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Discrete-time Signal Processing
Second Edition

离散时间信号处理(第2版)

Alan V. Oppenheim Ronald W. Schafer John R. Buck



SECOND EDITION

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DISCRETE-TIME SIGNAL PROCESSING

ALAN V. OPPENHEIM

MASSACHUSETTS INSTITUTE OF TECHNOLOGY

RONALD W. SCHAFER

GEORGIA INSTITUTE OF TECHNOLOGY

WITH

JOHN R. BUCK

University of Massachusetts Dartmouth

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出 版 说 明

郑大钟 清华大学信息科学与技术学院

当前,在我国的高等学校中,教学内容和课程体系的改革已经成为教学改革中的一个非常突出的问题,而为数不少的课程教材中普遍存在的"课程体系老化,内容落伍时代,本研层次不清"的现象又是其中的急需改变的一个重要方面。同时,随着科教兴国方针的贯彻落实,要求我们进一步转变观念扩大视野,使教学过程适应以信息技术为先导的技术革命和我国社会主义市场经济体制的需要,加快教学过程的国际化进程。在这方面,系统地研究和借鉴国外知名大学的相关教材,将会对推进我们的课程改革和推进我国大学教学的国际化进程,乃至对我们一些重点大学建设国际一流大学的努力,都将具有重要的借鉴推动作用。正是基于这种背景,我们决定在国内推出信息技术学科和电气工程学科国外知名大学原版系列教材。

本系列教材的组编将遵循如下的几点基本原则。(1)书目的范围限于信息技术学科和电气工程学科所属专业的技术基础课和主要的专业课。(2)教材的范围选自于具有较大影响且为国外知名大学所采用的教材。(3)教材属于在近5年内所出版的新书或新版书。(4)教材适合于作为我国大学相应课程的教材或主要教学参考书。(5)每本列选的教材都须经过国内相应领域的资深专家审看和推荐。(6)教材的形式直接以英文原版形式印刷出版。

本系列教材将按分期分批的方式组织出版。为了便于使用本系列教材的相关教师和学生 从学科和教学的角度对其在体系和内容上的特点和特色有所了解,在每本教材中都附有我们 所约请的相关领域资深教授撰写的影印版序言。此外,出于多样化的考虑,对于某些基本类 型的课程,我们还同时列选了多于一本的不同体系、不同风格和不同层次的教材,以供不同 要求和不同学时的同类课程的选用。

本系列教材的读者对象为信息技术学科和电气工程学科所属各专业的本科生,同时兼顾 其他工程学科专业的本科生或研究生。本系列教材,既可采用作为相应课程的教材或教学参 考书,也可提供作为工作于各个技术领域的工程师和技术人员的自学读物。.

组编这套国外知名大学原版系列教材是一个尝试。不管是书目确定的合理性,教材选择的恰当性,还是评论看法的确切性,都有待于通过使用和实践来检验。感谢使用本系列教材的广大教师和学生的支持。期望广大读者提出意见和建议。

Discrete-time Signal Processing, second edition

影印版推荐意见

随着计算机技术、微电子技术和信息学科的飞速发展,数字信号处理的理论与应用在过去的四十年间得到长足的进展,已经发展成为一门新兴的高新技术学科。目前国内外大学的电子、通信、计算机等专业都普遍开设数字信号处理课程,一大批教材随之诞生。

美国麻省理工学院教授奥本海姆(A. V. Oppenheim)是信号处理领域中的佼佼者,他的几本信号和信号处理著作已成为享誉世界的经典力作。本书的第一代著作《数字信号处理》(1975年版)被认为是这一领域的经典教材。他系统地总结了数字信号处理基本理论、基本概念和规律,全书概念清晰,逻辑性强,推理严密,是一本极好的数字信号处理教材。我国一些院校在70年代末及80年代初相继开设数字信号处理课程,在此期间奥本海姆本人来我国到清华及其他兄弟院校讲学;他的讲课录像带曾在国内广为流传。此后我国陆续出版的数字信号处理教材大多参考奥本海姆著作的体系写成。

随着时代的发展,作为新兴学科,与数字信号处理有关的新理论、新概念、新算法、新应用不断涌现。时隔二十年后,为适应形势的发展,对教学内容有新的需求。1989年,奥本海姆出版了《离散时间信号处理》一书(第一版),该书在原书的基础上做了大幅度调整,着重于基础内容的加强和更新,对一些新发展的基础性专题论述更为详细,基本概念阐述更加清晰,内容安排更为合理,同时大大增加例题和习题。

