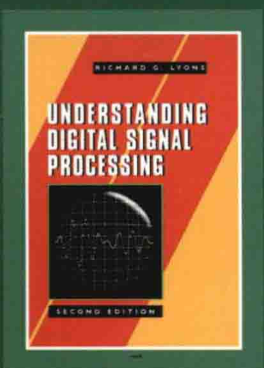
国外电子与通信教材系列

Pearson

精心改编

Understanding Digital Signal Processing



数字信号处理

（第二版）（英文版）

[美]　Richard G. Lyons　著
张建华　许晓东　孙松林　改编

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Understanding Digital Signal Processing

Second Edition

［美］ Richard G. Lyons　著

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電子工業出版社
Publishing House of Electronics Industry
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内容简介

本书在Richard G. Lyons所著*Understanding Digital Signal Processing, Second Edition*的基础上进行了改编，针对通信类学校本科教学大纲，删除了教学中一般不涉及的内容，调整了章节顺序，并增加了z反变换、滤波器结构、线性相位FIR滤波器及其结构、模拟滤波器简介的内容，使教材内容更加完整。全书在概述了离散序列和系统的定义和实例之后，详细讨论了离散系统的特性、信号的离散化和离散卷积、z变换、离散时间傅里叶变换和离散傅里叶变换、快速傅里叶变换、数字滤波器结构，以及有限和无限脉冲响应数字滤波器的设计等基本概念和基本理论。书中涉及的数学知识以简明形式给出，深入浅出，易于理解。本书每章都增加了例题、习题和MATLAB例题，以便加强对每章内容的理解和掌握。

本书可作为理工类大专院校电子、计算机、通信等专业的本科生双语教学教材，对于DSP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

教材出版委员会

改编者简介

张建华

女，博士，北京邮电大学岗位特聘教授。2002年被选送到德国汉堡–哈堡科技大学深造半年，从2005年开始从事数字信号处理课程的双语教学工作，2008年受国家留学基金委支持前往韩国访问学习半年。先后主持和参加了10余个国家自然科学基金、国家863计划重点项目、国家科技重大专项和企业合作项目。截至2010年4月，在国际会议和期刊已发表90余篇文章，其中SCI检索9篇（发表在JCN杂志的文章获得2008年唯一最佳论文奖），目前担任IEEE多种杂志和期刊的审稿人；申请专利20余项，授权4项；2005年入选北京市科技新星人才培养计划。2006年获中国通信学会科学技术一等奖1次，2008年获国家技术发明奖二等奖1次，2009年获中国通信标准化协会科学技术二等奖1次。鉴于此，2010年4月被北京邮电大学评为“岗位特聘教授”。

许晓东

男，博士，北京邮电大学副教授。研究方向为下一代移动通信系统及通信系统中的信号处理技术，于2007年开始从事数字信号处理课程的双语教学工作。目前已出版包括普通高等教育“十一五”国家级规划教材在内的教材及专著3部。已承担及参与自然科学基金重大项目子课题、863高技术项目、中瑞国际合作项目、北京市科技计划项目、企业合作项目10余项。在国内外期刊和国际会议发表学术论文40余篇，其中6篇为SCI检索论文，目前担任IEEE多种杂志及国内外学术期刊的审稿人。已申请移动通信及信号处理相关发明专利12项，其中1项已授权，3项专利相关的标准提案被3GPP和3GPP2等标准化组织接纳。

孙松林

男，博士，北京邮电大学副教授。研究方向为无线多媒体通信与信号处理、嵌入式系统。先后主持和参加了10余个国家自然科学基金、国家科技重大专项、省部级预研和企业合作项目。在国内外期刊和国际会议发表学术论文20余篇，合作出版教材和专著3部。是*IEEE Networking*和《电子学报》等多种期刊杂志的审稿人。

