国外电子与通信教材系列

数字信号处理
频谱计算与滤波器设计

Digital Signal Processing
Spectral Computation and Filter Design

英文版

[美] Chi-Tsong Chen 著

電子工業出版社
Publishing House of Electronics Industry
www.phei.com.cn

国外电子与通信教材系列

数 字 信 号 处 理

频谱计算与滤波器设计

（英文版）

Digital Signal Processing

Spectral Computation and Filter Design

[美] Chi-Tsong Chen 著

電子工業出版社
Publishing House of Electronics Industry
北京 · BEIJING

内 容 简 介

本书讲述了数字信号处理中的基本内容和基础知识，主要包括信号的频谱计算和滤波器设计两大部分。本书侧重于基本概念和程序，涵盖了这一领域的主要内容。广泛应用快速傅里叶变换（FFT）、实用性强是本书的主要特色。

本书可作为高等院校电子和通信等专业本科生教材，也可作为有关工程技术人员参考用书。

图书在版编目（CIP）数据

数字信号处理：频谱计算与滤波器设计/（美）陈其宗（Chi-Tsong Chen）著．—北京：电子工业出版社，2002.9
（国外电子与通信教材系列）
书名原文：Digital Signal Processing-Spectral Computation and Filter Design
ISBN 7-5053-7961-5

Ⅰ．数…　Ⅱ．陈…　Ⅲ．①数字信号—信号处理—高等学校—教材—英文②滤波器—设计—高等学校—教材—英文　Ⅳ．TN911.72

中国版本图书馆 CIP 数据核字（2002）第 063957 号

责任编辑：杨丽娟　徐　堃
印　　刷：北京东光印刷厂
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：787×980　1/16　印张：28.75
版　　次：2002 年 9 月第 1 版　2002 年 9 月第 1 次印刷
定　　价：43.00 元
版权贸易合同登记号　图字：01-2002-4293

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、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

This text is intended for use in a first course on digital signal processing (DSP), typically in senior or first-year graduate level. It may also be useful to engineers and scientists who use digital computers to process measured data. Some elementary knowledge on signals and systems is helpful but is not necessary. An attempt has been made to make this text as self-contained as possible.

As an introductory text, we discuss only two major topics: computer computation of frequency contents of signals and design of digital filters. These two topics follow logically a basic text or course on *Signals and Systems*, which now often covers both continuous-time (CT) and discrete-time (DT) cases. To be self-contained, we start from scratch. Chapter 1 introduces CT signals, DT signals, and digital signals. We discuss why CT signals are now widely processed digitally and how they are converted into digital signals. We also discuss why we study in this text, as in every other DSP text, only CT and DT signals even though digital signals are the signals processed on digital computers. In Chapter 2, we first discuss the frequency of CT and DT sinusoidal signals. We then introduce CT and DT Fourier series (frequency components) for periodic signals and establish their relationships. We then use the fast Fourier transform (FFT) to compute their frequency components.

Chapter 3 extends CT and DT Fourier series to CT and DT Fourier transforms (frequency spectra) that can reveal frequency contents of aperiodic signals as well. We establish the sampling theorem and discuss the effects of truncation of signals on frequency spectra. In Chapter 4, we show that computer computation of DT Fourier transform (DTFT) leads naturally to the discrete Fourier transform (DFT), which differs from the DT Fourier series only by a fraction. We then introduce a version of FFT, an efficient way of computing DFT. We use FFT to compute frequency spectra of DT and CT signals and to compute DT and CT signals from their frequency spectra. This completes the discussion of spectral computation of signals.

The second part of this text discusses the design of digital filters. We introduce in Chapter 5 the class of digital filters (linear, time-invariant, lumped, causal, and stable) to be designed and the mathematical tools (convolutions, impulse responses, z-transform, and transfer functions) to be used. We then introduce the concept of frequency responses for stable systems. Chapter 6 discusses specifications of digital filters and how to use poles and zeros to shape magnitude responses of simple digital filters. Chapter 7 introduces various methods to design finite-impulse-response (FIR) digital filters. In Chapter 8, we first discuss the reasons for not designing directly infinite-impulse-response (IIR) digital filters. We then introduce analog prototype filters, which are lowpass filters with 1 rad/s as their passband or stopband edge frequency. All other analog and all digital frequency-selective filters can then be obtained by using frequency transformations.

