国外电子与通信教材系列

数字信号处理实践方法

（第二版）

Digital Signal Processing

A Practical Approach, Second Edition

英文版

[英] Emmanuel C. Ifeachor
Barrie W. Jervis 著

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内容简介

本书根据实际工程应用和具体实例，详细介绍了数字信号处理（DSP）领域内的基本概念和相关技术。全书共分为14章，首先讲解了DSP的基本概念及其应用，并从实际的例子出发，阐述了DSP的一些基本内容，如信号的抽样、量化及其在实时DSP上的内涵。然后，作者介绍了离散变换（DFT和FFT）、离散时间信号与系统分析的工具（z变换）及DSP的基本运算（相关和卷积），并分析了数字滤波器设计的实际问题。本书还介绍了多抽样率数字信号处理、自适应数字滤波器、谱估计及其分析等现代数字信号处理理论，最后讨论了通用和专用数字信号处理器、定点DSP系统有限字长效应分析及DSP应用和设计实例。另外，书中还提供了有关范例和实验的MATLAB实现方法。

本书可作为通信与电子信息类专业高年级本科生和研究生的教材或教学参考书，而且对于相关学科的工程技术人员也具有很好的参考价值。

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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。

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教材出版委员会

Preface

Purpose of this book

In the last five years, digital signal processing (DSP) has continued to have a major and increasing impact in many key areas of technology, including telecommunication, digital television and media, biomedicine, digital audio and instrumentation. DSP is now at the core of many new and emerging digital products and applications in the information society (e.g. digital cellular phones, digital cameras and TV, and digital audio systems). The need and expectations for electronic, computer and communication engineers to be competent in DSP have grown even stronger since the first edition. DSP is now a core subject in most electronic/computer/communication engineering curricula.

This second edition of the book has been modernized by including additional topics of increasing importance, by providing MATLAB-based problems, and by offering a companion handbook and a home page on the web. These additions have been made in response to software developments, the wider availability of information technology, developments in the teaching of signal processing, and reader demand. Universities are increasingly making use of web-based materials and signal processing software tools such as MATLAB. We have consequently found a demand amongst our readers for MATLAB-based material. This high-level language permits sophisticated signal processing and the immediate display of results with relatively few commands. It is possible to have fun in developing signal processing and the solutions to problems without the distraction of having to produce detailed programs. We believe that the MATLAB examples and exercises in the book will enhance the learning experience of the student and increase the teaching resource available to the instructor.

As in the first edition, the second edition aims to bridge the gap between theory and practice. Thus, we have retained the main features of the book, namely coverage of modern topics and the provision of practical examples and applications. As in the first edition, we have mixed practical examples and systems with theory, to keep students' interest and motivation high in order to enhance learning. Many of the chapters have been extensively revised, to bring the materials up to date and to improve clarity. End-of-chapter problems have been extended to test, reinforce and extend understanding.

In revising the book, we have drawn from our experience and feedback from readers, worldwide, over the last eight years since the first edition was published.

The new topics introduced include oversampling and bandpass sampling techniques in analog/digital conversions to exploit the advantages that DSP offers; wavelet transforms used for time–frequency representation and resolution of signals; blind signal deconvolution for identifying input signals from the output of an unknown system; parametric spectrum estimation for greater resolution applicable to shorter signals and with fewer pitfalls; architectures of new DSP processors and practical schemes for round-off noise reduction in fixed-point DSP systems; and computer-based multi-choice questions to aid revision. Throughout the book, MATLAB-based examples and exercises are provided.

This book was born out of our experience in teaching practically oriented courses in digital signal processing to undergraduate students at the University of Plymouth and the Sheffield Hallam University, and to application engineers in industry for many years. It appeared to us that many of the available textbooks were either too elementary or too theoretical to be of practical use for undergraduates or application engineers in industry. As most readers will know from experience, the gap between learning the fundamentals in any subject and actually applying them is quite wide. We therefore decided to write this book which we believe undergraduates will understand and appreciate and which will equip them to undertake practical digital signal processing assignments and projects. We also believe that higher degree students and practising engineers and scientists will find this text most useful.

Our own research work over the last two and a half decades in applied DSP has also inspired the contents, by identifying practical issues for discussion and presentation to bridge the gap between theoretical concepts and practical implementation, and by suggesting application examples, case studies, and problems.

