

# Digital Signal Processing

K. Deergha Rao · M. N. S. Swamy

# Digital Signal Processing

Theory and Practice

 Springer

K. Deergha Rao  
Department of Electronics  
and Communication Engineering  
Vasavi College of Engineering  
(affiliated to Osmania University)  
Hyderabad, Telangana  
India

M. N. S. Swamy  
Department of Electrical and Computer  
Engineering  
Concordia University  
Montreal, QC  
Canada

ISBN 978-981-10-8080-7                      ISBN 978-981-10-8081-4 (eBook)  
<https://doi.org/10.1007/978-981-10-8081-4>

Library of Congress Control Number: 2018931504

© Springer Nature Singapore Pte Ltd. 2018

This work is subject to copyright. All rights are reserved by the Publisher, whether the whole or part of the material is concerned, specifically the rights of translation, reprinting, reuse of illustrations, recitation, broadcasting, reproduction on microfilms or in any other physical way, and transmission or information storage and retrieval, electronic adaptation, computer software, or by similar or dissimilar methodology now known or hereafter developed.

The use of general descriptive names, registered names, trademarks, service marks, etc. in this publication does not imply, even in the absence of a specific statement, that such names are exempt from the relevant protective laws and regulations and therefore free for general use.

The publisher, the authors and the editors are safe to assume that the advice and information in this book are believed to be true and accurate at the date of publication. Neither the publisher nor the authors or the editors give a warranty, express or implied, with respect to the material contained herein or for any errors or omissions that may have been made. The publisher remains neutral with regard to jurisdictional claims in published maps and institutional affiliations.

Printed on acid-free paper

This Springer imprint is published by the registered company Springer Nature Singapore Pte Ltd. part of Springer Nature  
The registered company address is: 152 Beach Road, #21-01/04 Gateway East, Singapore 189721, Singapore

असतोमा सदगमय  
तमसोमा ज्योतिर्गमय  
मृत्योर्मा अमृतं गमय

Lead me from the unreal to the real  
Lead me from darkness to enlightenment  
Lead me from death to immortality

*To*

*My parents Dalamma and Boddu,*

*My beloved wife Sarojini,*

*and my beloved teacher Prof. D. C. Reddy*

*K. Deerga Rao*

*To*

*My parents, teachers*

*and my beloved wife Leela.*

*M. N. S. Swamy*

# Preface

Digital signal processing (DSP) is now a core subject in electronics, communications, and computer engineering curricula. The motivation in writing this book is to include modern topics of increasing importance not included in the textbooks available on the subject of digital signal processing and also to provide a comprehensive exposition of all aspects of digital signal processing. The text is integrated with MATLAB-based programs to enhance the understanding of the underlying theories of the subject.

This book is written at a level suitable for undergraduate and master students as well as for self-study by researchers, practicing engineers, and scientists. Depending on the chapters chosen, this text can be used for teaching a one- or two-semester course, for example, introduction to digital signal processing, multirate digital signal processing, multirate and wavelet signal processing, digital filters design and implementation.

In this book, many illustrative examples are included in each chapter for easy understanding of the DSP concepts. An attractive feature of this book is the inclusion of MATLAB-based examples with codes to encourage readers to implement on their personal computers to become confident of the fundamentals and to gain more insight into digital signal processing. In addition to the problems that require analytical solutions, problems that require solutions using MATLAB are introduced to the reader at the end of each chapter. Another attractive feature of this book is that many real-life signal processing design problems are introduced to the reader by the use of MATLAB and programmable DSP processors. This book also introduces three chapters of growing interest not normally found in an upper division text on digital signal processing. In less than 20 years, wavelets have emerged as a powerful mathematical tool for signal and image processing. In this textbook, we have introduced a chapter on wavelets, wherein we have tried to make it easy for readers to understand the wavelets from basics to applications. Another chapter is introduced on adaptive digital filters used in the signal processing problems for faster and acceptable results in the presence of changing environments and changing system requirements. The last chapter included in this book is on DSP processors, which is a growing topic of interest in digital signal processing.