1999 年,奥本海姆又把《离散时间信号处理》作了必要的修改,以第二版形式出现。第二版保持了第一版的风格,但考虑到 20 世纪末,新世纪开始,"数字信号处理"课程已成为大学本科阶段普遍开设的课程,所以在内容处理上,作者认为: 作为教材,不是要去"涵盖"学科发展的各个方面,而是要揭示一个学科的基础知识、基本理论、基本方法,提升学生和工程师们的解决问题的能力。作者把一些不是本科阶段必需的内容予以剔除,例如把倒谱和同态信号处理的内容从正文中删除,放到习题中; 同时考虑到信号处理中对多采样率信号处理在过采样的 A/D 和 D/A 转换和数字滤波器实现中的重要性而扩展了这部分内容; 再如,把在数据压缩中起重要作用的余弦变换引入教材中。这些内容上的增减使教材在阐述经典内容的同时,不断引入新的概念,反映了教材的合理性、科学性和先进性。

第二版的另一大特点是实践性。"数字信号处理"是实践性很强的课程,奥本海姆保持了近几年来写书的风格,把丰富的习题留给读者。习题有一部分来自文献,有相当的难度。本版教材把习题由第一版的 371 道扩大到 588 道题,例题由 113 道扩大到 137 道;同时把习题分成三个层次:基本题(附答案),深入题和扩充题。这些习题对深入理解教材内容、巩固基本概念、培养分析问题和解决问题的能力是极有帮助的。学习数字信号处理离不开上机练习,本教材虽没有具体上机指南的内容,但正如书中前言所述,"本书的很多习题都可以

很容易变成用计算机来解的练习题",而且指出了具体的途径。

总之,本书是奥本海姆在数字信号处理教材方面的一本力作,其概念清晰、阐述严谨、内容安排合理、例题习题丰富,是一本很好的基础教材。虽然上述几种教材都已有相应的中译本,但要更好理解作者思路、确切弄清概念,有必要了解原著,使中英文版本二者相辅相成,体验其精髓并付诸实践。

郑君里 应启珩 清华大学电子工程系教授 2004年10月

To Phyllis, Justine and Jason

To Dorothy, Bill, Tricia, Ken and Kate and in memory of John

To Susan

PREFACE

This text is a second generation descendent of our text, *Digital Signal Processing*, which was published in 1975. At that time, the technical field of digital signal processing was in its infancy, but certain basic principles had emerged and could be organized into a coherent presentation. Although courses existed at a few schools, they were almost exclusively at the graduate level. The original text was designed for such courses.

By 1985, the pace of research and integrated circuit technology made it clear that digital signal processing would realize the potential that had been evident in the 1970s. The burgeoning importance of DSP clearly justified a revision and updating of the original text. However, in organizing that revision, it was clear that so many changes had occurred that it was most appropriate to develop a new textbook, strongly based on our original text, while keeping the original text in print. We titled the new book *Discrete-Time Signal Processing* to emphasize that most of the theory and design techniques discussed in the text apply to discrete-time systems in general.

By the time *Discrete-Time Signal Processing* was published in 1989, the basic principles of DSP were commonly taught at the undergraduate level, sometimes even as part of a first course on linear systems, or at a somewhat more advanced level in third-year, fourth-year, or beginning graduate subjects. Therefore, it was appropriate to expand considerably the treatment of such topics as linear systems, sampling, multirate signal processing, applications, and spectral analysis. In addition, more examples were included to emphasize and illustrate important concepts. We also removed and condensed some topics that time had shown were not fundamental to the understanding of discrete-time signal processing. Consistent with the importance that we placed on well constructed examples and homework problems, the new text contained more than 400 problems.

In the decade or so since Discrete-Time Signal Processing was published, some important new concepts have been developed, the capability of digital integrated circuits has grown exponentially, and an increasing number of applications have emerged. However, the underlying basics and fundamentals remain largely the same albeit with a refinement of emphasis, understanding and pedagogy. Consequently when we looked at what was needed to keep Discrete-Time Signal Processing up-to-date as a textbook emphasizing the fundamentals of DSP, we found that the changes needed were far less drastic than before. In planning this current revision we were guided by the principle that the main objective of a fundamental textbook is to uncover a subject rather than to cover it. Consequently, our goal in this current revision is to make the subject of discrete-time signal processing even more accessible to students and practicing engineers, without compromising on coverage of what we consider to be the essential concepts that define the field. Toward this end we have considerably expanded our coverage of multi-rate signal processing due to its importance in oversampled A-to-D and D-to-A conversion and digital filter implementation. We have added a discussion of the cosine transform, which plays a central role in data compression standards. We have also removed some material that we judged to be of lesser importance in the present xx Preface

context, or more appropriate for advanced textbooks and upper level graduate courses. Many of the concepts that were removed from the text (such as basic results on the cepstrum) have reappeared in some of the new homework problems.

