国外电子与通信教材系列

英文版

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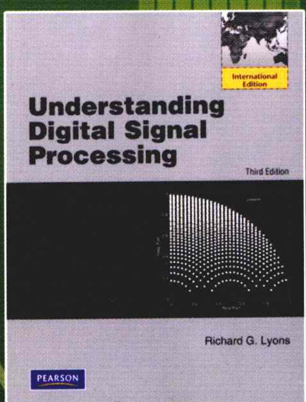
数字信号处理
（第三版）

Understanding Digital Signal Processing
Third Edition



[美] Richard G. Lyons 著

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Publishing House of Electronics Industry
北京 · BEIJING

内容简介

本书全面讨论了数字信号处理的基本概念、原理和应用。全书共13章，主要包括离散序列和系统、离散傅里叶变换及其快速算法、有限和无限冲激响应滤波器设计基本原理等基本数字信号处理内容，另外包括数字网络和滤波器、离散希尔伯特变换、抽样率的变换和信号平均、信号数字化及其影响等专业信号处理内容。全书包含作者多年总结出的一些数字信号处理技巧，包括如何进行复数的快速乘法、实序列的快速傅里叶变换、使用快速傅里叶变换的有限冲激响应滤波器设计等。附录对数字信号处理涉及的数学知识和术语给出了详细介绍和总结。相比于前版，本书每章都新增了部分内容，并附了习题，便于读者自学。

本书可作为理工类大专院校电子、计算机、通信等专业的双语课程教材，对于数字信号处理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、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。

电子工业出版社

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Preface[①]

This book is an expansion of previous editions of *Understanding Digital Signal Processing*. Like those earlier editions, its goals are (1) to help beginning students understand the theory of digital signal processing (DSP) and (2) to provide practical DSP information, not found in other books, to help working engineers/ scientists design and test their signal processing systems. Each chapter of this book contains new information beyond that provided in earlier editions.

It's traditional at this point in the preface of a DSP textbook for the author to tell readers why they should learn DSP. I don't need to tell you how important DSP is in our modern engineering world. You already know that. I'll just say that the future of electronics is DSP, and with this book you will not be left behind.

FOR INSTRUCTORS

This third edition is appropriate as the text for a one- or two-semester undergraduate course in DSP. It follows the DSP material I cover in my corporate training activities and a signal processing course I taught at the University of California Santa Cruz Extension. To aid students in their efforts to learn DSP, this third edition provides additional explanations and examples to increase its tutorial value. To test a student's understanding of the material, homework problems have been included at the end of each chapter. (For qualified instructors, a Solutions Manual is available. Please contact your local Pearson office.)

FOR PRACTICING ENGINEERS

To help working DSP engineers, the changes in this third edition include, but are not limited to, the following:

- Practical guidance in building discrete differentiators, integrators, and matched filters
- Descriptions of statistical measures of signals, variance reduction by way of averaging, and techniques for computing real-world signal-to- noise ratios (SNRs)
- A significantly expanded chapter on sample rate conversion (multirate systems) and its associated filtering
- Implementing fast convolution (FIR filtering in the frequency domain)
- IIR filter scaling
- Enhanced material covering techniques for analyzing the behavior and performance of digital filters
- Expanded descriptions of industry-standard binary number formats used in modern processing systems
- Numerous additions to the popular "Digital Signal Processing Tricks" chapter

FOR STUDENTS

Learning the fundamentals, and how to speak the language, of digital signal processing does not require profound analytical skills or an extensive background in mathematics. All you need is a little experience with elementary algebra, knowledge of what a sinewave is, this book, and enthusiasm. This may sound hard

① 采用本书作为教材的教师可获得本书配套教辅和习题解答（英文版）。详见书末所附“教学支持说明”。

to believe, particularly if you've just flipped through the pages of this book and seen figures and equations that look rather complicated. The content here, you say, looks suspiciously like material in technical journals and textbooks whose meaning has eluded you in the past. Well, this is not just another book on digital signal processing.

