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# Digital Signal Processing
## A Computer-Based Approach
Second Edition

# 数字信号处理 —— 基于计算机的方法
第 2 版

Sanjit K.Mitra

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# DIGITAL SIGNAL PROCESSING

A Computer-Based Approach

**Second Edition**

Sanjit K. Mitra
Department of Electrical and Computer Engineering
University of California, Santa Barbara

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# 国际知名大学原版教材

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# 出 版 说 明

郑大钟

清华大学信息技术科学与技术学院

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"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。

# Digital Signal Processing—A Computer-Based Approach
## (第2版)
# 影 印 版 序

清华大学出版社为配合清华大学创建世界一流大学的规划,决定批量引进国外著名大学最新出版的高水平的原版教材。这不但对清华大学,而且对全国所有高校的学科建设及人才培养都有着重要的意义。现在出版的《Digital Signal Processing—A Computer-Based Approach》一书即是最新引进的一种。这是一本值得推荐的好教材。

该书的中文名字可以译为《数字信号处理——基于计算机的方法》。它由国际著名的McGraw-Hill出版社于2001年最新出版。该书是美国加利福尼亚大学圣·巴巴拉分校的教材,作者是Sanjit K. Mitra教授。

Mitra教授是国际上著名的信号处理专家。他在加利福尼亚大学伯克利分校获得硕士和博士学位,先后在康奈尔大学、AT&T贝尔实验室、加利福尼亚大学戴维斯分校、圣·巴巴拉分校任教和工作。他曾任圣·巴巴拉分校电气与计算机工程系主任、IEEE电路与系统学会的主席,IEEE、AAAS和SPIE学会的Fellow,多个国际著名杂志的编委,获得过多项企业和学术界的奖励。他发表学术论文500多篇,出版了11本著作。本书在两年前出版了第1版,现在影印出版的是该书的第2版。

在介绍本书的特点之前,有必要先谈一下本书所讨论的主题——"数字信号处理"。

随着计算机和信息学科的飞速发展,数字信号处理(DSP)的理论与应用在过去的三十多年中获得了飞越式的发展,并已成长为一门极其重要的高新技术学科。简单地说,数字信号处理是利用计算机或专用处理设备,以数值计算的方法对信号进行采集、变换、综合、估值与识别等加工处理,借以达到提取信息和便于应用的目的。采用数字的方法处理信号比传统的模拟处理方法有着无法比拟的优点。

数字信号处理的特点是其理论性和实践性都很强。理论性强,是指在综合应用众多的数学、电路理论、信号、系统和信息等领域知识的基础上,发展并形成了自己丰富的理论体系。实践性强,一方面指的是该学科的理论目前已成为众多新兴学科(如现代通信理论、自动控制、模式识别、故障诊断等)的重要理论基础;另一方面,是指数字信号处理的应用极其广泛,如在通信、控制、仪器、仪表、电力系统、电力电子、生物医学工程、机械及力学等领域。可以说,凡是和"信号"有关的学科领域都要用到DSP。

目前,以数字信号处理器(DSP)为代表的高新技术产业正在世界范围内蓬勃兴起。另外,美国MathWork公司推出的Matlab科技应用软件现正在风靡全世界,其中的信号处理(Signal Processing)工具箱,以及和信号处理有关的工具箱(如小波、高阶谱分析等)更是学习和应用DSP的有力工具。DSP芯片的飞速发展及Matlab信号处理软件的不断完善又有力地促进了数字信号处理理论的发展,并为其开拓了更加广阔的应用空间。

成功地将数字信号处理理论和DSP芯片用于实际,需要大批既掌握DSP的理论,同时又具有DSP硬件知识的高水平人才。现在,国内外重点大学的电气、通讯与计算机类的大

部分院、系都为本科生或研究生开设了“数字信号处理”课程，有的还开设了“现代信号处理”课程。

凡是从事过数字信号处理教学的老师，或是学习过该课程的同学都有一个共同的体会，即由于数字信号处理的理论性特别强、内容又特别多，所以非常希望能有一本或几本高水平的教材。本人在阅读了 Mitra 教授的《Digital Signal Processing—A Computer-Based Approach》一书后，感到该书确实是一本值得推荐的好教材。该书的特点是：

1. 本书的选材以数字信号处理的基础内容为主，同时也给出了现代信号处理的部分内容。书中以主要篇幅讨论了离散信号和离散系统的基本概念及其时域分析、变换域分析、数字滤波器的结构与设计、有限字长分析及随机信号的基本概念等。这是数字信号处理中的经典内容，也是进一步学习和掌握更多信号处理理论的基础。此外，本书用一章讨论子多抽样率信号处理问题，其中的滤波器组(Filter Bank)是近十年来非常活跃的内容，广泛应用于子带编码和小波变换。本书最后一章介绍了数字信号处理的应用，并讨论了现代谱估计的部分内容。所以，从本书的选材看，它非常适合电类专业高年级本科生用作教材，当然，也可作为研究生的参考用书和工程技术人员的自学用书。

