



Statistical and Adaptive Signal Processing

Spectral Estimation, Signal Modeling,
Adaptive Filtering and Array Processing

统计与自适应信号处理

谱估计、信号建模、自适应滤波
和阵列信号处理

Dimitris G. Manolakis
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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。

Statistical and Adaptive Signal Processing

影印版序

最近十几年, 信号处理学科获得了迅速的发展, 其应用已遍及各重要工业部门。可以说, 地震勘探、雷达、声纳、移动通信、生物医学和生物信息、语音信号处理等的发展为信号处理学科提供了急迫的应用需求和辽阔的应用沃土, 大大刺激了信号处理理论与技术的发展。反过来, 信号处理的新理论、新方法和新技术的发展又有力地推动和促进了上述工业部门和领域的技术进步。不仅如此, 信号处理学科还与数学和物理学科密切相关: 信号处理可以说是应用数学味最浓的工程学科之一, 它的很多概念和术语又与物理学有着直接的渊源。例如, 信号的高阶统计分析和小波分析得力于数学的发展, 而量子物理和计算物理甚至分别催生了量子信号处理和 Monte Carlo 信号处理这两门新技术。概而言之, 信号处理由于理论性和实践性都很强, 已成为电子、通信、自动化、电机、机械工程、生物医学和生物信息、海洋或水声工程等工科高年级大学生和研究生越来越受欢迎的主修课程之一。

为了配合全国高等院校、科研院所的博士、硕士点学科建设和研究生教学, 清华大学出版社决定引进国外一批高水平的原版教材, 影印出版。其中, 信号处理的教材已引进过《Digital Signal Processing — A Computer-Based Approach》(Mitra SK, McGraw-Hill, 2001) 一书, 读者对象为高年级大学生; 本书《Statistical and Adaptive Signal Processing》是最新引进的一种, 主要面向研究生。

本书也有一个副标题 “Spectral Estimation, Signal Modeling, Adaptive Filtering and Array Processing”, 它形象地概括了全书的主要内容为谱估计、信号建模、自适应滤波和阵列信号处理。围绕这四部分主要内容, 全书共编排了 12 章。前 4 章为准备知识: 第 1 章为引言, 介绍统计与自适应信号处理的基本概念和应用; 第 2 和第 3 章复习离散时间信号处理的基础知识, 讨论时域和频域的随机向量与随机序列, 介绍估计理论的基本概念; 第 4 章介绍线性信号模型。从第 5 章开始, 进入本书的主体部分。第 5 章的主题是基于离散 Fourier 变换的非参数化功率谱估计。第 6 和第 7 章分别介绍最优线性滤波器的理论、算法及结构。第 8 章专门讨论最小二乘滤波和预测。第 9 章为信号建模与参数化功率谱估计。第 10 章介绍自适应滤波器的理论和学习算法。第 11 章(阵列信号处理)可以看作是时域的谱估计和滤波理论在空间域的拓广: 阵列处理包含有波达方向估计和波束形成两大任务, 而波达方向估计实际上是一种空间谱估计, 波束形成则是一种空间滤波。最后, 第 12 章介绍信号处理的几个专题, 它们分别是高阶统计量理论、盲反卷积、盲均衡、分数零极点信号模型和自相似随机信号模型。

在阅读了全书后, 感到这是一本值得推荐的高水平教材, 其特点如下:

1. 全书内容非常丰富, 涵盖了谱估计、信号建模、自适应滤波和阵列信号处理的基本

理论与方法。

2. 书中所有的主要算法以及说明这些算法应用的例子都提供了 MATLAB 函数, 它们可以通过 <http://www.mhhe.com/catalogs/007040512.mhtml> 找到。

3. 插图丰富, 并编有大量习题。

该书由美国麻省理工学院和东北大学的三位学者联合著作。其中, 第一作者 Manolakis 博士和第二作者 Ingle 副教授近几年曾在多所大学讲授过信号处理课, 并有着丰富的研究经历, 还分别出版过其他多本信号处理教材。第三作者 Kogon 博士则主要从事信号处理的研究。三位学者丰富的教学经验和对信号处理的深入研究, 使得本书既具有先进的理论水平, 又面向诸多的实际应用。因此, 本书不仅适合于电子、通信、自动化、电机工程、生物医学和生物信号、机械工程等专业研究生作为教材或教学参考书, 也适合用作广大工程技术人员的自学读本或参考用书。

清华大学出版社引进出版的一批国外教材已深受广大读者的好评和欢迎。笔者相信本书也一定会成为广大读者所喜爱的教材或参考书。

张贤达
清华大学自动化系
2003 年 6 月 12 日

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To my beloved wife, Anna, and to the loving memory of my father, Gregory.
DGM

To my beloved wife, Usha, and adoring daughters, Natasha and Trupti.
VKI

To my wife and best friend, Lorna, and my children, Gabrielle and Matthias.
SMK

*One must learn by doing the thing;
for though you think you know it
You have no certainty, until you try.*
—Sophocles, *Trachiniae*

PREFACE

The principal goal of this book is to provide a unified introduction to the theory, implementation, and applications of statistical and adaptive signal processing methods. We have focused on the key topics of spectral estimation, signal modeling, adaptive filtering, and array processing, whose selection was based on the grounds of theoretical value and practical importance. The book has been primarily written with students and instructors in mind. The principal objectives are to provide an introduction to basic concepts and methodologies that can provide the foundation for further study, research, and application to new problems. To achieve these goals, we have focused on topics that we consider fundamental and have either multiple or important applications.