The last chapter discusses a number of block diagrams for digital filters and some problems due to finite-word-length implementation. This chapter is essentially independent of Chapters 7 and 8 and may be studied right after Chapter 6.

Although most topics in this text are standard, our presentation and emphasis are significantly different from other existing DSP texts. They are discussed in the following.

- Most DSP texts discuss systems and signals intertwined. This text discusses in the first part only signals and their spectral computation. Thus the discussion can be more focused. It is also more efficient for those who wish to learn how to use FFT to analyze measured data.
- This text shows that the frequency of a DT sinusoidal sequence cannot be defined from its fundamental period as in the CT case. It is then defined from a CT sinusoid and justified using a physical argument. Thus this text uses the same notation to denote frequencies in CT and DT signals as opposed to different notations used in most DSP texts. We do use different notations when the analog frequency range $(-\infty, \infty)$ is compressed into the digital frequency range $(-\pi, \pi]$ in bilinear transformations.
- Assuming the reader to have had Fourier analysis of CT signals, most DSP texts cover only Fourier analysis of DT signals. Because signals processed by DSP are mostly CT, this text covers Fourier analyses of both CT and DT signals. The discussion of the CT part, however, is not exhaustive. It is introduced to the extent to show the differences between CT and DT Fourier analyses and to establish their relationships. We establish sampling theorems for pure sinusoids, periodic signals, and general signals.
- Most DSP texts assume the sampling period T to be 1 in DT Fourier analysis. Although the equations involved are simpler, they become more complex in relating spectra of CT signals and their sampled sequences. This text does not assume $T = 1$ and discusses how to select T in spectral computation of CT signals from their time samples. In digital filter design, however, we can select T to be 1 or any other value such as π in MATLAB,[1] as discussed in Section 5.3.2.
- Many DSP texts introduce DFT after the Fourier transforms and discuss it as an independent mathematical entity Our discussion of DFT is brief because it is essentially the same as the DT Fourier series. It is introduced as a computational tool.
- Most DSP texts introduce first the two-sided z-transform and then reduce it to the one-sided z-transform. This text does not need the two-sided transform. Thus we concentrate on the one-sided z-transform and give reasons for forgoing the concept of the region of convergence, which is a difficult concept and rarely needed in application.
- Fourier analysis of DT systems are covered in most DSP texts. It is not covered in this text because Fourier analysis of DT systems is less general and more complex than the z-transform (Section 5.5.3).
- Most DSP texts use exclusively negative-power transfer functions. This text uses both negative-power and positive-power transfer functions because the latter is more convenient in introducing the concepts of properness, degrees, poles, and zeros. Furthermore, both forms are used in MATLAB.

[1] MATLAB is a trademark of the MathWorks, Inc.

- The bilinear transformation introduced in many DSP texts normalize the frequency range by assuming $T = 1$ and yet introduce the factor $T/2$ where T may be different from 1. This is inconsistent. We introduce an arbitrary factor and show that the design is independent of the factor. This is more logical and can also justify the inconsistence.

In addition, this text discusses a number of topics not found in most DSP texts. We list some of them in the following:

- Give a formal definition of the frequency of sinusoidal sequences and give a precise frequency range $(-\pi/T, \pi/T]$ for DT signals (Section 2.2).
- Establish a simplified version of the sampling theorem for periodic signals and then use FFT to compute frequency components of CT periodic signals from their sampled sequences. Discuss how to use the MATLAB functions fftshift, ceil, floor and a newly defined function shift to plot FFT computed frequency components in $(-\pi/T, \pi/T]$ or $[-\pi/T, \pi/T)$ (Sections 2.6 and 2.7).
- Discuss the nonuniqueness of inverse DFT and a method of determining the location of a time signal computed from frequency samples of its frequency spectrum (Section 4.7).
- Use FFT to compute the inverse z-transform (Section 5.4.2).
- Discuss steady-state and transient responses of digital filters, and give an estimated time for a transient response to die out (Section 5.7.1). Compare two ways of introducing the concept of frequency responses. We argue that although the concept can be more easily introduced by assuming an input to be applied from $-\infty$, two concepts of practical importance may often be concealed (Section 5.7.3).
- Give a mathematical justification of using an antialiasing analog filter in digital signal processing (Section 6.7.1).
- Introduce a discrete least squares method to design FIR filters. The method, although very simple, does not seem to appear anywhere in the literature. The method is more flexible than the method of frequency sampling and leads naturally to the MATLAB function firls (Sections 7.6 and 7.6.1).
- Introduce an analog bandstop transformation that yields better bandstop filters than the ones generated by MATLAB (Section 8.4).
- Although FFT is very efficient in computing the convolution of two long sequences, we give possible reasons for not using FFT in the MATLAB function conv (Section 9.4.1).

