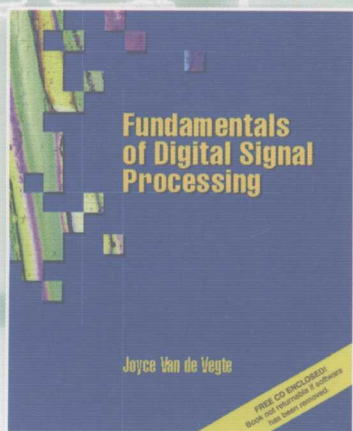


国外电子与通信教材系列

英文版

PEARSON
Prentice
Hall

数字信号处理基础



Fundamentals of Digital Signal Processing

[加] Joyce Van de Vegte 著

尹霄丽 改编



电子工业出版社
PUBLISHING HOUSE OF ELECTRONICS INDUSTRY

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Publishing House of Electronics Industry

北京·BEIJING

内 容 简 介

本书在 Joyce Van de Vegte 所著 Fundamentals of Digital Signal Processing 的基础上进行了缩减, 针对本科教学大纲, 删除了教学中一般不涉及的内容, 修订了部分印刷错误, 并增加了循环卷积的内容, 使教材的内容更完整。全书在概述了数字信号的产生、定义和处理实例之后, 详细讨论了差分方程、数字卷积、 z 变换、离散时间傅里叶变换和傅里叶级数、数字滤波器、传输函数、频率响应、频谱、无限和有限脉冲响应数字滤波器的设计以及离散傅里叶变换和快速傅里叶变换、循环卷积等基本概念和基本理论。书中涉及的数学知识以简明形式给出, 深入浅出, 易于理解。本书示例丰富, 并附有大量的例题和习题。

本书可作为理工类大专院校电子、计算机、通信等专业的本科生双语教学教材, 对于 DSP 领域的工程技术人员也有很好的参考价值。

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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 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。

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改编者序

数字信号处理目前成为大部分高等院校电子、电气、通信等专业首选的专业基础课。对于计算机工程等专业的学生来说,学习数字信号处理课程也是非常有利的,能够懂得如何把数学作为思考工程问题的语言,建立数字计算与系统实现之间的联系,并为后续课程打下有用的基础。

本书在Joyce Van de Vegte所著Fundamentals of Digital Signal Processing的基础上进行了缩减,针对本科教学大纲,删除了教学中一般不涉及的内容,修订了部分印刷错误,并增加了循环卷积的内容,使教材的内容更完整。

北京邮电大学电子工程学院从2003年开始,使用此教材进行了数字信号处理双语教学(2学分课程)的授课和课程建设,期间得到了教育部理工科基地双语教学课程建设项目“数字信号处理双语教学课程建设”的立项资助。我们认为这本教材内容覆盖全面,深入浅出,比较适合低年级本科生和学时较少的课程教学。我们使用该教材5年来,体会到它的4个最显著的特点:

1. 在内容结构方面,本书系统地介绍了数字信号处理的基本概念和基本原理——信号的频谱计算和数字滤波器的设计。值得一提的是,本书将滤波的概念与数字信号处理技术紧密结合起来(第4章讲差分方程与滤波,第5章讲卷积与滤波),使学生在学习这些基本运算的同时了解滤波的处理方法。鉴于FFT目前已成为非常普遍的算法,有大量的软硬件来实现,所以作者将FFT算法介绍安排在数字滤波器设计之后,即在第7章讲数字滤波器的频率响应特性,在第8章介绍数字信号的频谱,分别给出了非周期信号和周期信号频谱的分析方法,第9章和第10章分别讲解FIR和IIR数字滤波器的设计。这样做可以使学生在理解了信号的频谱(DFS和DFT)及数字滤波器的频率响应特性的基础上,有充足的时间来完成数字滤波器设计任务,直接使用MATLAB的fft等函数来计算滤波器的频率响应特性,从而克服了以往在接近学期末学习滤波器设计时学生只是了解了算法而没有时间用于设计实践的矛盾。
2. 在内容编排上,本书采用深入浅出的方法处理与数字信号处理相关的数学知识,突出物理概念的阐述,易于理解,为大学本科低年级学生的学习和工程技术人员的自学提供了便利条件。
3. 在材料选择上,本书中包含了大量实用的信号和系统及其分析,例题和习题丰富,对于读者自学或巩固课堂学习内容都非常有帮助。
4. 在语言表达上,作者Joyce Van de Vegte为加拿大Camosun学院计算机与电子工程技术系教授,该书语言流畅,有利于提高学生的英语实践能力。