A major part of our emphasis in this revision has been directed toward the homework problems and examples. We have significantly increased the number of examples which are important in illustrating and understanding the basic concepts, and we have increased the number of homework problems. Furthermore, the homework problems have been reorganized according to their level of difficulty and sophistication, and answers are provided to a selected set of problems. The instructor's manual available from the publisher contains updated solutions for all of the problems in the book. These solutions were prepared by Li Lee and Maya Said of MIT and Jordan Rosenthal and Greg Slabaugh of Georgia Tech. This manual also contains some suggested exam problems based on our courses at MIT, Georgia Tech and the University of Massachusetts Dartmouth.

As in the earlier texts, it is assumed that the reader has a background of advanced calculus, along with a good understanding of the elements of complex numbers and variables. In this edition, we have refrained from the use of complex contour integration in order to make the discussion accessible to a wider audience. An exposure to linear system theory for continuous-time signals, including Laplace and Fourier transforms, as taught in most undergraduate electrical and mechanical engineering curricula is still a basic prerequisite. With this background, the book is self-contained. In particular, no prior experience with discrete-time signals, z-transforms, discrete Fourier transforms, and the like is assumed. In later sections of some chapters, some topics such as quantization noise are included that assume a basic background in stochastic signals. A brief review of the background for these sections is included in Chapter 2 and in Appendix A.

It has become common in many signal processing courses to include exercises to be done on a computer, and many of the homework problems in this book are easily turned into problems to be solved with the aid of a computer. As in the first edition, we have purposely avoided providing special software to implement algorithms described in this book, for a variety of reasons. Foremost among them is that there are a variety of inexpensive signal processing software packages readily available for demonstrating and implementing signal processing on any of the popular personal computers and workstations. These packages are well documented and have excellent technical support, and many of them have excellent user interfaces that make them easily accessible to students. Furthermore, they are in a constant state of evolution, which strongly suggests that available software for classroom use should be constantly reviewed and updated. We share the enthusiasm of many for MATLAB, which an increasing number of students are learning at early stages of their education. However, we continue to prefer a presentation that utilizes the power of computational tools such as MATLAB to create examples and illustrations of the theory and fundamentals for use in the text, but does not let issues of programming syntax and functionality of the software environment detract from the emphasis on the concepts and the way that they are used. We firmly believe that there is enormous value in hands-on experience. Indeed, software tools such as MATLAB allow students to implement sophisticated signal processing systems on their own personal computers, and we feel that there is great benefit to this once the student is confident of the fundamentals and is capable of sorting out programming mistakes from conceptual errors. For this reason, the instructor's manual contains a secPreface xxi

tion of suggestions for assignments in the inexpensive texts Computer-Based Exercises for Signal Processing Using Matlab 5 by McClellan, et al., and Computer Explorations in Signals and Systems Using Matlab by Buck, Daniel and Singer, both of which are also available from Prentice-Hall, Inc. These suggestions link projects in these computer exercise books to specific sections, examples and problems in this textbook. This will allow instructors to design computer assignments which are related to the material and examples they have covered in class, and to link these computer assignments to traditional analytic homework problems to reinforce the concepts demonstrated there.

The material in this book is organized in a way that provides considerable flexibility in its use at both the undergraduate and graduate level. A typical one-semester undergraduate elective might cover in depth Chapter 2, Sections 2.0-2.9; Chapter 3; Chapter 4, Sections 4.0-4.6; Chapter 5, Sections 5.0-5.3; Chapter 6, Sections 6.0-6.5; Chapter 7, Sections 7.0-7.3 and a brief overview of Sections 7.4-7.5. If students have studied discrete-time signals and systems in a general signals and systems course, it would be possible to move more quickly through the material of Chapters 2, 3, and 4, thus freeing time for covering Chapter 8. A first-year graduate course could augment the above topics with the remaining topics in Chapter 5, a discussion of multirate signal processing (Section 4.7) an exposure to some of the quantization issues introduced in Section 4.8 and perhaps an introduction to noise shaping in A/D and D/A converters as discussed in Section 4.9. A first-year graduate course should also include exposure to some of the quantization issues addressed in Sections 6.6-6.9, to a discussion of optimal FIR filters as incorporated in Sections 7.4 and 7.5, and a thorough treatment of the discrete Fourier transform (Chapter 8) and its computation using the FFT (Chapter 9). The discussion of the DFT can be effectively augmented with many of the examples in Chapter 10. In a two-semester graduate course, the entire text together with a number of additional advanced topics can be covered.

