

Applied Digital Signal Processing

Master the basic concepts and methodologies of digital signal processing with this systematic introduction, without the need for an extensive mathematical background. The authors lead the reader through the fundamental mathematical principles underlying the operation of key signal processing techniques, providing simple arguments and cases rather than detailed general proofs. Coverage of practical implementation, discussion of the limitations of particular methods, and plentiful MATLAB illustrations allow readers to better connect theory and practice. A focus on algorithms that are of theoretical importance or useful in real-world applications ensures that students cover material relevant to engineering practice, and equips students and practitioners alike with the basic principles necessary to apply DSP techniques to a variety of applications. Chapters include worked examples, problems, and computer experiments, helping students to absorb the material they have just read. Lecture slides for all figures and solutions to the numerous problems are available to instructors.

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THEORY AND PRACTICE

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To my wife and best friend Anna
and in memory of Eugenia, Gregory, and Elias

DGM

To my loving wife Usha and daughters
Natasha and Trupti for their endless support.

VKI

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Frontmatter
[More information](#)

CONTENTS

Preface	<i>page</i> xiii
1 Introduction	1
1.1 Signals	2
1.2 Systems	9
1.3 Analog, digital, and mixed signal processing	13
1.4 Applications of digital signal processing	16
1.5 Book organization	18
Learning summary	20
Terms and concepts	20
Further reading	21
Review questions	21
2 Discrete-time signals and systems	23
2.1 Discrete-time signals	24
2.2 Signal generation and plotting in MATLAB	27
2.3 Discrete-time systems	31
2.4 Convolution description of linear time-invariant systems	37
2.5 Properties of linear time-invariant systems	45
2.6 Analytical evaluation of convolution	50
2.7 Numerical computation of convolution	55
2.8 Real-time implementation of FIR filters	57
2.9 FIR spatial filters	59
2.10 Systems described by linear constant-coefficient difference equations	61
2.11 Continuous-time LTI systems	69
Learning summary	75
Terms and concepts	75
Further reading	78
Review questions	78
Problems	79
3 The z-transform	89
3.1 Motivation	90
3.2 The z -transform	91
3.3 The inverse z -transform	99
3.4 Properties of the z -transform	103
3.5 System function of LTI systems	106

Contents

3.6	LTI systems characterized by linear constant-coefficient difference equations	110
3.7	Connections between pole-zero locations and time-domain behavior	114
3.8	The one-sided z -transform	118
	Learning summary	121
	Terms and concepts	122
	Further reading	123
	Review questions	123
	Problems	124
4	Fourier representation of signals	134
4.1	Sinusoidal signals and their properties	135
4.2	Fourier representation of continuous-time signals	142
4.3	Fourier representation of discrete-time signals	157
4.4	Summary of Fourier series and Fourier transforms	169
4.5	Properties of the discrete-time Fourier transform	171
	Learning summary	188
	Terms and concepts	189
	Further reading	191
	Review questions	191
	Problems	192
5	Transform analysis of LTI systems	201
5.1	Sinusoidal response of LTI systems	202
5.2	Response of LTI systems in the frequency domain	210
5.3	Distortion of signals passing through LTI systems	215
5.4	Ideal and practical filters	221
5.5	Frequency response for rational system functions	224
5.6	Dependence of frequency response on poles and zeros	231
5.7	Design of simple filters by pole-zero placement	237
5.8	Relationship between magnitude and phase responses	247
5.9	Allpass systems	249
5.10	Invertibility and minimum-phase systems	254
5.11	Transform analysis of continuous-time LTI systems	258
	Learning summary	274
	Terms and concepts	275
	Further reading	276
	Review questions	277
	Problems	278
6	Sampling of continuous-time signals	292
6.1	Ideal periodic sampling of continuous-time signals	293
6.2	Reconstruction of a bandlimited signal from its samples	297
6.3	The effect of undersampling: aliasing	300

Contents

6.4	Discrete-time processing of continuous-time signals	311
6.5	Practical sampling and reconstruction	318
6.6	Sampling of bandpass signals	327
6.7	Image sampling and reconstruction	333
	Learning summary	339
	Terms and concepts	340
	Further reading	341
	Review questions	342
	Problems	343
7	The Discrete Fourier Transform	353
7.1	Computational Fourier analysis	354
7.2	The Discrete Fourier Transform (DFT)	357
7.3	Sampling the Discrete-Time Fourier Transform	363
7.4	Properties of the Discrete Fourier Transform	374
7.5	Linear convolution using the DFT	392
7.6	Fourier analysis of signals using the DFT	396
	Learning summary	418
	Terms and concepts	419
	Further reading	421
	Review questions	422
	Problems	423
8	Computation of the Discrete Fourier Transform	434
8.1	Direct computation of the Discrete Fourier Transform	435
8.2	The FFT idea using a matrix approach	436
8.3	Decimation-in-time FFT algorithms	440
8.4	Decimation-in-frequency FFT algorithms	450
8.5	Generalizations and additional FFT algorithms	454
8.6	Practical considerations	456
8.7	Computation of DFT for special applications	459
	Learning summary	470
	Terms and concepts	470
	Further reading	472
	Review questions	473
	Problems	474
9	Structures for discrete-time systems	485
9.1	Block diagrams and signal flow graphs	486
9.2	IIR system structures	488
9.3	FIR system structures	501
9.4	Lattice structures	511
9.5	Structure conversion, simulation, and verification	519
	Learning summary	522