2. 本书说理清楚，疑难处讨论得比较详尽。如书中关于 FT、DTFT 存在性的讨论，关于各种离散系统性质的讨论以及滤波器设计的讨论等都很有特色。这样，当学生预习或是自学书中的内容时，一般不会遇到困难。

3. 本书给出了大量的例子来讨论所要介绍的理论问题，这些例子共有 231 个，再加上说理清楚的特点，就使本书更具有可读性。这一方面有利于读者掌握书中的理论，另一方面有利于读者知道如何把这些理论应用于实际。

4. 本书另外一个重要的特点是把传统理论的讨论和 Matlab 相结合。前面已指出，Matlab 是当前最优秀的科技应用软件，其中信号处理的内容极其丰富，包含了绝大部分信号处理算法的程序。二者的结合可以帮助读者在学习了信号处理理论的同时也学习了 Matlab，并可以互相促进。书中给出了 90 个 Matlab 的子程序，基本上覆盖了本书要讨论的内容。这一特点也正是本书书名“基于计算机的方法”的内涵。

5. 本书的习题也很有特色，一是多，计 684 个；二是习题的质量相当高，有利于培养读者的思考能力和创新能力；三是给出了 186 个 Matlab 的练习，这非常有利于读者在计算机上去实践信号处理的众多理论和算法，同时也为把这些理论和算法用于工程实际打下了很好的基础。

总之，由这本书可以看出，Mitra 教授在数字信号处理教学方面有着丰富的经验；也可以看出，他在这本书的编写中确实下了很大的功夫，并把自己的教学经验反映在书中。所以，作为在国内长期从事数字信号处理教学的教师，我向国内的读者推荐 Mitra 教授的这本好教材。

胡广书　教授

清华大学电机工程与应用电子技术系

2001 年 5 月

# About the Author

**Sanjit K. Mitra** received his M.S. and Ph.D. in electrical engineering from the University of California, Berkeley, and an Honorary Doctorate of Technology from Tampere University of Technology in Finland. After holding the position of assistant professor at Cornell University until 1965 and working at AT&T Bell Laboratories, Holmdel, New Jersey, until 1967, he joined the faculty of the University of California at Davis. Dr. Mitra then transferred to the Santa Barbara campus in 1977, where he served as department chairman from 1979 to 1982 and is now a Professor of Electrical and Computer Engineering. Dr. Mitra has published more than 500 journal and conference papers, and 11 books, and holds 5 patents. He served as President of the IEEE Circuits and Systems Society in 1986 and is currently a member of the editorial boards for four journals: *Multidimensional Systems and Signal Processing; Signal Processing; Journal of the Franklin Institute*; and *Automatika*. Dr. Mitra has received many distinguished industry and academic awards, including the 1973 F. E. Terman Award, the 1985 AT&T Foundation Award of the American Society of Engineering Education, the 1989 Education Award of the IEEE Circuits and Systems Society, the 1989 Distinguished Senior U.S. Scientist Award from the Alexander von Humboldt Foundation of Germany, the 1996 Technical Achievement Award of the IEEE Signal Processing Society, the 1999 Mac Van Valkenburg Society Award and the CAS Golden Jubilee Medal of the IEEE Circuits & System Society, and the IEEE Millennium Medal in 2000. He is an Academician of the Academy of Finland. Dr. Mitra is a Fellow of the IEEE, AAAS, and SPIE and is a member of EURASIP and the ASEE.

# Preface

The field of digital signal processing (DSP) has seen explosive growth during the past three decades, as phenomenal advances both in research and application have been made. Fueling this growth have been the advances in digital computer technology and software development. Almost every electrical and computer engineering department in this country and abroad now offers one or more courses in digital signal processing, with the first course usually being offered at the senior level. This book is intended for a two-semester course on digital signal processing for seniors or first-year graduate students. It is also written at a level suitable for self-study by the practicing engineer or scientist.

Even though the first edition of this book was published barely two years ago, based on the feedback received from professors who adopted this book for their courses and many readers, it was clear that a new edition was needed to incorporate the suggested changes to the contents. A number of new topics have been included in the second edition. Likewise, a number of topics that are interesting but not practically useful have been removed because of size limitations. It was also felt that more worked-out examples were needed to explain new and difficult concepts.