改编者序

本书是在 *Understanding Digital Signal Processing, Second Edition* 的基础上改编的英文版教材，适合数字信号处理（DSP）基础相关课程40学时到64学时的英语或双语教学使用。

本书的作者Richard G. Lyons毕业于美国加州阿克隆大学。他曾是美国国家安全局和TRW（天合）汽车集团公司众多DSP系统的首席硬件工程师，负责系统设计、开发、测试和安装。他曾在加利福尼亚大学圣塔克鲁兹分校任教，并发表过很多有关DSP的文章。作为*IEEE Signal Processing Magazine* 的副主编，由他开设了DSP Tips & Tricks专栏并且负责编辑。正是由于其在DSP领域的长期工作经验和教学经历，编写的此书受到了很多数字信号处理初学者和使用者的高度评价，认为此书将“理论与实践完美地结合起来”。由于数字信号处理是一门应用性很强的课程，需要大量的经验积累，而此书通过具有启发性的解释和精心挑选的例子，采用读者可以理解的数学表示方法，对数字信号处理技术进行了解释，帮助读者从整体上掌握DSP，并逐步掌握较高层次的DSP概念和应用。这也是改编者选择此书的初衷，希望读者不仅了解DSP的概念，而且通过例子的学习，在今后的科研工作中，会用DSP的知识解决实际问题。

Understanding Digital Signal Processing, Second Edition 的内容体系与国内高校数字信号处理基础相关课程的教学内容有较高的统一性，但是，通过亲自实践以及进一步考察国内高校相关学科开展双语教学的情况，也发现本书与国内高校的教学内容和课程安排存在一定不匹配之处。自国内高校教育改革以来，大部分院校都开设了DSP课程，而且内容也基本形成了一致，而本书的内容相对国内的授课学时和知识体系显得内容过于庞大。例如，关于信号量化的内容在国内高等院校的授课中通常是介绍性的，而原书用了整章的篇幅（第8章）来详细介绍；关于离散希尔伯特变换、信号平均和数字信号格式等内容，通常在国内数字信号处理基础课程教学中也很少涉及，或者属于只要求学生了解的内容。但对于数字信号处理使用的常用方法，MATLAB在国内一般都会作为数字信号处理实验的内容来介绍，但本书却没有涉及。

为了使教材内容更合理，范围宽窄适度，内容深浅适中，进而满足国内和国际高等教育相关专业数字信号处理课程的教学要求，适应国内教育教学特色，并便于学生在有限学时下把握重点，深入理解，而且学以致用，我们以精简内容和突出重点为目标对这本书进行了改编。删除了教学中一般不涉及的内容，调整了章节顺序，并增加了z反变换、滤波器结构、线性相位FIR滤波器及其结构、模拟滤波器简介等内容，使教材内容更加完整。全书在概述了离散序列和系统的定义及实例之后，详细讨论了离散系统的特性、信号的离散化和离散卷积、z变换、离散时间傅里叶变换和离散傅里叶变换、快速傅里叶变换、数字滤波器结构及有限和

无限脉冲响应数字滤波器的设计等基本概念和基本理论。书中涉及的数学知识以简明形式给出，深入浅出，易于理解。本书每章都增加了例题、习题和MATLAB例题，便于加强对每章内容的理解和掌握。本书可作为理工类大专院校的电子、计算机、通信等专业的本科生教材，对于DSP领域的工程技术人员和专业技术人员也有很好的参考价值。

本书第1章至第3章由许晓东老师负责改编，第4章至第6章由孙松林老师负责改编，第7章和第8章由张建华老师负责改编，全书由张建华老师负责统校。在本书的改编过程中，一方面结合了改编者们在北京邮电大学国际学院开展数字信号处理双语授课的实践经验（教学用PPT等教辅资源可联系malan@phei.com.cn申请），另一方面，北京邮电大学门爱东教授、苏菲教授和已退休的周利清教授都给予我们许多帮助和指导，在此表示衷心感谢！最后感谢热心的国际学院已毕业的学生王禹宁和闫屾等对本书的关注，感谢许多无法一一提及的支持此书的老师和同学们！