In this book I provide a gentle, but thorough, explanation of the theory and practice of DSP. The text is not written so that you *may* understand the material, but so that you *must* understand the material. I've attempted to avoid the traditional instructor–student relationship and have tried to make reading this book seem like talking to a friend while walking in the park. I've used just enough mathematics to help you develop a fundamental understanding of DSP theory and have illustrated that theory with practical examples.

I have designed the homework problems to be more than mere exercises that assign values to variables for the student to plug into some equation in order to compute a result. Instead, the homework problems are designed to be as educational as possible in the sense of expanding on and enabling further investigation of specific aspects of DSP topics covered in the text. Stated differently, the homework problems are not designed to induce "death by algebra," but rather to improve your understanding of DSP. Solving the problems helps you become proactive in your own DSP education instead of merely being an inactive recipient of DSP information.

THE JOURNEY

Learning digital signal processing is not something you accomplish; it's a journey you take. When you gain an understanding of one topic, questions arise that cause you to investigate some other facet of digital signal processing.[①] Armed with more knowledge, you're likely to begin exploring further aspects of digital signal processing much like those shown in the diagram on page xviii. This book is your tour guide during the first steps of your journey.

You don't need a computer to learn the material in this book, but it would certainly help. DSP simulation software allows the beginner to verify signal processing theory through the time-tested *trial and error* process.[②] In particular, software routines that plot signal data, perform the fast Fourier transforms, and analyze digital filters would be very useful.

As you go through the material in this book, don't be discouraged if your understanding comes slowly. As the Greek mathematician Menaechmus curtly remarked to Alexander the Great, when asked for a quick explanation of mathematics, "There is no royal road to mathematics." Menaechmus was confident in telling Alexander the only way to learn mathematics is through careful study. The same applies to digital signal processing. Also, don't worry if you need to read some of the material twice. While the concepts in this book are not as complicated as quantum physics, as mysterious as the lyrics of the song "Louie Louie," or as puzzling as the assembly instructions of a metal shed, they can become a little involved. They deserve your thoughtful attention. So, go slowly and read the material twice if necessary; you'll be glad you did. If you show persistence, to quote Susan B. Anthony, "Failure is impossible."

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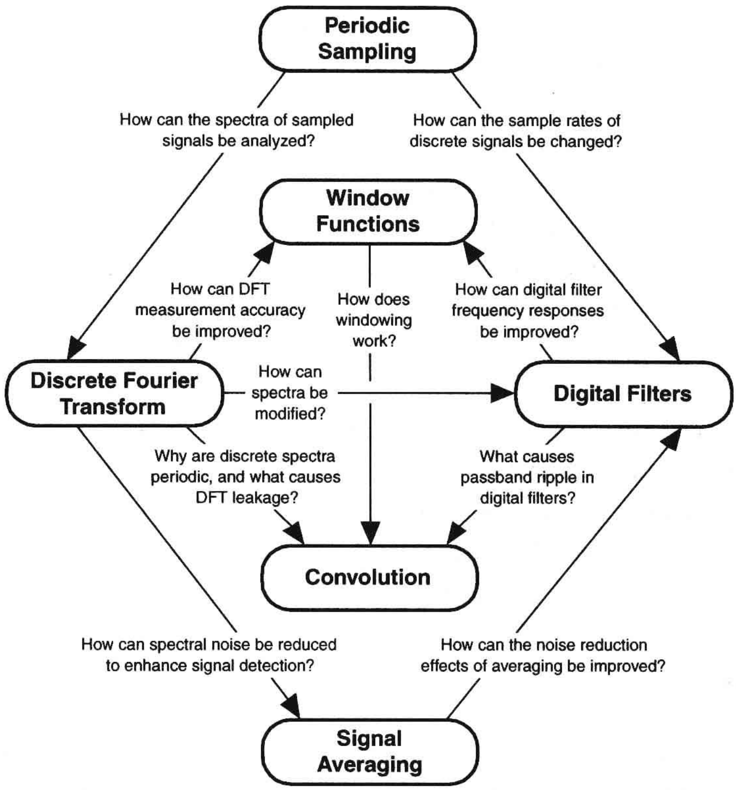
① "You see I went on with this research just the way it led me. This is the only way I ever heard of research going. I asked a question, devised some method of getting an answer, and got—a fresh question. Was this possible, or that possible? You cannot imagine what this means to an investigator, what an intellectual passion grows upon him. You cannot imagine the strange colourless delight of these intellectual desires" (Dr. Moreau—infamous physician and vivisectionist from H.G. Wells' *Island of Dr. Moreau,* 1896).