Contents

Terms and concepts	522
Further reading	524
Review questions	525
Problems	526
10 Design of FIR filters	537
10.1 The filter design problem	538
10.2 FIR filters with linear phase	544
10.3 Design of FIR filters by windowing	556
10.4 Design of FIR filters by frequency sampling	573
10.5 Chebyshev polynomials and minimax approximation	582
10.6 Equiripple optimum Chebyshev FIR filter design	586
10.7 Design of some special FIR filters	601
Learning summary	608
Terms and concepts	608
Further reading	610
Review questions	610
Problems	612
11 Design of IIR filters	624
11.1 Introduction to IIR filter design	625
11.2 Design of continuous-time lowpass filters	627
11.3 Transformation of continuous-time filters to discrete-time IIR filters	653
11.4 Design examples for lowpass IIR filters	668
11.5 Frequency transformations of lowpass filters	673
11.6 Design examples of IIR filters using MATLAB	680
Learning summary	687
Terms and concepts	687
Further reading	689
Review questions	689
Problems	691
12 Multirate signal processing	705
12.1 Sampling rate conversion	706
12.2 Implementation of multirate systems	727
12.3 Filter design for multirate systems	736
12.4 Two-channel filter banks	746
12.5 Multichannel filter banks	759
Learning summary	764
Terms and concepts	764
Further reading	766
Review questions	766
Problems	768

13	Random signals	777
	13.1 Probability models and random variables	778
	13.2 Jointly distributed random variables	786
	13.3 Covariance, correlation, and linear estimation	792
	13.4 Random processes	796
	13.5 Some useful random process models	809
	Learning summary	815
	Terms and concepts	816
	Further reading	818
	Review questions	818
	Problems	820
14	Random signal processing	829
	14.1 Estimation of mean, variance, and covariance	830
	14.2 Spectral analysis of stationary processes	834
	14.3 Optimum linear filters	858
	14.4 Linear prediction and all-pole signal modeling	866
	14.5 Optimum orthogonal transforms	877
	Learning summary	884
	Terms and concepts	885
	Further reading	886
	Review questions	887
	Problems	888
15	Finite wordlength effects	902
	15.1 Number representation	903
	15.2 Statistical analysis of quantization error	909
	15.3 Oversampling A/D and D/A conversion	919
	15.4 Quantization of filter coefficients	928
	15.5 Effects of finite wordlength on digital filters	936
	15.6 Finite wordlength effects in FFT algorithms	950
	Learning summary	952
	Terms and concepts	953
	Further reading	954
	Review questions	955
	Problems	956
	References	968
	Index	977

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PREFACE

During the last three decades Digital Signal Processing (DSP) has evolved into a core area of study in electrical and computer engineering. Today, DSP provides the methodology and algorithms for the solution of a continuously growing number of practical problems in scientific, engineering, and multimedia applications.

Despite the existence of a number of excellent textbooks focusing either on the theory of DSP or on the application of DSP algorithms using interactive software packages, we feel there is a strong need for a book bridging the two approaches by combining the best of both worlds. This was our motivation for writing this book, that is, to help students and practicing engineers understand the fundamental mathematical principles underlying the operation of a DSP method, appreciate its practical limitations, and grasp, with sufficient details, its practical implementation.

Objectives

The principal objective of this book is to provide a systematic introduction to the basic concepts and methodologies for digital signal processing, based whenever possible on fundamental principles. A secondary objective is to develop a foundation that can be used by students, researchers, and practicing engineers as the basis for further study and research in this field. To achieve these objectives, we have focused on material that is fundamental and where the scope of application is not limited to the solution of specialized problems, that is, material that has a broad scope of application. Our aim is to help the student develop sufficient intuition as to how a DSP technique works, be able to apply the technique, and be capable of interpreting the results of the application. We believe this approach will also help students to become intelligent users of DSP techniques and good critics of DSP techniques performed by others.