This book is divided into 13 chapters. Chapter 1 presents an introduction to digital signal processing with typical examples of digital signal processing applications. Chapter 2 discusses the time-domain representation of discrete-time signals and systems, linear time-invariant (LTI) discrete-time systems and their properties, characterization of discrete-time systems, representation of discrete-time signals and systems in frequency domain, representation of sampling in frequency domain, reconstruction of a bandlimited signal from its samples, correlation of discrete-time signals, and discrete-time random signals. Chapter 3 deals with z-transform and analysis of LTI discrete-time systems. In Chap. 4, discrete Fourier transform (DFT), its properties, and fast Fourier transform (FFT) are discussed. Chapter 5 deals with analog filter approximations and IIR filter design methodologies. Chapter 6 discusses FIR filter design methodologies. Chapter 7 covers various structures such as direct form I & II, cascade, parallel, and lattice structures for the realization of FIR and IIR digital filters. The finite word length effects on these structures are also analyzed. Chapters 8 and 9 provide an in-depth study of the multirate signal processing concepts and design of multirate filter banks. A deeper understanding of Chaps. 8 and 9 is required for a thorough understanding of the discrete wavelet transforms discussed in Chap. 10. The principle of adaptive digital filters and their applications are presented in Chap. 11. Chapter 12 deals with the estimation of spectra from finite duration observations of the signal using both parametric and nonparametric methods. Programmable DSP processors are discussed in Chap. 13.

The salient features of this book are as follows.

- Provides comprehensive exposure to all aspects of DSP with clarity and in an easy way to understand.
- Provides an understanding of the fundamentals, design, implementation, and applications of DSP.
- DSP techniques and concepts are illustrated with several fully worked numerical examples.
- Provides complete design examples and practical implementation details such as assembly language and C language programs for DSP processors.
- Provides MATLAB implementation of many concepts:
  - Digital FIR and IIR filter design
  - Finite word length effects analysis
  - Discrete Fourier transform
  - Fast Fourier transform
  - $z$ -Transform
  - Multirate analysis
  - Filter banks
  - Discrete wavelet transform
  - Adaptive filters

- Power spectral estimation
- Design of digital filters using MATLAB graphical user interface (GUI) filter designer SPTOOL
- Provides examples of important concepts and to reinforce the knowledge gained.

Hyderabad, India  
Montreal, Canada

K. Deergha Rao  
M. N. S. Swamy

# Contents

<b>1</b>	<b>Introduction</b> . . . . .	1
1.1	What Is Digital Signal Processing? . . . . .	1
1.2	Why Digital Signal Processing? . . . . .	1
1.3	Typical Signal Processing Operations . . . . .	3
1.3.1	Elementary Time-Domain Operations . . . . .	3
1.3.2	Correlation . . . . .	4
1.3.3	Digital Filtering . . . . .	4
1.3.4	Modulation and Demodulation . . . . .	4
1.3.5	Discrete Transformation . . . . .	4
1.3.6	Quadrature Amplitude Modulation . . . . .	5
1.3.7	Multiplexing and Demultiplexing . . . . .	5
1.4	Application Areas of DSP . . . . .	5
1.5	Some Application Examples of DSP . . . . .	6
1.5.1	Telecommunications . . . . .	6
1.5.2	Noise Reduction in Speech Signals . . . . .	7
1.5.3	ECG Signal Processing . . . . .	8
1.5.4	Audio Processing . . . . .	8
1.5.5	Image Processing . . . . .	9
1.5.6	GPS Signal Processing . . . . .	11
1.5.7	Genomic Signal Processing . . . . .	13
1.6	Scope and Outline . . . . .	15
	References . . . . .	17
<b>2</b>	<b>Discrete-Time Signals and Systems</b> . . . . .	19
2.1	Discrete-Time Signals . . . . .	19
2.1.1	Elementary Operations on Sequences . . . . .	21
2.1.2	Basic Sequences . . . . .	23
2.1.3	Arbitrary Sequence . . . . .	25