Preface

This book is an expansion of the original *Understanding Digital Signal Processing* textbook published in 1997 and, like the first edition, its goal is to help beginners understand this relatively new technology of digital signal processing (DSP). Additions to this second edition include:

- Expansion and clarification of selected spectrum analysis and digital filtering topics covered in the first edition making that material more valuable to the DSP beginner.
- Discussions of Frequency Sampling, Interpolated FIR, and CIC filters; giving these important filters greater exposure than they've received in past DSP textbooks.
- Revision of the terminology making it more consistent with the modern day language of DSP.

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.

LEARNING DIGITAL SIGNAL PROCESSING

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 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 the material in technical journals and textbooks that, in the past, have successfully resisted your attempts to understand. Well, this is not just another book on digital signal processing.

This book's goal is to gently provide explanation followed by illustration, not so that you may understand the material, but that you must understand the material.[①] Remember the first time you saw two people playing chess? The game probably appeared to be mysterious and confusing. As you now know, no individual chess move is complicated. Given a little patience, the various chess moves are easy to learn. The game's complexity comes from deciding what combinations of moves to make and when to make them. So it is with understanding digital signal processing. First we learn the fundamental rules and processes, and then practice using them in combination.

If learning digital signal processing is so easy, then why does the subject have the reputation of being hard to understand? The answer lies partially in how the material is typically presented in the literature. It's difficult to convey technical information, with its mathematical subtleties, in written

① "Here we have the opportunity of expounding more clearly what has already been said" (Rene Descartes, 1596-1650).

form. It's one thing to write equations, but it's another matter altogether to explain what those equations really mean from a practical standpoint, and that's the goal of this book.

Too often, written explanation of digital signal processing theory appears in one of two forms: either mathematical miracles occur and the reader is simply given a short and sweet equation without further explanation, or the reader is engulfed in a flood of complex variable equations and phrases such as "it is obvious that," and "with judicious application of the homogeneity property." In their defense, authors usually do provide the needed information, but too often the reader must figuratively grab a pick and shovel, put on a miner's helmet, and try to dig the information out of a mountain of mathematical expressions. (This book presents the results of several fruitful mining expeditions.) How many times have you followed the derivation of an equation, after which the author states they're going to illustrate that equation with an example—which turns out to be just another equation? Although mathematics is necessary to describe digital signal processing, I've tried to avoid overwhelming the reader with math because a recipe for technical writing that's too rich in equations is hard for the beginner to digest.

The intent of this book is expressed by a popular quote from E. B. White in the introduction of his *Elements of Style* (Macmillan Publishing, New York, 1959):

> "Will (Strunk) felt that the reader was in serious trouble most of the time, a man floundering in a swamp, and that it was the duty of anyone attempting to write English to drain the swamp quickly and get his man up on dry ground, or at least throw him a rope."

I've attempted to avoid the traditional instructor-student relationship, but rather to make reading this book like talking to a friend while walking in the park. I've used just enough mathematics to develop a fundamental understanding of the theory, and then illustrate that theory with practical examples.