The great interest and developments in DSP in both industry and academia are likely to continue for the foreseeable future. The availability of numerous digital signal processors is a testimony to the commercial importance of DSP. Its major attraction lies in the ability to achieve guaranteed accuracy and perfect reproducibility, and in its inherent flexibility compared with analog signal processing. In industry, many engineers still lack the necessary knowledge and expertise in DSP to utilize the immense potential of the very powerful digital signal processors now available off the shelf. This book provides insight and practical guidance to enable engineers to design and develop practical DSP systems using these devices.

In academia, DSP is generally regarded as one of the more mathematical topics in the electrical engineering curriculum, and based on our experiences of teaching we have reduced the mathematical content to what we consider useful, essential, and interesting; we have also emphasized points of difficulty. Our experiences indicate that students learn best if they are aware of the practical relevance of a subject, and while more theoretical texts are essential for completeness and reference as the student matures in the subject, we believe in producing graduates equipped also with practical knowledge and skills. This book was written with these considerations in mind.

The book is not a comprehensive text on DSP, but it covers most aspects of the subject found in undergraduate electrical, electronic or communication engineering degree courses. A number of DSP techniques which are of particular relevance to industry are also covered and in the last few years are beginning to find their way into undergraduate curricula. These include techniques such as adaptive filtering and multirate processing.

The emphasis throughout the book is on the practical aspects of DSP. An important feature of the second edition is the inclusion of MATLAB examples and exercises for signal processing, analysis, design and exploration in a time-efficient manner. The reader is encouraged to carry out the MATLAB exercises to gain further insight into DSP. We have also provided the C language DSP software tool from the first edition, after minor revisions, as this has proved popular.

MATLAB is now widely used as a generic tool in industry and academia and requires less programming skills than C. It has good graphics and display facilities and provides a good environment for developing DSP. We believe that MATLAB is a useful tool for students to become familiar with, and competence in it is a valuable transferable skill to acquire. All the MATLAB m-files referred to in the book are available electronically via the web. These include MATLAB m-files which may be used to perform similar tasks as several C-language programs in the first edition. In addition, the m-files (as well as the C-language programs from the first edition) are also available on the CD in the companion handbook (see below for details of how to obtain copies of these).

Main features of the book

- Provides an understanding of the fundamentals, implementation and applications of DSP techniques from a practical point of view.
- Clear and easy to read, with mathematical contents reduced to that which is necessary for comprehension.
- DSP techniques and concepts are illustrated with practically oriented, fully worked, real-world examples designed to provide insight into DSP.
- Provides practical guidance to enable readers to design and develop actual DSP systems. Complete design examples and practical implementation details are given, including assembly language programs for DSP processors.
- MATLAB examples and problems to provide hands-on experience.
- Provides C language implementation of many DSP algorithms and functions, including programs for:
  - digital FIR and IIR filter design,
  - finite wordlength effects analysis of user-designed fixed-point IIR filters,
  - converting from cascade to parallel realization structures,
  - correlation computation,
  - discrete and fast Fourier transform algorithms,
  - inverse z-transformation,

- frequency response estimation, and
- multirate processing systems design.

- PC-based MATLAB m-files are available electronically on the web (the C programs from the first edition are available on the CD that comes with the companion handbook – see the section 'Website, CD and companion handbook for this book' in this preface for details).
- Contains many end-of-chapter problems and provides multiple-choice questions to assist with revision.
- Uses realistic examples to illustrate important concepts and to reinforce the knowledge gained.

The intended audience

The book is aimed at engineering, science and computer science students, and application engineers and scientists in industry who wish to gain a working knowledge of DSP. In particular, final year students studying for a degree in electronics, electrical or communication engineering will find the book valuable for both taught courses as well as their project work, as increasingly a greater proportion of student project work involves aspects of DSP. Postgraduates studying for a master's degree or PhD in the above subjects will also find the book useful.

Undergraduate students will find the fundamental topics very attractive and, we believe, the book will be a valuable source of information both throughout their course as well as when they go into industry.

Large commercial or government organizations who undertake their own internal DSP short courses could base them on the book. We believe the book will serve as a good teaching text as well as a valuable self-learning text for undergraduate, graduate and application engineers.