APPROACH AND PREREQUISITES

The adopted approach is intended to help both students and practicing engineers understand the fundamental mathematical principles underlying the operation of a method, appreciate its inherent limitations, and provide sufficient details for its practical implementation. The academic flavor of this book has been influenced by our teaching whereas its practical character has been shaped by our research and development activities in both academia and industry. The mathematical treatment throughout this book has been kept at a level that is within the grasp of upper-level undergraduate students, graduate students, and practicing electrical engineers with a background in digital signal processing, probability theory, and linear algebra.

ORGANIZATION OF THE BOOK

Chapter 1 introduces the basic concepts and applications of statistical and adaptive signal processing and provides an overview of the book. Chapters 2 and 3 review the fundamentals of discrete-time signal processing, study random vectors and sequences in the time and frequency domains, and introduce some basic concepts of estimation theory. Chapter 4 provides a treatment of parametric linear signal models (both deterministic and stochastic) in the time and frequency domains. Chapter 5 presents the most practical methods for the estimation of correlation and spectral densities. Chapter 6 provides a detailed study of the theoretical properties of optimum filters, assuming that the relevant signals can be modeled as stochastic processes with known statistical properties; and Chapter 7 contains algorithms and structures for optimum filtering, signal modeling, and prediction. Chapter

8 introduces the principle of least-squares estimation and its application to the design of practical filters and predictors. Chapters 9, 10, and 11 use the theoretical work in Chapters 4, 6, and 7 and the practical methods in Chapter 8, to develop, evaluate, and apply practical techniques for signal modeling, adaptive filtering, and array processing. Finally, Chapter 12 introduces some advanced topics: definition and properties of higher-order moments, blind deconvolution and equalization, and stochastic fractional and fractal signal models with long memory. Appendix A contains a review of the matrix inversion lemma, Appendix B reviews optimization in complex space, Appendix C contains a list of the MATLAB functions used throughout the book, Appendix D provides a review of useful results from matrix algebra, and Appendix E includes a proof for the minimum-phase condition for polynomials.

USE OF THE BOOK

This book can be used in a number of different ways to teach either one-semester or two-semester graduate courses in a typical electrical engineering curriculum. Most topics in the book can be covered in a two-semester course on statistical and adaptive signal processing. Typical one-term courses are outlined in the following table:

Courses	Chapters
Discrete-time random signals and statistical signal processing	Review of Chapters 1, 2, and 3; Sections 4.1–4.4, 6.1–6.6
Spectrum estimation and signal modeling	Review of Chapters 1, 2, and 3; Chapters 4 and 5, Sections 6.1–6.5, 7.4–7.5, 8.1–8.9, Chapter 9, and possibly Sections 12.1, 12.5–12.6
Adaptive filtering	Review of Chapters 1, 2, and 3; Chapters 6, 7, 8, 10, and Sections 12.1–12.4
Introduction to array processing	Review of Chapter 3, Sections 6.1–6.5, review of Section 2.2, Sections 5.1, 11.1–11.4, Chapter 8, Sections 9.5, 9.6, 11.5, 11.6, 11.7

THEORY AND PRACTICE

It is our belief that sound theoretical understanding goes hand-in-hand with practical implementation and application to real-world problems. Therefore, the book includes a large number of computer experiments that illustrate important concepts and help the reader to easily implement the various methods. Every chapter includes examples, problems, and computer experiments that facilitate the comprehension of the material. To help the reader understand the theoretical basis and limitations of the various methods and apply them to real-world problems, we provide MATLAB functions for all major algorithms and examples illustrating their use. The MATLAB files and additional material about the book can be found at <http://www.mhhe.com/catalogs/0070400512.mhtml>. A Solutions Manual with detailed solutions to all the problems is available to the instructors adopting the book for classroom use.

FEEDBACK

Although we are fully aware that there always exists room for improvement, we believe that this book is a big step forward for an introductory textbook in statistical and adaptive signal processing. However, as engineers, we know that every search for the optimum

requires the will to change and quest for additional improvement. Thus, we would appreciate feedback from teachers, students, and engineers using this book for self-study at vingle@lynx.neu.edu.

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We are indebted to a number of individuals who have contributed in different, but important, ways to the shaping of our knowledge in general and the preparation of this book in particular. In this respect we wish to extend our appreciation to C. Caroubalos, G. Carayannis, J. Makhoul (DGM), M. Schetzen (VKI), and J. Holder, S. Krich, and D. Williams (SMK). We are grateful to E. Baranoski, G. Borsari, J. Schodorf, S. Smith, A. Steinhardt, and J. Ward for helping us to improve the presentation in various parts of the book.

We express our sincere gratitude to Kevin Donohue, University of Kentucky; Amro El-Jaroudi, University of Pittsburgh; Edward A. Lee, University of California-Berkeley; Ray Liu, University of Maryland; Randy Moses, The Ohio State University; and Kristina Ropella, Marquette University, for their constructive and helpful reviews.

Lynn Cox persuaded us to choose McGraw-Hill, Inc., as our publisher, and we have not regretted that decision. We are grateful to Lynn for her enthusiasm and her influence in shaping the scope and the objectives of our book. The fine team at McGraw-Hill, including Michelle Flomenhoft, Catherine Fields, Betsy Jones, and Nina Kreiden, has made the publication of this book an exciting and pleasant experience. We also thank N. Bullock and Mathworks, Inc., for promptly providing various versions of *MATLAB* and A. Turcotte for helping with some of the drawings in the book.

Last, but not least, we would like to express our sincere appreciation to our families for their full-fledged support and understanding over the past several years. We fully realize that the completion of such a project would not be possible without their continual sustenance and encouragement.

Dimitris G. Manolakis
Vinay K. Ingle
Stephen M. Kogon

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