The new topics included in the second edition are: calculation of total solution, zero-input response, zero-state response, and impulse response of finite-dimensional discrete-time systems (Sections 2.6.1–2.6.3), correlation of signals and its applications (Section 2.7), inverse systems (Section 4.9), system identification (Section 4.10), matched filter and its application (Section 4.14), sampling of bandpass signals (Section 5.3), design of highpass, bandpass, and bandstop analog filters (Section 5.5), effect of sample-and-hold operation (Section 5.11), design of highpass, bandpass, and bandstop IIR digital filters (Section 7.4), design of FIR digital filters with least-mean-square error (Section 7.8), constrained least-square design of FIR digital filters (Section 7.9), perfect reconstruction two-channel FIR filter banks (Section 10.9), cosine-modulated $L$-channel filter banks (Section 10.11), spectral analysis of random signals (Section 11.4), and sparse antenna array design (Section 11.14). The topics that have been removed from the first edition are as follows: state-space representation of LTI discrete-time systems from Chapter 2, signal flow-graph representation and state-space structures from Chapter 6, impulse invariance method of IIR filter design and FIR filter design based on the frequency-sampling approach from Chapter 7, reduction of product round-off errors from state-space structures from Chapter 9, and voice privacy system from Chapter 11. The fractional sampling rate conversion using the Lagrange interpolation has been moved to Chapter 10. Materials in each chapter are now organized more logically.

A key feature of this book is the extensive use of MATLAB® -based[1] examples that illustrate the program's powerful capability to solve signal processing problems. The book uses a three-stage pedagogical structure designed to take full advantage of MATLAB and to avoid the pitfalls of a "cookbook" approach to problem solving. First, each chapter begins by developing the essential theory and algorithms. Second, the material is illustrated with examples solved by hand calculation. And third, solutions are derived using MATLAB. From the beginning, MATLAB codes are provided with enough details to permit the students to repeat the examples on their computers. In addition to conventional theoretical problems requiring analytical solutions, each chapter also includes a large number of problems requiring solution via MATLAB. This book requires a minimal knowledge of MATLAB. I believe students learn the intricacies of problem solving with MATLAB faster by using tested, complete programs, and then writing simple programs to solve specific problems that are included at the ends of Chapters 2 to 11.

Because computer verification enhances the understanding of the underlying theories and, as in the first edition, a large library of worked-out MATLAB programs are included in the second edition. The original MATLAB programs of the first edition have been updated to run on the newer versions of MATLAB and the *Signal Processing Toolbox*. In addition, new MATLAB programs and code fragments have been added in this edition. The reader can run these programs to verify the results included in the book. Altogether there are 90 MATLAB programs in the text that have been tested under version 5.3 of MATLAB and version 4.2 of the *Signal Processing Toolbox*. Some of the programs listed in this book are not necessarily the fastest with regard to their execution speeds, nor are they the shortest. They have been written for maximum clarity without detailed explanations.

A second attractive feature of this book is the inclusion of 231 simple but practical examples that expose the reader to real-life signal processing problems which has been made possible by the use of computers in solving practical design problems. This book also covers many topics of current interest not normally found in an upper-division text. Additional topics are also introduced to the reader through problems at the end of each chapter. Finally, the book concludes with a chapter that focuses on several important, practical applications of digital signal processing. These applications are easy to follow and do not require knowledge of other advanced-level courses.

The prerequisite for this book is a junior-level course in linear continuous-time and discrete-time systems, which is usually required in most universities. A minimal review of linear systems and transforms is provided in the text, and basic materials from linear system theory are included, with important materials summarized in tables. This approach permits the inclusion of more advanced materials without significantly increasing the length of the book.

The book is divided into 11 chapters. Chapter 1 presents an introduction to the field of signal processing and provides an overview of signals and signal processing methods. Chapter 2 discusses the time-domain representations of discrete-time signals and discrete-time systems as sequences of numbers and describes classes of such signals and systems commonly encountered. Several basic discrete-time signals that play important roles in the time-domain characterization of arbitrary discrete-time signals and discrete-time systems are then introduced. Next, a number of basic operations to generate other sequences from one or more sequences are described. A combination of these operations is also used in developing a discrete-time system. The problem of representing a continuous-time signal by a discrete-time sequence is examined for a simple case. Finally, the time-domain characterization of discrete-time random signals is discussed.

Chapter 3 is devoted to the transform-domain representations of a discrete-time sequence. Specifically discussed are the discrete-time Fourier transform (DTFT), the discrete Fourier transform (DFT), and the $z$-transform. Properties of each of these transforms are reviewed and a few simple applications outlined. The chapter ends with a discussion of the transform-domain representation of a random signal.

This book concentrates almost exclusively on the linear time-invariant discrete-time systems, and

[1] MATLAB is a registered trademark of The MathWorks, Inc., 24 Prime Park Way, Natick, MA 01760-1500, Phone: 508-647-7000, **http://www.mathworks.com**.

Chapter 4 discusses their transform-domain representations. Specific properties of such transform-domain representations are investigated, and several simple applications are considered.