Contents and organization

Chapter 1 contains an overview of DSP and its applications to make the reader aware of the meaning of DSP and its importance. In Chapter 2 we present, from a practical point of view using real-world examples, many fundamental topics which form the cornerstone of DSP, such as sampling and quantization of signals and their implications in real-time DSP. New features include important topics such as oversampling techniques in AD/DA conversion, sampling of bandpass signals, and uniform and non-uniform quantization. Discrete-time signals and systems are introduced in this chapter, and discussed further in Chapter 4.

Discrete transforms, particularly the discrete and fast Fourier transforms (FFT), provide important mathematical tools in DSP as well as relating the time and frequency domains. They are introduced and described in Chapter 3 with a discussion of some applications to put them in context. The derivation of the discrete Fourier transform (DFT) from the Fourier transform and the exponential Fourier series provides a logical justification for the DFT which does not require coverage of the discrete Fourier series which would unnecessarily increase the length of the book (and the amount of work for the student!). The discussion has also been restricted to the description and implementation of the transforms. In particular, the topic of windowing has not been included in this chapter but is more appropriately discussed in detail in Chapter 11 on spectrum analysis. As an important application of the discrete cosine transform the JPEG standard for image compression is described. The wavelet transform has been growing increasingly popular for a variety of applications because of its applicability to non-stationary signals and its ability to resolve signals in both frequency and time. An introduction to the topic has therefore been included. Applications to multi-resolution analysis and singularity detection for the denoising of signals are described.

In Chapter 4 the basics of discrete-time signals and systems are discussed. Important aspects of the z-transform, an invaluable tool for representing and analyzing discrete-time signals and systems, are discussed. Many applications of the z-transform are highlighted, for example its use in the design, analysis and computation of the frequency response of discrete-time signals and systems. As in the rest of the book, the concepts as well as applications of the z-transform are illustrated with fully worked examples.

Correlation and convolution are fundamental and closely related topics in DSP and are covered in depth in Chapter 5. The authors consider an awareness of all the contents of this chapter to be essential for DSP, but after a preliminary scanning of the contents the reader may well be advised to build up his or her detailed knowledge by progressing through the chapter in stages. The contents might well be spread over several years of an undergraduate course. In this second edition the additional topics of system identification, deconvolution and blind deconvolution have been included. Blind deconvolution is especially interesting, since by exploiting information maximization it is possible to determine an unknown input signal measured at the output of a system of unknown impulse response.

Chapters 6, 7 and 8 include detailed practical discussions of digital filter design, one of the most important topics in DSP, being at the core of most DSP systems. Digital filter design is a vast topic and those new to it can find this somewhat overwhelming. Chapter 6 provides a general framework for filter design. A simple but general step-by-step guide for designing digital filters is given.

Techniques for designing FIR (finite impulse response) filters from specifications through to filter implementations are discussed in Chapter 7. Several fully worked examples are given throughout the chapter to consolidate the important concepts. In this edition, additional topics covered include automatic design of frequency FIR filters. A complete filter design example is included to show how all the stages of filter design fit together.

IIR (infinite impulse response) filter design is discussed in detail in Chapter 8, based on the simple step-by-step guide. This chapter has been substantially reorganized and extended. In particular, the sections on coefficient calculation have been reorganized for clarity and new materials added to cover important topics in IIR filter design, in response to feedback from readers. Additionally, fully worked examples have been included to help the reader to design IIR filters from specifications through to implementation. Design examples using MATLAB as well as C language software are given.

We have reduced the overall material on IIR filter design, by moving the material on finite wordlength effects to Chapter 13. Thus, in response to readers' feedback the materials in Chapters 1–8 contain essential materials for most DSP courses. The more advanced DSP topics now appear in later chapters. Detailed treatment of finite wordlength effects in DSP algorithms now appear logically together in Chapter 13.

Multirate processing techniques allow data to be processed at more than one sampling rate and have made possible such novel applications as single-bit ADCs and DACs (digital-to-analog converters), and oversampled digital filtering, which are exploited in a number of modern digital systems, including for example the familiar compact disc player. In Chapter 9, the basic concepts of multirate processing are explained, illustrated with fully worked examples and by the design of actual multirate systems. The materials in this chapter have been extended to include polyphase. More design examples and applications have been integrated into the theory to illustrate both the principles and design issues in practical multirate systems.