2.2	Classification of Discrete-Time Signals . . . . .	25
2.2.1	Symmetric and AntiSymmetric Signals . . . . .	25
2.2.2	Finite and Infinite Length Sequences . . . . .	26
2.2.3	Right-Sided and Left-Sided Sequences . . . . .	26
2.2.4	Periodic and Aperiodic Signals . . . . .	27
2.2.5	Energy and Power Signals . . . . .	27
2.3	The Sampling Process of Analog Signals . . . . .	30
2.3.1	Impulse-Train Sampling . . . . .	30
2.3.2	Sampling with a Zero-Order Hold . . . . .	31
2.3.3	Quantization and Coding . . . . .	33
2.4	Discrete-Time Systems . . . . .	35
2.4.1	Classification of Discrete-Time Systems . . . . .	35
2.4.2	Impulse and Step Responses . . . . .	39
2.5	Linear Time-Invariant Discrete-Time Systems . . . . .	40
2.5.1	Input–Output Relationship . . . . .	40
2.5.2	Computation of Linear Convolution . . . . .	42
2.5.3	Computation of Convolution Sum Using MATLAB . . . . .	44
2.5.4	Some Properties of the Convolution Sum . . . . .	45
2.5.5	Stability and Causality of LTI Systems in Terms of the Impulse Response . . . . .	48
2.6	Characterization of Discrete-Time Systems . . . . .	50
2.6.1	Non-recursive Difference Equation . . . . .	50
2.6.2	Recursive Difference Equation . . . . .	51
2.6.3	Solution of Difference Equations . . . . .	52
2.6.4	Computation of Impulse and Step Responses Using MATLAB . . . . .	57
2.7	Representation of Discrete-Time Signals and Systems in Frequency Domain . . . . .	58
2.7.1	Fourier Transform of Discrete-Time Signals . . . . .	58
2.7.2	Theorems on DTFT . . . . .	59
2.7.3	Some Properties of the DTFT of a Complex Sequence $x(n)$ . . . . .	61
2.7.4	Some Properties of the DTFT of a Real Sequence $x(n)$ . . . . .	64
2.8	Frequency Response of Discrete-Time Systems . . . . .	74
2.8.1	Frequency Response Computation Using MATLAB . . . . .	80
2.9	Representation of Sampling in Frequency Domain . . . . .	84
2.9.1	Sampling of Lowpass Signals . . . . .	86
2.10	Reconstruction of a Band-Limited Signal from Its Samples . . . . .	88

2.11	Discrete-Time Random Signals . . . . .	91
2.11.1	Statistical Properties of Discrete-Time Random Signals . . . . .	91
2.11.2	Power of White Noise Input . . . . .	92
2.11.3	Statistical Properties of LTI System Output for White Noise Input . . . . .	93
2.11.4	Correlation of Discrete-Time Signals . . . . .	95
2.12	Problems . . . . .	97
2.13	MATLAB Exercises . . . . .	99
<b>3</b>	<b>The <math>z</math>-Transform and Analysis of LTI Systems in the Transform Domain . . . . .</b>	<b>101</b>
3.1	Definition of the $z$ -Transform . . . . .	101
3.2	Properties of the Region of Convergence for the $z$ -Transform . . . . .	104
3.3	Properties of the $z$ -Transform . . . . .	108
3.4	$z$ -Transforms of Some Commonly Used Sequences . . . . .	115
3.5	The Inverse $z$ -Transform . . . . .	120
3.5.1	Modulation Theorem in the $z$ -Domain . . . . .	121
3.5.2	Parseval's Relation in the $z$ -Domain . . . . .	122
3.6	Methods for Computation of the Inverse $z$ -Transform . . . . .	124
3.6.1	Cauchy's Residue Theorem for Computation of the Inverse $z$ -Transform . . . . .	124
3.6.2	Computation of the Inverse $z$ -Transform Using the Partial Fraction Expansion . . . . .	125
3.6.3	Inverse $z$ -Transform by Partial Fraction Expansion Using MATLAB . . . . .	129
3.6.4	Computation of the Inverse $z$ -Transform Using the Power Series Expansion . . . . .	130
3.6.5	Inverse $z$ -Transform via Power Series Expansion Using MATLAB . . . . .	134
3.6.6	Solution of Difference Equations Using the $z$ -Transform . . . . .	134
3.7	Analysis of Discrete-Time LTI Systems in the $z$ -Transform Domain . . . . .	136
3.7.1	Rational or IIR Transfer Function . . . . .	137
3.7.2	FIR Transfer Function . . . . .	138
3.7.3	Poles and Zeros of a Rational Transfer Function . . . . .	138
3.7.4	Frequency Response from Poles and Zeros . . . . .	139
3.7.5	Stability and Causality . . . . .	141
3.7.6	Minimum-Phase, Maximum-Phase, and Mixed-Phase Systems . . . . .	146
3.7.7	Inverse System . . . . .	147