Chapter 5 is concerned primarily with the discrete-time processing of continuous-time signals. The conditions for discrete-time representation of a bandlimited continuous-time signal under ideal sampling and its exact recovery from the sampled version are first derived. Several interface circuits are used for the discrete-time processing of continuous-time signals. Two of these circuits are the anti-aliasing filter and the reconstruction filter, which are analog lowpass filters. As a result, a brief review of the basic theory behind some commonly used analog filter design methods is included, and their use is illustrated with MATLAB. Other interface circuits discussed in this chapter are the sample-and-hold circuit, the analog-to-digital converter, and the digital-to-analog converter.

A structural representation using interconnected basic building blocks is the first step in the hardware or software implementation of an LTI digital filter. The structural representation provides the relations between some pertinent internal variables with the input and the output, which in turn provides the keys to the implementation. There are various forms of the structural representation of a digital filter, and two such representations are reviewed in Chapter 6, followed by a discussion of some popular schemes for the realization of real causal IIR and FIR digital filters. In addition, it describes a method for the realization of IIR digital filter structures that can be used for the generation of a pair of orthogonal sinusoidal sequences.

Chapter 7 considers the digital filter design problem. First, it discusses the issues associated with the filter design problem. Then it describes the most popular approach to IIR filter design, based on the conversion of a prototype analog transfer function to a digital transfer function. The spectral transformation of one type of IIR transfer function into another type is discussed. Then a very simple approach to FIR filter design is described. Finally, the chapter reviews computer-aided design of both IIR and FIR digital filters. The use of MATLAB in digital filter design is illustrated.

Chapter 8 is concerned with the implementation aspects of DSP algorithms. Two major issues concerning implementation are discussed first. The software implementations of digital filtering and DFT algorithms on a computer using MATLAB are reviewed to illustrate the main points. This is followed by a discussion of various schemes for the representation of number and signal variables on digital machines, which is basic to the development of methods for the analysis of finite wordlength effects considered in Chapter 9. Algorithms used to implement addition and multiplication, the two key arithmetic operations in digital signal processing, are reviewed next, along with operations developed to handle overflow. Finally, the chapter outlines two general methods for the design and implementation of tunable digital filters, followed by a discussion of algorithms for the approximation of certain special functions.

Chapter 9 is devoted to analysis of the effects of the various sources of quantization errors; it describes structures that are less sensitive to these effects. Included here are discussions on the effect of coefficient quantization.

Chapter 10 discusses multirate discrete-time systems with unequal sampling rates at various parts. The chapter includes a review of the basic concepts and properties of sampling rate alteration, design of decimation and interpolation digital filters, and multirate filter bank design.

The final chapter, Chapter 11, reviews a few simple practical applications of digital signal processing to provide a glimpse of its potential.

The materials in this book have been used in a two-quarter course sequence on digital signal processing at the University of California, Santa Barbara, and have been extensively tested in the classroom for over 10 years. Basically, Chapters 2 through 6 form the basis of an upper-division course, while Chapters 7 through 10 form the basis of a graduate-level course.

Many topics included in this text can be omitted from class discussion, depending on the coverage of other courses in the curriculum. Because a senior-level course on random signals and systems is required of all electrical and computer engineering majors in most universities, materials in Sections 2.7, 3.10, and 4.9 can be excluded from an upper-division course on digital signal processing. However, these topics are important in the analysis of wordlength effects discussed in Chapter 9, and readers not familiar with this subject are encouraged to review these sections before reading Chapter 9. Likewise, Section 8.4 on number representation and Section 8.5 on arithmetic operations can similarly be omitted from discussion since most students taking a digital signal processing course usually take a course on digital hardware design.

This text contains 231 examples, 90 MATLAB programs, 684 problems, and 186 MATLAB exercises.

Every attempt has been made to ensure the accuracy of all materials in this book, including the MATLAB programs. I would, however, appreciate readers bringing to my attention any errors that may appear in the printed version for reasons beyond my control and that of the publisher. These errors and any other comments can be communicated to me by e-mail addressed to: **mitra@ece.ucsb.edu**.

Finally, I have been particularly fortunate to have had the opportunity to work with the outstanding students who were in my research group during my teaching career, which spans over 35 years. I have benefited immensely, and continue to do so, both professionally and personally, from my friendship and association with them, and to them I dedicate this book.

Sanjit K. Mitra

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## Supplements

All MATLAB programs included in this book are available via anonymous file transfer protocol (FTP) from the Internet site **iplserv.ece.ucsb.edu** in the directory **/pub/mitra/Book_2e**.

A solutions manual prepared by Rajeev Gandhi, Serkan Hatipoglu, Zhihai He, Luca Lucchese, Michael Moore, and Mylene Queiroz de Farias and containing the solutions to all problems and MATLAB exercises is available to instructors from the publisher.

A companion book *Digital Signal Processing Laboratory Using MATLAB* by the author is also available from McGraw-Hill.

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