与本书配套的学习辅助资料包括一些声音文件、图像文件、数据、视频片段、频谱图以及基于HTML的软件,页边的太阳形符号表明有相应的资料。读者可以通过运行runDSP打开与课程配套的教学软件,其内容包括多媒体形式的学习总结(multimedia)、小测验(quiz)和MATLAB辅导。这些辅导资料可使读者在学习实际听到和看到处理前后的信号,加深对基础知识的理解。另外,本书作者提供了教师教学辅助材料,包括用PowerPoint制作的主要图表的幻灯片、实验指导、作业及小测验答案^①。

在本书改编过程中,结合了本人在北京邮电大学电子工程学院讲授双语课程的经验,北京邮电大学周利清教授和张健明老师给予了本人许多指导和帮助,在此表示衷心感谢!

^① 上述资料的获取方式,参见本书末所附“教学支持说明”。

PREFACE

Digital signal processing (DSP) can no longer be considered the domain of graduate students and researchers. It now pervades the technology that we take for granted in our homes and offices, and its influence is growing. This book was written to create an accessible resource for college students, engineers, and computer scientists wanting to gain a working knowledge of the principles, applications, and language of DSP.

First and foremost, DSP is exciting! The author attempts to give readers a sense of this excitement immediately by starting with a nonmathematical crash course in DSP. In the chapters that follow, examples frequently focus on real-life sounds—such as speech, whale songs, and seismic vibrations—and on real-life images—such as fingerprints, bacteria, and airport X-rays. Once the necessary theory has been covered, in-depth applications of speech recognition, image processing, motor control, and encryption are studied, among others.

All key concepts of DSP are covered in this text, including details of how to perform transforms and design filters. This coverage is heavily supported by examples throughout. To make the ideas as accessible as possible, no calculus is used at all in the main text. While the mathematical techniques that are used are not trivial, they are always presented in as straightforward a manner as possible. Even students with strong mathematical backgrounds will appreciate the chance to focus on issues of DSP rather than mathematics. Essential mathematical topics that are prerequisite to understanding the material in the text are included in an appendix.

Chapter 1 contains the “Crash Course in Digital Signal Processing.” It provides a surface treatment of all the major topics of the book, without going into details of the underlying mathematics. Chapter 2 explains how to obtain a digital signal from the analog signals that surround us, and Chapter 3 provides some experience with defining and handling digital signals. Chapters 4 through 8 contain the majority of the important underlying theory for DSP. Topics covered in these chapters include difference equations, digital convolution, z transforms, discrete time Fourier transforms, and discrete Fourier series. The essential concepts of filter, transfer function, frequency response, and spectrum are developed. Filter design is taken care of in Chapters 9 and 10, for both finite impulse

response and infinite impulse response filters. Practical aspects are covered beginning with Chapter 11, which discusses discrete and fast Fourier transforms. Applications of DSP for sounds and images are investigated in Chapters 12. Appendix A contains "The Math You Need," while the other appendices prove claims made in the text so that the text may stand on its own, without the need for outside references. End-of-chapter summaries and questions are provided for each chapter. All comments, suggestions, and reports of errors in the text or software will be most appreciated. They may be sent to Joyce Mills (née Van de Vegte) at millsj@camosun.bc.ca.

Because the mathematical requirements of this text are moderate, the book can be used as early as the second year of a college engineering or technology program. Increasingly, DSP will be considered an essential technical skill. Perhaps this text can be of use as the pressure grows to teach DSP earlier in the curriculum.

I must thank my colleagues in the Computer and Electronics Engineering Technology Department of Camosun College for lending their expert advice, both solicited and unsolicited, on many topics addressed herein. Faculty in other departments, particularly Stewart Langton and Mile Erlic, were also most generous with their time, and Jon Jacox and other students were kind enough to proof many of the question and solution sets. Thanks go also to the Dean of Technology, Baldev Pooni, and the Vice President of the College, Bob Priebe, for their support of this initiative.

Several individuals at Prentice Hall helped to make the text a reality. I wish to thank my editor, Charles Stewart, for his contagious excitement about the project when it first began, and also editorial representative Carmen Batsford for her excellent advice and good sense of humor throughout. A special thank you must go to Delia Uhrec, assistant editor, who promptly and expertly answered my questions and did everything possible to remove obstacles from my path. Delia's well-timed words of encouragement in the final phases of manuscript preparation were most appreciated.

