals representing the three primary colors: red, green, and blue (RGB). For transmission purposes, the RGB television signal is transformed into another type of three-channel signal composed of a luminance component and two chrominance components.

The value of the signal at a specific value(s) of the independent variable(s) is called its *amplitude*. The variation of the amplitude as a function of the independent variable(s) is called its *waveform*.

For a 1-D signal, the independent variable is usually labeled as *time*. If the independent variable is continuous, the signal is called a *continuous-time signal*. If the independent variable is discrete, the signal is called a *discrete-time signal*. A continuous-time signal is defined at every instant of time. On the other hand, a discrete-time signal takes certain numerical values at specified discrete instants of time, and between these specified instants of time, the signal is not defined. Hence, a discrete-time signal is basically a sequence of numbers.

A continuous-time signal with a continuous amplitude is usually called an *analog signal*. A speech signal is an example of an analog signal. Analog signals are commonly encountered in our daily life and are usually generated by natural means. A discrete-time signal with discrete-valued amplitudes represented by a finite number of digits is referred to as a *digital signal*. An example of a digital signal is the digitized music signal stored in a CD-ROM disk. A discrete-time signal with continuous-valued amplitudes is called a *sampled-data signal*. This last type of signal occurs in switched-capacitor (SC) circuits. A digital signal is thus a quantized sampled-data signal. Finally, a continuous-time signal with discrete-valued amplitudes has been referred to as a *quantized boxcar signal* [Ste93]. The latter type of signals occurs in digital electronic circuits where the signal is kept at fixed level (usually one of two values) between two instants of clocking. Figure 1.1 illustrates the four types of signals.

The functional dependence of a signal in its mathematical representation is often explicitly shown. For a continuous-time 1-D signal, the continuous independent variable is usually denoted by t , whereas for a discrete-time 1-D signal, the discrete independent variable is usually denoted by n . For example, $u(t)$ represents a continuous-time 1-D signal and $\{v[n]\}$ represents a discrete-time 1-D signal. Each member, $v[n]$, of a discrete-time signal is called a *sample*. In many applications, a discrete-time signal is generated from a parent continuous-time signal by sampling the latter at uniform intervals of time. If the discrete instants of time at which a discrete-time signal is defined are uniformly spaced, the independent discrete variable n can be normalized to assume integer values.

In the case of a continuous-time 2-D signal, the two independent variables are usually the spatial coordinates, which are usually denoted by x and y . For example, the intensity of a black-and-white image can be expressed as $u(x, y)$. A color image $u(x, y)$, is composed of three signals representing the three primary colors, red, green, and blue:

$$\mathbf{u}(x, y) = \begin{bmatrix} r(x, y) \\ g(x, y) \\ b(x, y) \end{bmatrix}.$$

On the other hand, a digitized image is a 2-D discrete signal, and its two independent variables are discretized spatial variables often denoted by m and n . Hence, a digitized image can be represented as $v[m, n]$. Likewise, a black-and-white video sequence is a 3-D signal and can be represented as $u(x, y, t)$, where x and y denote the two spatial variables and t denotes the temporal variable time. A color video signal is a vector signal composed of three video signals representing the three primary colors, red, green, and blue.

There is another classification of signals that depends on the certainty by which the signal can be uniquely described. A signal that can be uniquely determined by a well-defined process such as a mathematical expression or rule, or table look-up, is called a *deterministic signal*. A signal that is generated in a random fashion and cannot be predicted ahead of time is called a *random signal*. In this text, we are primarily concerned with the processing of discrete-time deterministic signals.

Some typical signal processing operations performed on analog signals are reviewed in the following section.



Speech Demo 1



Image Demo 1

Video Demo 1