Pedagogical philosophy

Our experience in teaching undergraduate and graduate courses in digital signal processing has reaffirmed the belief that the ideal blend of simplified mathematical analysis and computer-based reasoning and simulations enhances both the teaching and the learning of digital signal processing. To achieve these objectives, we have used mathematics to support underlying intuition rather than as a substitute for it, and we have emphasized practicality without turning the book into a simplistic “cookbook.” The purpose of MATLAB[®] code integrated with the text is to illustrate the implementation of core signal processing algorithms; therefore, we use standard language commands and functions that have remained relatively stable during the most recent releases. We also believe that in-depth

Preface

understanding and full appreciation of DSP is not possible without familiarity with the fundamentals of continuous-time signals and systems. To help the reader grasp the full potential of DSP theory and its application to practical problems, which primarily involve continuous-time signals, we have integrated relevant continuous-time background into the text. This material can be quickly reviewed or skipped by readers already exposed to the theory of continuous-time signals and systems. Another advantage of this approach is that some concepts are easier to explain and analyze in continuous-time than in discrete-time or vice versa.

Instructional aids

We have put in a considerable amount of effort to produce instructional aids that enhance both the teaching and learning of DSP. These aids, which constitute an integral part of the textbook, include:

- **Figures** The graphical illustrations in each figure are designed to provide a mental picture of how each method works or to demonstrate the performance of a specific DSP method.
- **Examples** A large number of examples are provided, many generated by MATLAB[®] to reflect realistic cases, which illustrate important concepts and guide the reader to easily implement various methods.
- **MATLAB[®] functions and scripts** To help the reader apply the various algorithms and models to real-world problems, we provide MATLAB[®] functions for all major algorithms along with examples illustrating their use.
- **Learning summaries** At the end of each chapter, these provide a review of the basic yet important concepts discussed in that chapter in the form of a bullet point list.
- **Review questions** Conceptual questions are provided at the end of each chapter to reinforce the theory, clarify important concepts, and help relate theory to applications.
- **Terms and concepts** Important phrases and notions introduced in the chapter are again explained in a concise manner for a quick overview.
- **Problems** A large number of problems, ranging from simple applications of theory and computations to more advanced analysis and design tasks, have been developed for each chapter. These problems are organized in up to four sections. The first set of problems termed as Tutorial Problems contains problems whose solutions are available on the website. The next section, Basic Problems, belongs to problems with answers available on the website. The third section, Assessment Problems, contains problems based on topics discussed in the chapter. Finally, the last section, Review Problems, introduces applications, review, or extension problems.
- **Book website** This website will contain additional in-depth material, signal datasets, MATLAB[®] functions, power-point slides with all figures in the book, etc., for those who want to delve intensely into topics. This site will be constantly updated. It will also provide tutorials that support readers who need a review of background material.
- **Solutions manual** This manual, which contains solutions for all problems in the text, is available to instructors from the publisher.

Preface

Audience and prerequisites

The book is primarily aimed as a textbook for upper-level undergraduate and for first-year graduate students in electrical and computer engineering. However, researchers, engineers, and industry practitioners can use the book to learn how to analyze or process data for scientific or engineering applications. The mathematical complexity has been kept at a level suitable for seniors and first-year graduate students in almost any technical discipline. More specifically, the reader should have a background in calculus, complex numbers and variables, and the basics of linear algebra (vectors, matrices, and their manipulation).

Course configurations

The material covered in this text is intended for teaching to upper-level undergraduate or first-year graduate students. However, it can be used flexibly for the preparation of a number of courses. The first six chapters can be used in a junior level signals and systems course with emphasis on discrete-time. The first 11 chapters can be used in a typical one-semester undergraduate or graduate DSP course in which the first six chapters are reviewed and the remaining five chapters are emphasized. Finally, an advanced graduate level course on modern signal processing can be taught by combining some appropriate material from the first 11 chapters and emphasizing the last four chapters. The pedagogical coverage of the material also lends itself to a well-rounded graduate level course in DSP by choosing selected topics from all chapters.

Feedback

Experience has taught us that errors – typos or just plain mistakes – are an inescapable byproduct of any textbook writing endeavor. We apologize in advance for any errors you may find and we urge you to bring them or additional feedback to our attention at vingle@ece.neu.edu

Acknowledgments

We wish to express our sincere appreciation to the many individuals who have helped us with their constructive comments and suggestions. Special thanks go to Sidi Niu for the preparation of the *Solutions Manual*. Phil Meyler persuaded us to choose Cambridge University Press as our publisher, and we have been happy with that decision. We are grateful to Phil for his enthusiasm and his influence in shaping the scope and the objectives of our book. The fine team at CUP, including Catherine Flack, Chris Miller, and Richard Smith, has made the publication of this book an exciting and pleasant experience. Finally, we express our deepest thanks to our wives, Anna and Usha, for their saintly understanding and patience.

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