② "One must learn by doing the thing; for though you think you know it, you have no certainty until you try it" (Sophocles, 496–406 B.C.).



COMING ATTRACTIONS

Chapter 1 begins by establishing the notation used throughout the remainder of the book. In that chapter we introduce the concept of discrete signal sequences, show how they relate to continuous signals, and illustrate how those sequences can be depicted in both the time and frequency domains. In addition, Chapter 1 defines the operational symbols we'll use to build our signal processing system block diagrams. We conclude that chapter with a brief introduction to the idea of linear systems and see why linearity enables us to use a number of powerful mathematical tools in our analysis.

Chapter 2 introduces the most frequently misunderstood process in digital signal processing, periodic sampling. Although the concept of sampling a continuous signal is not complicated, there are mathematical subtleties in the process that require thoughtful attention. Beginning gradually with simple examples of lowpass sampling, we then proceed to the interesting subject of bandpass sampling. Chapter 2 explains and quantifies the frequency-domain ambiguity (aliasing) associated with these important topics.

Chapter 3 is devoted to one of the foremost topics in digital signal processing, the discrete Fourier transform (DFT) used for spectrum analysis. Coverage begins with detailed examples illustrating the important properties of the DFT and how to interpret DFT spectral results, progresses to the topic of windows used to reduce DFT leakage, and discusses the processing gain afforded by the DFT. The chapter concludes with a detailed discussion of the various forms of the transform of rectangular functions that the reader is likely to encounter in the literature.

Chapter 4 covers the innovation that made the most profound impact on the field of digital signal processing, the fast Fourier transform (FFT). There we show the relationship of the popular radix-2 FFT to the DFT, quantify the powerful processing advantages gained by using the FFT, demonstrate why the FFT functions as it does, and present various FFT implementation structures. Chapter 4 also includes a list of recommendations to help the reader use the FFT in practice.

Chapter 5 ushers in the subject of digital filtering. Beginning with a simple lowpass finite impulse response (FIR) filter example, we carefully progress through the analysis of that filter's frequency-domain magnitude and phase response. Next, we learn how window functions affect, and can be used to design, FIR filters. The methods for converting lowpass FIR filter designs to bandpass and highpass digital filters are presented, and the popular Parks-McClellan (Remez) Exchange FIR filter design technique is introduced and illustrated by example. In that chapter we acquaint the reader with, and take the mystery out of, the process called *convolution.* Proceeding through several simple convolution examples, we conclude Chapter 5 with a discussion of the powerful convolution theorem and show why it's so useful as a qualitative tool in understanding digital signal processing.

Chapter 6 is devoted to a second class of digital filters, infinite impulse response (IIR) filters. In discussing several methods for the design of IIR filters, the reader is introduced to the powerful digital signal processing analysis tool called the *z*-transform. Because the *z*-transform is so closely related to the continuous Laplace transform, Chapter 6 starts by gently guiding the reader from the origin, through the properties, and on to the utility of the Laplace transform in preparation for learning the *z*-transform. We'll see how IIR filters are designed and implemented, and why their performance is so different from that of FIR filters. To indicate under what conditions these filters should be used, the chapter concludes with a qualitative comparison of the key properties of FIR and IIR filters.

Chapter 7 introduces specialized networks known as *digital differentiators*, *integrators*, and *matched filters*. In addition, this chapter covers two specialized digital filter types that have not received their deserved exposure in traditional DSP textbooks. Called *interpolated FIR* and *frequency sampling* filters, providing improved lowpass filtering computational efficiency, they belong in our arsenal of filter design techniques. Although these are FIR filters, their introduction is delayed to this chapter because familiarity with the *z*-transform (in Chapter 6) makes the properties of these filters easier to understand.