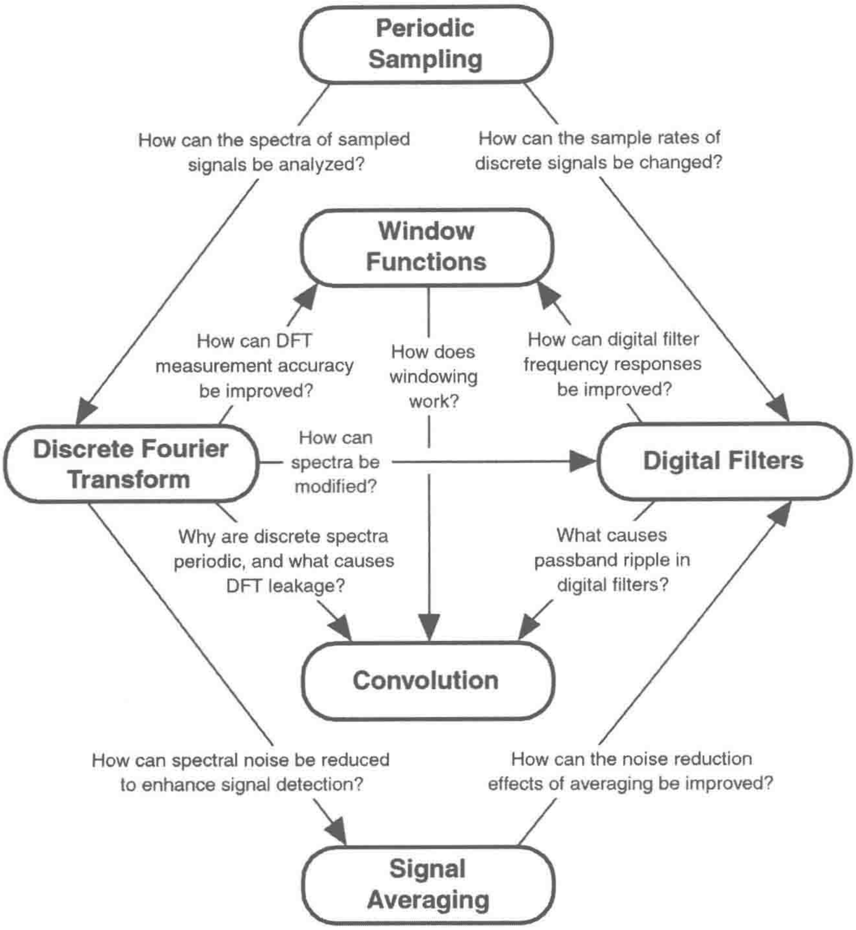
THE JOURNEY

Learning digital signal processing is not something you accomplish; it's a journey you take. When you gain an understanding of some 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 following diagram. 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 sure 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.

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—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.).



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 have 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 do get a little involved. They deserve your attention and thought. So go slow and read the material twice if you have to; you'll be glad you did. If you show persistence, to quote a phrase from Susan B. Anthony, "Failure is impossible."

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 it's straightforward to grasp the concept of sampling a continuous signal, there are mathematical subtleties in the process that require thoughtful attention. Beginning gradually with simple examples of low-pass sampling, Chapter 2 explains and quantifies the frequency domain ambiguity (aliasing) associated with these important topics.

Chapter 3 is one chapter to explain z transform. z transform is one of very useful tools to analyze discrete signals and systems. Thus we will learn how to transform signals or systems to z domain and also how to inverse transform z signals and system to its time domain form. From this chapter, we will master more powerful mathematical tools in later DSP studying.

Chapter 4 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 beginner is likely to encounter in the literature. That last topic is included there to clarify and illustrate the DFT of both real and complex sinusoids.

Chapter 5 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 5 also includes a list of recommendations to help the reader use the FFT in practice.

Chapter 6 introduces reader about FIR and IIR filter structures. For structure will affect the efficiency and stability of filter. In Chapter 6, block structure is firstly presented and Mason theorem is used to get its transfer function for the specified structure. FIR and IIR typical structure are also summarized in this chapter.

Chapter 7 ushers in the subject of digital filtering. Beginning with a simple low-pass 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 low-pass FIR filter designs to bandpass and high pass digital filters are presented.

Chapter 8 is devoted to a second class of digital filters, infinite impulse response (IIR) filters. In discussing several methods for the design of IIR filters, Chapter 8 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 FIR filters. To indicate under what conditions these filters should be used, that chapter concludes with a qualitative comparison of the key properties of FIR and IIR filters.

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