3.7.8	Allpass System . . . . .	149
3.7.9	Allpass and Minimum-Phase Decomposition . . . . .	152
3.8	One-Sided $z$ -Transform . . . . .	154
3.8.1	Solution of Difference Equations with Initial Conditions . . . . .	157
3.9	Problems . . . . .	158
3.10	MATLAB Exercises . . . . .	160
	Reference . . . . .	161
<b>4</b>	<b>The Discrete Fourier Transform . . . . .</b>	<b>163</b>
4.1	The Discrete-Time Fourier Series . . . . .	163
4.1.1	Periodic Convolution . . . . .	165
4.2	The Discrete Fourier Transform . . . . .	167
4.2.1	Circular Operations on a Finite Length Sequence . . . . .	173
4.3	Basic Properties of the Discrete Fourier Transform . . . . .	176
4.4	Symmetry Relations of DFT . . . . .	181
4.4.1	Symmetry Relations of DFT of Complex-Valued Sequences . . . . .	181
4.4.2	Symmetry Relations of DFT of Real-Valued Sequences . . . . .	184
4.4.3	DFTs of Two Real Sequences from a Single N-Point DFT . . . . .	187
4.5	Computation of Circular Convolution . . . . .	188
4.5.1	Circulant Matrix Method . . . . .	188
4.5.2	Graphical Method . . . . .	188
4.5.3	DFT Approach . . . . .	189
4.6	Linear Convolution Using DFT . . . . .	190
4.6.1	Linear Convolution of Two Finite Length Sequences . . . . .	190
4.6.2	Linear Convolution of a Finite Length Sequence with a Long Duration Sequence . . . . .	192
4.7	Fast Fourier Transform . . . . .	197
4.7.1	Decimation-in-Time FFT Algorithm with Radix-2 . . . . .	198
4.7.2	In-Place Computation . . . . .	209
4.7.3	Decimation-in-Frequency FFT Algorithm with Radix-2 . . . . .	210
4.7.4	Radix-4 DIF FFT Algorithm . . . . .	217
4.8	Comparison of Computational Complexity . . . . .	220
4.9	DFT Computation Using the Goertzel Algorithm and the Chirp Transform . . . . .	221
4.9.1	The Goertzel Algorithm . . . . .	221
4.9.2	The Chirp Transform Algorithm . . . . .	224

- 4.10 Decimation-in-Time FFT Algorithm for a Composite Number . . . . . 225
- 4.11 The Inverse Discrete Fourier Transform . . . . . 227
- 4.12 Computation of DFT and IDFT Using MATLAB . . . . . 230
- 4.13 Application Examples . . . . . 231
  - 4.13.1 Detection of Signals Buried in Noise . . . . . 231
  - 4.13.2 Denoising of a Speech Signal . . . . . 232
  - 4.13.3 DTMF Tone Detection Using Goertzel Algorithm. . . . . 233
- 4.14 Problems . . . . . 239
- 4.15 MATLAB Exercises . . . . . 240
- References . . . . . 240
- 5 IIR Digital Filter Design . . . . . 241**
  - 5.1 Analog Lowpass Filter Design . . . . . 241
    - 5.1.1 Filter Specifications . . . . . 242
    - 5.1.2 Butterworth Analog Lowpass Filter . . . . . 242
    - 5.1.3 Chebyshev Analog Lowpass Filter . . . . . 247
    - 5.1.4 Elliptic Analog Lowpass Filter . . . . . 255
    - 5.1.5 Bessel Filter . . . . . 258
    - 5.1.6 Comparison of Various Types of Analog Filters . . . . . 258
    - 5.1.7 Design of Analog Highpass, Bandpass, and Bandstop Filters . . . . . 261
  - 5.2 Design of Digital Filters from Analog Filters . . . . . 267
    - 5.2.1 Digital Filter Specifications . . . . . 267
    - 5.2.2 Design of Digital Filters Using Impulse-Invariant Method . . . . . 268
    - 5.2.3 Design of Digital Filters Using Bilinear Transformation . . . . . 271
  - 5.3 Design of Digital Filters Using Digital-to-Digital Transformations . . . . . 291
  - 5.4 Design of IIR Digital Filters Using MATLAB . . . . . 295
  - 5.5 Design of IIR Filters Using MATLAB GUI Filter Designer SPTOOL . . . . . 303
  - 5.6 Design of Specialized Digital Filters by Pole-Zero Placement. . . . . 304
    - 5.6.1 Notch Filter . . . . . 304
    - 5.6.2 Comb Filter . . . . . 308
  - 5.7 Some Examples of IIR Filters for Audio Processing Applications . . . . . 310
    - 5.7.1 Suppression of Power Supply Hum in Audio Signals . . . . . 310
    - 5.7.2 Generation of Artificial Reverberations. . . . . 312