In Chapter 2, we introduce the basic class of discrete-time signals and systems and define basic system properties such as linearity, time invariance, stability, and causality. The primary focus of the book is on linear time-invariant systems because of the rich set of tools available for designing and analyzing this class of systems. In particular, in Chapter 2 we develop the time-domain representation of linear time-invariant systems through the convolution sum and introduce the class of linear time-invariant systems represented by linear constant-coefficient difference equations. In Chapter 6, we develop this class of systems in considerably more detail. Also in Chapter 2 we introduce the frequency-domain representation of signals and systems through the Fourier transform. The primary focus in Chapter 2 is on the representation of sequences in terms of the Fourier transform, i.e., as a linear combination of complex exponentials, and the development of the basic properties of the Fourier transform.

In Chapter 3, we develop the z-transform as a generalization of the Fourier transform. This chapter focuses on developing the basic theorems and properties of the z-transform and the development of the partial fraction expansion method for the inverse transform operation. In Chapter 5, the results developed in Chapters 3 and 4 are used extensively in a detailed discussion of the representation and analysis of linear time-invariant systems.

In Chapter 4, we carry out a detailed discussion of the relationship between continuous-time and discrete-time signals when the discrete-time signals are obtained through periodic sampling of continuous-time signals. This includes a development of xxii Preface

the Nyquist sampling theorem. In addition, we discuss upsampling and downsampling of discrete-time signals, as used, for example, in multirate signal processing systems and for sampling rate conversion. The chapter concludes with a discussion of some of the practical issues encountered in conversion from continuous time to discrete time including prefiltering to avoid aliasing, modeling the effects of amplitude quantization when the discrete-time signals are represented digitally, and the use of oversampling in simplifying the A-to-D and D-to-A conversion processes.

In Chapter 5 we apply the concepts developed in the previous chapters to a detailed study of the properties of linear time-invariant systems. We define the class of ideal, frequency-selective filters and develop the system function and pole-zero representation for systems described by linear constant-coefficient difference equations, a class of systems whose implementation is considered in detail in Chapter 6. Also in Chapter 5, we define and discuss group delay, phase response and phase distortion, and the relationships between the magnitude response and the phase response of systems, including a discussion of minimum-phase, allpass, and generalized linear phase systems.

In Chapter 6, we focus specifically on systems described by linear constant-coefficient difference equations and develop their representation in terms of block diagrams and linear signal flow graphs. Much of this chapter is concerned with developing a variety of the important system structures and comparing some of their properties. The importance of this discussion and the variety of filter structures relate to the fact that in a practical implementation of a discrete-time system, the effects of coefficient inaccuracies and arithmetic error can be very dependent on the specific structure used. While these basic issues are similar whether the technology used for implementation is digital or discrete-time analog, we illustrate them in this chapter in the context of a digital implementation through a discussion of the effects of coefficient quantization and arithmetic roundoff noise for digital filters.

While Chapter 6 is concerned with the representation and implementation of linear constant-coefficient difference equations, Chapter 7 is a discussion of the procedures for obtaining the coefficients of this class of difference equations to approximate a desired system response. The design techniques separate into those used for infinite impulse response (IIR) filters and those used for finite impulse response (FIR) filters.

In continuous-time linear system theory, the Fourier transform is primarily an analytical tool for representing signals and systems. In contrast, in the discrete-time case, many signal processing systems and algorithms involve the explicit computation of the Fourier transform. While the Fourier transform itself cannot be computed, a sampled version of it, the discrete Fourier transform (DFT), can be computed, and for finite-length signals the DFT is a complete Fourier representation of the signal. In Chapter 8, the discrete Fourier transform is introduced and its properties and relationship to the discrete-time Fourier transform are developed in detail. In this chapter we also provide an introduction to the discrete cosine transform which is playing an increasingly important role in many applications including audio and video compression. In Chapter 9, the rich and important variety of algorithms for computing or generating the discrete Fourier transform is introduced and discussed, including the Goertzel algorithm, the fast Fourier transform (FFT) algorithms, and the chirp transform.

With the background developed in the earlier chapters and particularly Chapters 2, 3, 5, and 8, we focus in Chapter 10 on Fourier analysis of signals using the discrete Fourier

Preface xxiii

transform. Without a careful understanding of the issues involved and the relationship between the DFT and the Fourier transform, using the DFT for practical signal analysis can often lead to confusions and misinterpretations. We address a number of these issues in Chapter 10. We also consider in some detail the Fourier analysis of signals with time-varying characteristics by means of the time-dependent Fourier transform.

In Chapter 11, we introduce the discrete Hilbert transform. This transform arises in a variety of practical applications, including inverse filtering, complex representations for real bandpass signals, single-sideband modulation techniques, and many others.

With this edition we thank and welcome Professor John Buck. John has been a long time contributor to this book through his teaching of the subject while a student at MIT and more recently as a member of the faculty at the University of Massachusetts Dartmouth. In this edition he has taken the major responsibility for a total reworking and reorganization of the homework problems and many of the examples in the book. His insight and dedication to the task are obvious in the final result.

Alan V. Oppenheim Ronald W. Schafer

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