All terminology in this text is carefully defined. For example, both the Fourier series and Fourier transforms of periodic signals are often called discrete frequency spectra in the literature. This text reserves *frequency spectra* exclusively for the Fourier transforms, and calls Fourier series *frequency components*. We attempt to make all discussion mathematically correct; however, we will not be constrained by pure mathematics (Section 3.9 and the footnote in Section 6.3). As a text intended for practical application, the discussion is not necessarily rigorous. For example, we use the terms "very close" and "practically zero" loosely. Because of the information explosion, we all have less time to study a subject area. Thus the discussion in this text is not exhaustive; we concentrate only on concepts and results that, in our opinion, are essential in practical application.

MATLAB is an integral part of this text. We use *while loops* and *if-else-end structures* to develop MATLAB programs. However, our emphasis is not on MATLAB but rather on basic ideas and procedures in digital signal processing. Thus this text lists only essential MATLAB functions in most programs and skips functions that adjust shapes and sizes of plots, and draw horizontal and vertical coordinates. It is recommended that the reader repeat each program. Even though he may not obtain identical results, all essential information will be generated. Clearly any other DSP package can also be used.

All numerical examples in the text and most problems at the end of each chapter are very simple and can be solved analytically by hand. The reader is urged to do so. After obtaining results by hand, one can then compare them with computer-generated results. This is the best way of learning a topic and a computational tool. Once mastering the subject and tool, we can then apply the tool to real-world problems. A solutions manual, in which complete programs are listed for problems that use MATLAB, is available from the publisher.

This text is a complete restructure and rewriting of *One-Dimensional Digital Signal Processing*, published in 1979. We deleted the discussion of the two-sided z-transform, its region of convergence, and the topics such as error analyses, Wiener FIR, and IIR filters, that require statistical methods. All aforementioned sections, except Section 6.6.1, are new in this text.

I am indebted to many people in developing this text. First I like to thank Professors Petar Djuric, Adrian Leuciuc, John Murray, Nam Phamdo, Stephen Rappaport, Kenneth Short, and Armen Zemanian, my colleagues at Stony Brook. I went to them whenever I had any question or doubt. Many people reviewed a earlier version of this text. Their comments prompted me to rewrite many sections. I thank them all.

I am grateful to my editor Peter C. Gordon for his confidence in and enthusiasm about this project. Many people at Oxford University Press including Christine D'Antonio, Jacki Hartt, Anita Sung, and Jasmine Urmeneta, were most helpful in this undertaking. Finally I thank my wife, Bih-Jau, for her support.