5.7.3	Audio Peaking Equalizers . . . . .	314
5.7.4	Generation and Detection of DTMF Tones . . . . .	316
5.8	Problems . . . . .	321
5.9	MATLAB Exercises . . . . .	322
	References . . . . .	324
<b>6</b>	<b>FIR Digital Filter Design . . . . .</b>	<b>325</b>
6.1	Ideal Impulse Response of FIR Filters . . . . .	326
6.2	Linear Phase FIR Filters . . . . .	329
6.2.1	Types of Linear Phase FIR Transfer Functions . . . . .	331
6.2.2	Zero Locations of Linear Phase FIR Transfer Functions . . . . .	335
6.3	FIR Filter Design Using Windowing Method . . . . .	338
6.3.1	Gibb's Oscillations . . . . .	340
6.3.2	Fixed Window Functions . . . . .	341
6.3.3	Comparison of the Fixed Windows . . . . .	343
6.3.4	Design of FIR Filters Using Fixed Windows . . . . .	348
6.3.5	Kaiser Window . . . . .	354
6.3.6	Design Procedure for Linear Phase FIR Filter Using Kaiser Window . . . . .	355
6.4	FIR Differentiator Design . . . . .	366
6.5	Hilbert Transformer . . . . .	369
6.6	Kaiser Window-Based Linear Phase FIR Filter Design Using MAT LAB . . . . .	370
6.7	Design of Linear Phase FIR Filters Using the Frequency Sampling Method . . . . .	373
6.8	Design of Optimal Linear Phase FIR Filters . . . . .	375
6.8.1	Optimal (Equiripple) Linear Phase FIR Filter Design Using MATLAB . . . . .	385
6.9	Design of Minimum-Phase FIR Filters . . . . .	389
6.9.1	Design of Minimum-Phase FIR Filters Using MATLAB . . . . .	392
6.10	Design of FIR Differentiator and Hilbert Transformer Using MATLAB . . . . .	393
6.11	Linear Phase FIR Filter Design Using MATLAB GUI Filter Designer SPTOOL . . . . .	396
6.12	Effect of FIR Filtering on Voice Signals . . . . .	398
6.12.1	FIR Filters to Recover Voice Signals Contaminated with Sinusoidal Interference . . . . .	403
6.13	Design of Two-Band Digital Crossover Using FIR Filters . . . . .	408
6.14	Comparison Between FIR and IIR Filters . . . . .	410

6.15	Problems	410
6.16	MATLAB Exercises	412
	References	414
<b>7</b>	<b>Structures for Digital Filter Realization and Effects of Finite Word Length</b>	<b>415</b>
7.1	Signal Flow Graphs and Block Diagram Representation	415
7.1.1	Transposition	417
7.2	IIR Filter Structures	420
7.2.1	Direct Form I Structure	420
7.2.2	Direct Form II Structure	421
7.2.3	Cascade Structure	422
7.2.4	Parallel Form Structure	424
7.2.5	Lattice Structure of an All-Pole System	426
7.2.6	Gray–Markel’s Lattice Structure for a General IIR Filter	431
7.3	Realization of IIR Structures Using MATLAB	434
7.4	FIR Filter Structures	440
7.4.1	Direct Form Realization	440
7.4.2	Cascade Realization	442
7.4.3	Linear Phase FIR Structures	443
7.4.4	FIR Lattice Structures	445
7.4.5	Realization of FIR Lattice Structure Using MATLAB	448
7.5	Effect of Finite Word Length	449
7.5.1	Number Representation	449
7.5.2	Effect of Quantization	454
7.5.3	Fixed-Point Number Quantization	454
7.5.4	Quantization of Filter Coefficients	458
7.5.5	Approach for Determining Optimum Coefficient Word Length	469
7.5.6	Input-Quantization Errors	470
7.5.7	Effect of Product Quantization	475
7.6	Scaling in Fixed-Point Realization of IIR Digital Filters	486
7.6.1	Scaling for a Second-Order Filter	489
7.6.2	Scaling in a Parallel Structure	491
7.6.3	Scaling in a Cascade Structure	492
7.6.4	Pole-Zero Pairing and Ordering of the Cascade Form	496
7.7	Limit Cycles in IIR Digital Filters	499
7.7.1	Limit Cycles Due to Round-off and Truncation of Products	499
7.7.2	Overflow Limit Cycles	503