In Chapter 10, key aspects of adaptive filters are described, based on the LMS (least-mean-squares) and RLS (recursive least-square) algorithms which are two of the most widely used algorithms in adaptive signal processing. The treatment is practical with only the essential theory included in the main text.

In Chapter 11, the important topic of spectrum estimation and analysis, used to describe and study signals in the frequency domain, is described. With the introduction of software packages for parametric spectrum estimation it seemed appropriate to provide a detailed introduction to these methods. Provided the signals are accurately represented by models of the correct order, parametric spectrum estimation is applicable to shorter signal lengths and provides spectrum estimates of improved resolution compared with non-parametric methods. An application of autoregressive spectrum estimation of evoked response signals in electroencephalogram signals is used to illustrate the method. Readers who are particularly interested in spectral analysis should study both Chapters 11 and 3 as Chapter 11 draws on explanations and worked examples given in Chapter 3. Those who master the contents of these chapters will be well placed to become competent in the analysis of signals in the frequency domain.

In the last decade and a half, tremendous progress has been made in DSP hardware, and this has led to the wide availability of low cost digital signal processors. For a successful application of DSP using these processors, it is necessary to appreciate the underlying concepts of DSP hardware and software. Chapter 12 discusses the key issues underlying general- and special-purpose processors for DSP, the impact of DSP algorithms on the hardware and software architectures of these processors, and the architectural requirements for efficient execution of DSP functions. The materials in this chapter have been brought up to date. In particular, we have discussed new DSP architectures such as very long instruction word and super scalar, and new fixed and floating point DSP processors (including Texas Instruments fixed point processors, e.g. TMS320C54 and TMS320C62, Motorola fixed point processors DSP56300, and Analog Devices TigerSHARC IS0001).

In Chapter 13, a detailed analysis of finite wordlength effects in modern fixed point DSP systems is presented. Solutions are provided, where appropriate, to the degrading effects of using fixed precision arithmetic.

Chapter 14 is new (although some materials from the first edition are retained) and serves as a teaching and learning resource for the instructor and the student. The chapter includes a description of low-cost DSP boards for implementation of DSP algorithms and a description of a number of real-world applications in the form of case studies. Other features include computer-based, multiple-choice questions which cover key aspects of the topics covered in earlier chapters, and are valuable for revision and for assessing large classes. Complete laboratory exercises are described and case study/project ideas provided.

How to use the book

A useful approach for undergraduate teaching will be to cover the materials in Chapters 1 and 2, to provide the understanding of fundamental topics such as the sampling theorem and discrete-time signals and systems, and to establish the benefits and applications of DSP. Then discrete transforms should be introduced, starting with the DFT and FFT (Chapter 3), and the z-transform (Chapter 4). Aspects of Chapters 11 and 5 may be used to illustrate the application of the DFT and FFT. After an introduction to correlation processing using a selection of materials from Chapter 5, a detailed treatment of digital filters should be undertaken.

In our experience students learn more when they are given realistic assignments to carry out. To this end we would encourage substantial assignments on, for example, filter design, the inverse z-transform, the DFT and FFT. Laboratory work should also be designed to demonstrate and reinforce the techniques taught. It is important that students actually participate as well as attend lectures.

For final-year undergraduates and postgraduate students the approach could be the same but the pace will be more brisk, and the more specialist topics of multirate processing and adaptive filters will also be included.

Website, CD and companion handbook for this book

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Electronic copies of all the MATLAB m-files can be downloaded from the companion web site for this book at:

www.booksites.net/ifeachor

These include a number of MATLAB m-files which the reader can use to perform similar tasks as they would with several C-language programs in the first edition. The MATLAB m-files, C programs and assembly language codes are also available on the CD that comes with the companion handbook. The C-language programs, taken from the first edition (after minor revision), are available in both executable form and as source codes. A C compiler is required to run the source codes, but not to run the executable codes. The programs were written in standard ANSI C under Borland Turbo C version 2.0. The companion handbook *A Practical Guide for MATLAB and C Language Implementations of DSP Algorithms*, published by Pearson, together with the CD, can be purchased separately. The handbook also contains many illustrative examples of the use of the MATLAB m-files and C programs in the main book. You will find an order form at the back of this book.

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