Chi-Tsong Chen

国外电子与通信教材系列 ①

类别	中文书名	英文书名	作者	译者	版别
电路理论与应用	电路（第六版）	Electric Circuits, 6e	James W. Nilsson	路而红	中文版
	工程电路分析（第六版）	Engineering Circuit Analysis, 6e	William H. Hayt, Jr.	王大鹏	中文版 英文版
	射频电路设计 —— 理论与应用	RF Circuit Design: Theory and Applications	Reinhold Ludwig	王子宇	中文版
	电子学原理	Eletronic Concepts: An Introduction	Jerrold H. Krenz	马爱文	中文版
	数字电路简明教程	Digital Electronics: A Simplified Approach	Robert D. Thompson	高　鹏	中文版
	逻辑电路设计基础	Introduction to Logic Design	Alan B. Marcovitz	殷洪玺	中文版
	数字集成电路设计	Digital Integrated Circuit Design	Ken Martin		英文版
	线性电路分析基础（第二版）	Elementary Linear Circuit Analysis	Leonard S. Bobrow		英文版
	半导体器件电子学	Semiconductor-Device Electronics	R. M. Warner		英文版
	电路基础（第五版）	Introduction to Eelectric Circuits, 5e	Richard C. Dorf		英文版
信号与系统	信号与系统（第二版）	Signals and Systems, 2e	Alan V. Oppenheim		英文版
	应用 Web 和 MATLAB 的信号与系统基础（第二版）	Fundamentals of Signals and Systems Using the Web and MATLAB, 2e	Edward W. Kamen	高　强	中文版
	统计与自适应信号处理	Statistical and Adaptive Signal Processing	Dimitris G. Manolakis	周　正	中文版
	自适应滤波器原理（第四版）	Adaptive Filter Theory, 4e	Simon Haykin	郑宝玉	中文版 英文版
数字信号处理	数字图像处理（第二版）	Digital Image Processing, 2e	Rafael C. Gonzalez	阮秋琦	中文版 英文版
	数字信号处理基础	Fundamentals of Digital Signal Processing	Joyce Van de Vegte	侯正信	中文版
	数字通信 —— 基础与应用（第二版）	Digital Communications: Fundamentals and Applications，2e	Bernard Sklar	沈连丰	中文版 英文版
	实时信号处理 —— 信号处理系统的设计与实现	Real-Time Signal Processing: Design and Implementation of Signal Processing Systems	John G. Ackenhusen	彭启琮	中文版
	小波与傅里叶分析基础	A First Course in Wavelets with Fourier Analysis	Albert Boggess		英文版
	信号处理滤波器设计 —— 基于 MATLAB 和 Mathematica 的设计方法	Filter Design for Signal Processing Using MATLAB and Mathematica	Miroslav D. Lutovac		英文版
	数字信号处理 ——频谱计算与滤波器设计	Digital Signal Processing: Spectral Computation and Filter Design	Chi-Tsong Chen		英文版
	LabVIEW 6i 实用教程	LabVIEW 6i Student Edition	Robert H. Bishop	朱世华	中文版
	VHDL 数字系统设计	Digital System Design with VHDL	Mark Zwolinski		英文版
	MATLAB 原理与工程应用	An Engineer's Guide to MATLAB	Edward B. Magrab	高会生	中文版

国外电子与通信教材系列 ②

类别	中文书名	英文书名	作者	译者	版别
微电子	CMOS 射频集成电路设计	The Design of CMOS Radio-Frequency Integrated Circuits	Thomas H. Lee		英文版
	专用集成电路	Application-Specific Integrated Circuits	Michael J. S. Smith	章倩苓	中文版 英文版
	CMOS 模拟电路设计（第二版）	CMOS Analog Circuit Design, 2e	Phillip E. Allen		英文版
	微电子制造科学原理与工程技术	The Science and Engineering of Microelectronic Fabrication	Stephen A. Campbell	周润德	中文版
通信系统	通信系统（第四版）	Communication Systems, 4e	Simon Haykin	沈连丰	中文版 英文版
	通信系统工程（第二版）	Communication Systems Engineering, 2e	John G. Proakis	沈连丰	中文版 英文版
	电子通信系统（第四版）	Electronic Communications Systems: Fundamentals through Advanced, 4e	Wayne Tomasi	薛荣华	中文版 英文版
	数字与模拟通信系统（第六版）	Digital and Analog Communication Systems, 6e	Leon W. Couch, II	罗新民	中文版
	光纤通信（第三版）	Optical Fiber Communications, 3e	Gerd Keiser	李玉权	中文版
	光纤通信系统（第三版）	Fiber-Optic communication System, 3e	Govind P. Agrawal		英文版
	现代通信光波电子学（第五版）	Optical Electronics in Modern Communications, 5e	Amnon Yariv		英文版
	CDMA 蜂窝移动通信与网络安全	CDMA Cellular Mobile Communications and Network Security	Man Young Rhee	袁超伟	中文版 英文版
	无线通信——原理与实践（第二版）	Wireless Communications: Principles and Practice, 2e	Theodore S. Rappaport		英文版
	宽带无线数字通信	Wideband Wireless Digital Communications	Andreas Molisch	姚 彦	中文版 英文版
	第三代移动通信系统原理与工程设计——IS-95 CDMA 和 cdma 2000	IS-95 CDMA and cdma2000: Cellular/ PCS Systems Implementation	Vijay K. Garg		英文版
	视频处理与通信	Video Processing and Communications	Yao Wang	侯正信	中文版
电磁场与微波	天线（第三版）	Antennas: For All Applications, 3e	John D. Kraus	章文勋	中文版
	射频与微波电子学	Radio Frequency and Microwave Electronics Illustrated	Matthew M. Radmanesh		英文版
	现代无线通信系统电波传播	Radio Propagation for Modern Wireless Systems	Henry L. Bertoni	徐承和	中文版 英文版

目录概览

CONTENTS

PART 2 Digital Filter Design