- 7.8 Quantization Effect in FFT Computation . . . . . 505
  - 7.8.1 Direct Computation of the DFT . . . . . 505
  - 7.8.2 FFT Computation . . . . . 506
- 7.9 Problems . . . . . 508
- 7.10 MATLAB Exercises . . . . . 510
- References . . . . . 512
- 8 Basics of Multirate Digital Signal Processing . . . . . 513**
  - 8.1 Advantages of Multirate Signal Processing . . . . . 513
  - 8.2 Multirate Signal Processing Concepts . . . . . 514
    - 8.2.1 Down-Sampling: Decimation by an Integer Factor . . . . . 514
    - 8.2.2 Up Sampling: Interpolation by an Integer Factor . . . . . 523
    - 8.2.3 Changing the Sampling Rate by a Non-integer Factor . . . . . 530
    - 8.2.4 Sampling Rate Conversion Via Multistage Approach . . . . . 533
  - 8.3 Practical Sampling Rate Converter Design . . . . . 533
    - 8.3.1 Overall Filter Specifications . . . . . 534
    - 8.3.2 Filter Requirements for Individual Stages . . . . . 534
    - 8.3.3 Illustrative Design Examples . . . . . 535
  - 8.4 Polyphase Decomposition . . . . . 546
    - 8.4.1 Structures for Decimators and Interpolators . . . . . 548
  - 8.5 Resolution Analysis of Oversampling ADC . . . . . 552
    - 8.5.1 Reduction of ADC Quantization Noise by Oversampling . . . . . 552
    - 8.5.2 Sigma-Delta Modulation ADC . . . . . 555
  - 8.6 Design of Multirate Bandpass and Bandstop Filters . . . . . 559
  - 8.7 Application Examples . . . . . 568
    - 8.7.1 Digital Audio System . . . . . 568
    - 8.7.2 Compact Disk Player . . . . . 569
  - 8.8 Problems . . . . . 570
  - 8.9 MATLAB Exercises . . . . . 572
  - References . . . . . 573
- 9 Multirate Filter Banks . . . . . 575**
  - 9.1 Uniform DFT Filter Banks . . . . . 575
    - 9.1.1  $L$ th-Band Filter . . . . . 576
    - 9.1.2 Design of Linear Phase  $L$ th-Band FIR Filter . . . . . 577
  - 9.2 Polyphase Implementations of Uniform Filter Banks . . . . . 579
  - 9.3 Two-Channel Quadrature Mirror Filter (QMF) Bank . . . . . 582
    - 9.3.1 The Filter Bank Structure . . . . . 582
    - 9.3.2 Analysis of Two-Channel QMF Bank . . . . . 583

9.3.3	Alias-Free and Perfect Reconstruction for $M$ -Channel QMF Bank . . . . .	586
9.3.4	Polyphase Representation of $M$ -Channel QMF Banks . . . . .	588
9.3.5	Conditions for Existence of FIR Analysis/Synthesis Filters for Perfect Reconstruction . . . . .	594
9.4	Methods for Designing Linear Phase FIR PR QMF Banks . . .	599
9.4.1	Johnston Method . . . . .	599
9.4.2	Linear Phase FIR PR QMF Banks with Lattice Structures . . . . .	602
9.4.3	Design of Perfect Reconstruction Two-Channel FIR Filter Bank Using MATLAB . . . . .	605
9.5	Tree-Structured Filter Banks . . . . .	606
9.5.1	Maximally Decimated Filter Banks . . . . .	606
9.5.2	Tree-Structured Filter Banks with Equal Passband Width . . . . .	607
9.5.3	Tree-Structured Filter Banks with Unequal Passband Widths . . . . .	610
9.6	Application Examples . . . . .	613
9.6.1	Transmultiplexers . . . . .	613
9.6.2	Subband Coding of Speech Signals . . . . .	614
9.6.3	Analog Voice Privacy System . . . . .	614
9.7	Problems . . . . .	615
9.8	MATLAB Exercises . . . . .	617
	References . . . . .	617
<b>10</b>	<b>Discrete Wavelet Transforms . . . . .</b>	<b>619</b>
10.1	Time–Frequency Representation of Signals . . . . .	620
10.2	Short-Time Fourier Transform (STFT) . . . . .	621
10.2.1	Inverse STFT . . . . .	623
10.3	Scaling Functions and Wavelets . . . . .	624
10.3.1	Expansion of a Signal in Series Form . . . . .	624
10.3.2	Scaling Functions . . . . .	625
10.3.3	Wavelet Functions . . . . .	626
10.3.4	Dilation Equations . . . . .	628
10.4	The Discrete Wavelet Transform (DWT) . . . . .	630
10.4.1	Computation of Wavelet Coefficients . . . . .	634
10.5	Multiresolution Analysis . . . . .	636
10.6	Wavelet Reconstruction . . . . .	636
10.7	Required Properties of Wavelets . . . . .	637
10.7.1	Orthogonality . . . . .	637
10.7.2	Regularity Condition . . . . .	640