Chapter 8 presents a detailed description of quadrature signals (also called *complex* signals). Because quadrature signal theory has become so important in recent years, in both signal analysis and digital communications implementations, it deserves its own chapter. Using three-dimensional illusrations, this chapter gives solid physical meaning to the mathematical notation, processing advantages, and use of quadrature signals. Special emphasis is given to quadrature sampling (also called *complex down-conversion*).

Chapter 9 provides a mathematically gentle, but technically thorough, description of the Hilbert transform—a process used to generate a quadrature (complex) signal from a real signal. In this chapter we describe the properties, behavior, and design of practical Hilbert transformers.

Chapter 10 presents an introduction to the fascinating and useful process of sample rate conversion (changing the effective sample rate of discrete data sequences through decimation or interpolation). Sample rate conversion—so useful in improving the performance and reducing the computational complexity of many signal processing operations—is essentially an exercise in lowpass filter design. As such, polyphase and cascaded integrator-comb filters are described in detail in this chapter.

Chapter 11 covers the important topic of signal averaging. There we learn how averaging increases the accuracy of signal measurement schemes by reducing measurement background noise. This accuracy enhancement is called *processing gain*, and the chapter shows how to predict the processing gain associated with averaging signals in both the time and frequency domains. In addition, the key differences between coherent and incoherent averaging techniques are explained and demonstrated with examples. To complete that chapter the popular scheme known as *exponential averaging* is covered in some detail.

Chapter 12 presents an introduction to the various binary number formats the reader is likely to encounter in modern digital signal processing. We establish the precision and dynamic range afforded by these formats along with the inherent pitfalls associated with their use. Our exploration of the critical subject of binary data word width (in bits) naturally leads to a discus- sion of the numerical resolution limitations of analog-to-digital (A/D) converters and how to determine the optimum A/D converter word size for a given application. The problems of data value overflow roundoff errors are covered along with a statistical introduction to the two most popular remedies for overflow, truncation and rounding. We end that chapter by covering the interesting subject of floating-point binary formats that allow us to overcome most of the limitations induced by fixed-point binary formats, particularly in reducing the ill effects of data overflow.

Chapter 13 provides the literature's most comprehensive collection of *tricks of the trade* used by DSP professionals to make their processing algorithms more efficient. These techniques are compiled into a chapter at the end of the book for two reasons. First, it seems wise to keep our collection of tricks in one chapter so that we'll know where to find them in the future. Second, many of these clever schemes require an understanding of the material from the previous chapters, making the last chapter an appropriate place to keep our arsenal of clever tricks. Exploring these techniques in detail verifies and reiterates many of the important ideas covered in previous chapters.

The appendices include a number of topics to help the beginner under- stand the nature and mathematics of digital signal processing. A comprehensive description of the arithmetic of complex numbers is covered in Appendix A, and Appendix B derives the often used, but seldom explained, closed form of a geometric series. The subtle aspects and two forms of time reversal in discrete systems (of which zero-phase digital filtering is an application) are explained in Appendix C. The statistical concepts of mean, variance, and standard deviation are introduced and illustrated in Appendix D, and Appendix E provides a discussion of the origin and utility of the logarithmic decibel scale used to improve the magnitude resolution of spectral representations. Appendix F, in a slightly different vein, provides a glossary of the terminology used in the field of digital filters. Appendices G and H provide supplementary information for designing and analyzing specialized digital filters. Appendix I explains the computation of Chebyshev window sequences.

ACKNOWLEDGMENTS

Much of the new material in this edition is a result of what I've learned from those clever folk on the USENET newsgroup comp.dsp. (I could list a dozen names, but in doing so I'd make 12 friends and 500 enemies.) So, I say thanks to my DSP pals on comp.dsp for teaching me so much signal processing theory.

In addition to the reviewers of previous editions of this book, I thank Randy Yates, Clay Turner, and Ryan Groulx for their time and efforts to help me improve the content of this book. I am especially indebted to my eagleeyed mathematician friend Antoine Trux for his relentless hard work to both enhance this DSP material and create a homework Solutions Manual.

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If you're still with me this far in this Preface, I end by saying I had a ball writing this book and sincerely hope you benefit from reading it. If you have any comments or suggestions regarding this material, or detect any errors no matter how trivial, please send them to me at R.Lyons@ieee.org. I promise I will reply to your e-mail.

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