As this text was being prepared, a number of reviewers provided constructive comments and suggestions that have certainly improved the finished product: Kefu Xue, Wright State University; Anthony Oxtoby, Purdue University; Mark Hihghum, Bay De Noc Community College; Charles A. Cipari, Arizona State University; and Charles J. Eckard, ITT Technical Institute. Also, my father, Dr. J. Van de Vegte, a textbook author himself, painstakingly edited not one but two complete draft manuscripts. He was surely my harshest critic, but the book is many times better as a result of his input, and my mother assures me that "rewrite this section" is just his way of saying "I love you."

In my home, my children whisper the word "textbook" reverently, as if it were one of the great and mysterious wonders of the world. I am humbled by how generously my family has accommodated my obsession. While I was ensconced at my computer, my dear husband Don juggled job, children, and housework, and only occasionally reminisced aloud about how life used to be. Indeed, the only downside to finishing the text is that I will have to start doing dishes again. My children—Stevin, Jesika, and Eric—will be joyful when my time is theirs again, as will I.

Jessy and Eric say this book will make you smarter. I hope they're right. Of course, their other idea was to use it to start the campfire: There's a lesson here somewhere.

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1

CRASH COURSE IN DIGITAL SIGNAL PROCESSING

This chapter skates across the surface of the topics to be studied in greater depth in the remainder of the text. The chapter:

- distinguishes between analog and digital signals
- presents the basic steps in analog-to-digital conversion
- presents the basic steps in digital-to-analog conversion
- introduces the relationship between a signal and its spectrum
- explains the basic concepts of filtering
- discusses applications of digital signal processing

1.1 SIGNALS AND SYSTEMS

Computers operate on digital signals. As computers proliferate, the need for efficient handling of digital signals increases. Furthermore, high speed processing capabilities of modern computers attract applications that use digital signals, which further drives the development of digital signal techniques. Digital signal processing, or DSP, is essential to an enormous variety of old and new applications, a few of which are listed in Figure 1.1.

At the heart of DSP lie the signals to be processed. Signals are variations that carry information from one place to another. The outside world, for example, provides variations in pressure or light intensity that humans can perceive. Changes in pressure at the eardrum are heard as sounds. Variations in light intensity at the retina are seen as images.

- | | |
|---|-------------------------------------|
| • Touch-Tone™ telephones | • Speech synthesis |
| • Edge detection in images | • Echo cancellation |
| • Digital signal and image filtering | • Cochlear implants |
| • Seismic analysis | • Antilock brakes |
| • Text recognition | • Signal and image compression |
| • Speech recognition | • Noise reduction |
| • Magnetic resonance image (MRI) scans | • Companding |
| • Music synthesis | • High definition television (HDTV) |
| • Bar code readers | • Digital audio |
| • Sonar processing | • Encryption |
| • Satellite image analysis | • Motor control |
| • Digital mapping | • Remote medical monitoring |
| • Cellular telephones | • Smart appliances |
| • Digital cameras | • Home security |
| • Detection of narcotics and explosives | • High speed modems |

FIGURE 1.1

Applications of DSP.

These signals are **analog signals**. They can take any value from a continuum of values and are defined at every instant of time. Sounds are one-dimensional analog signals: The size, or amplitude, of pressure variations changes with time. As another example, the voltage available from an electrical outlet in North America varies smoothly from its minimum to its maximum and back to its minimum again, 60 times per second. Figure 1.2 supplies a few examples of one-dimensional signals. Images are two-dimensional analog signals: Brightness varies along both the horizontal and vertical axes of the image. Figure 1.3 shows a sample of a black and white image, and Figure 1.4 shows four frames from a high speed digital video sequence.

In order to process signals, they must first be captured. Sound signals, for example, are acquired using a microphone, which converts acoustic signals into electrical signals. Images, on the other hand, are captured using an analog or digital camera. In analog cameras, light signals control chemical reactions on a photographic film. In digital cameras, light signals from objects in view create charge packets that are converted into electrical signals on a two-dimensional grid. These electrical signals are, like the light signals that produced them, analog in nature. Because they carry information at an infinite number of levels and points in time, analog signals are not suited to computer processing. They must be sampled and converted into digital form before they can be processed. **Digital signals** are perfectly suited for computer processing because they are defined at only a finite number of levels and points in time.

Both analog and digital signals are present in most digital processing systems. Analog signals at the input to the system are converted to digital form for processing. After processing, signals in digital form are converted back to analog form for output. It is in the pro-