10.8	Generation of Daubechies Orthonormal Scaling Functions and Wavelets . . . . .	642
10.8.1	Relation Between Decomposition Filters ( $H_0, H_1$ ) and Reconstruction Filters ( $G_0, G_1$ ) for Perfect Reconstruction . . . . .	642
10.8.2	Daubechies Wavelet Filter Coefficients . . . . .	644
10.8.3	Generation of Daubechies Scaling Functions and Wavelets Using MATLAB . . . . .	651
10.9	Biorthogonal Scaling Functions and Wavelets Generation . . . . .	653
10.9.1	Biorthogonal Wavelet Filter Coefficients . . . . .	656
10.10	The Impact of Wavelet Properties . . . . .	660
10.11	Computation of One-Dimensional Discrete Wavelet Transform (1-D DWT) and Inverse DWT (1-D IDWT) . . . . .	660
10.12	2-D Discrete Time Wavelet Transform . . . . .	665
10.12.1	Computation of 2-D DWT and IDWT Using MATLAB . . . . .	667
10.13	Wavelet Packets . . . . .	668
10.14	Image Wavelet Packets . . . . .	673
10.15	Some Application Examples of Wavelet Transforms . . . . .	675
10.15.1	Signal Denoising . . . . .	675
10.15.2	Image Denoising . . . . .	679
10.15.3	Digital Image Water Marking . . . . .	679
10.15.4	Image Compression . . . . .	683
10.16	Problems . . . . .	684
10.17	MATLAB Exercises . . . . .	685
	References . . . . .	690
<b>11</b>	<b>Adaptive Digital Filters . . . . .</b>	<b>693</b>
11.1	Adaptive Digital Filter Principle . . . . .	693
11.2	Adaptive Filter Structures . . . . .	694
11.2.1	Adaptive FIR Filter Structure . . . . .	694
11.2.2	Adaptive IIR Filter Structure . . . . .	695
11.3	Cost Functions . . . . .	695
11.3.1	The Mean Square Error (MSE) Cost Function . . . . .	695
11.3.2	The Exponentially Weighted Least Square (WLS) Error Function . . . . .	696
11.4	Algorithms for Adaptive Filters . . . . .	696
11.4.1	The LMS Algorithm . . . . .	696
11.4.2	The NLMS Algorithm . . . . .	697
11.4.3	The RLS Algorithm . . . . .	698
11.5	Comparison of the LMS and RLS Algorithms . . . . .	700
11.5.1	Computational Complexity . . . . .	700
11.5.2	Rate of Convergence . . . . .	700

- 11.6 Applications of Adaptive Filters . . . . . 702
  - 11.6.1 Unknown System Identification . . . . . 702
  - 11.6.2 Adaptive Interference Canceller . . . . . 708
- 11.7 Problems . . . . . 719
- References . . . . . 719
- 12 Spectral Analysis of Signals . . . . . 721**
  - 12.1 Nonparametric Methods for Power Spectrum Estimation . . . . . 721
    - 12.1.1 Periodogram . . . . . 722
    - 12.1.2 Bartlett Method . . . . . 724
    - 12.1.3 Performance Comparison of the Nonparametric Methods . . . . . 730
  - 12.2 Parametric or Model-Based Methods for Power Spectrum Estimation . . . . . 731
    - 12.2.1 Relationships Between the Autocorrelation and the Model Parameters . . . . . 733
    - 12.2.2 Power Spectrum Estimation Based on AR Model via Yule–Walker Method . . . . . 735
    - 12.2.3 Power Spectrum Estimation Based on AR Model via Burg Method . . . . . 736
    - 12.2.4 Selection of Model Order . . . . . 740
    - 12.2.5 Power Spectrum Estimation Based on MA Model . . . . . 741
    - 12.2.6 Power Spectrum Estimation Based on ARMA Model . . . . . 742
  - 12.3 Subspace Methods for Power Spectrum Estimation . . . . . 742
    - 12.3.1 Pisarenko Harmonic Decomposition Method . . . . . 743
    - 12.3.2 Multiple Signal Classification (MUSIC) Method . . . . . 743
    - 12.3.3 Eigenvector Method . . . . . 746
  - 12.4 Spectral Analysis of Non-stationary Signals . . . . . 747
    - 12.4.1 MATLAB Exercises . . . . . 749
  - References . . . . . 751
- 13 DSP Processors . . . . . 753**
  - 13.1 Evolution of DSP Processors . . . . . 753
    - 13.1.1 Conventional DSP Processors . . . . . 753
  - 13.2 Modern DSP Processors and Its Architectures . . . . . 754
    - 13.2.1 Single-Cycle Multiply–Accumulate (MAC) Unit . . . . . 754
    - 13.2.2 Modified Bus Structures and Memory Access Schemes . . . . . 755
    - 13.2.3 On-Chip Memory . . . . . 755
    - 13.2.4 Pipelining . . . . . 755
    - 13.2.5 Parallelism . . . . . 756
    - 13.2.6 Special Addressing Modes . . . . . 756

- 13.2.7 Specialized Execution Control . . . . . 757
- 13.2.8 Streamlined I/O . . . . . 757
- 13.2.9 Specialized Instruction Sets . . . . . 757
- 13.2.10 On-Chip Peripherals . . . . . 757
- 13.3 Choice of DSP Processor . . . . . 757
- 13.4 TMS320C54xx Digital Signal Processor . . . . . 758
  - 13.4.1 Features of TMS320C54xx Digital Signal Processor . . . . . 758
  - 13.4.2 The Architecture of TMS320C54xx Digital Signal Processor . . . . . 758
  - 13.4.3 Data Addressing Modes of TMS320C54xx Digital Signal Processors . . . . . 759
- 13.5 Code Composer Studio . . . . . 761
- 13.6 TMS320C67xx Digital Signal Processor . . . . . 764
  - 13.6.1 Features of TMS320C67xx Digital Signal Processor . . . . . 764
  - 13.6.2 The Architecture of TMS320C67xx Digital Signal Processor . . . . . 764
  - 13.6.3 Design of a Digital Lowpass FIR Filter and Implementation Using TMS320C67xx DSP Processor . . . . . 767
- 13.7 TigerSHARC TS201 DSP Processor . . . . . 769
  - 13.7.1 Architectural Features of TS201 Digital Signal Processor . . . . . 770
  - 13.7.2 Addressing Modes . . . . . 772
- 13.8 Implementation Example Using TigerSHARC (TS201) DSP Processor . . . . . 775
  - 13.8.1 Creating a VisualDSP++ Project . . . . . 775
  - 13.8.2 Building the Project . . . . . 778
- Index . . . . . 783**

## About the Authors

**K. Deergha Rao** is currently a Professor in the Department of Electronics and Communication Engineering, Vasavi College of Engineering, affiliated to Osmania University, Hyderabad, India, and is Former Director and Professor at the Research and Training Unit for Navigational Electronics (NERTU), Osmania University. He was a Postdoctoral Fellow and Part-Time Professor in the Department of Electrical and Computer Engineering, Concordia University, Montreal, Canada, for 4 years. He has conducted several research projects for leading Indian organizations. His teaching areas include digital signal processing, channel coding, signals and systems, and MIMO wireless communications, and his current research focuses on wireless channel coding, OFDM wireless communication channel estimation, compressive sensing, and VLSI for communications. He has presented papers at several IEEE international conferences in the USA, Switzerland, Thailand, and Russia. He has more than 100 publications to his credit, including more than 60 in IEEE journals and conference proceedings. He is a senior member of IEEE. He was awarded the IETE K. S. Krishnan Memorial Award in 2013 for the best system-oriented paper. He has authored two books, *Channel Coding Techniques for Wireless Communications* and *Signals and Systems* (both with Springer), and co-authored the book *Digital Signal Processing*.

**M. N. S. Swamy** is a Research Professor and holds the Concordia Chair in Signal Processing in the Department of Electrical and Computer Engineering at Concordia University, Montreal, Canada, where he served as the Founding chair of the Department of Electrical Engineering from 1970 to 1977 and the Dean of Engineering and Computer Science from 1977 to 1993. He received his M.Sc. and Ph.D. degrees in Electrical Engineering from the University of Saskatchewan, Saskatoon, Canada. He was made an Honorary Professor by the National Chiao Tung University, Taiwan, in 2009, which is equivalent to an honorary doctorate at the institute. He has also been associated with the Department of Electrical Engineering, Technical University of Nova Scotia, Halifax, NS, Canada; the University of Calgary, Calgary, AB, Canada; and the Department of Mathematics, University of Saskatchewan. He has published extensively in the areas of circuits,

systems, and signal processing and holds six patents. He is a coauthor of eight books and several book chapters. He is a recipient of numerous IEEE Circuits and Systems Society Awards, including the Education Award in 2000, the Golden Jubilee Medal in 2000, and Guillemin–Cauer Best Paper Award. He is a Fellow of several professional societies including IEEE. The Indian Institute of Science (IISc) in Bengaluru, a premier research institution of India, has established a gold medal and a scholarship in his name. In 1993, he was awarded the commemorative medal for the 125th Anniversary of Canada, issued by the Governor General of Canada, in recognition of his significant contributions to Canada and the community. Recently, the journal *Circuits, Systems and Signal Processing* (CSSP) has instituted a Best Paper Award in his name.