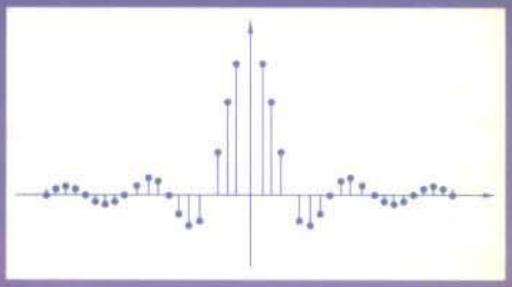
Third Edition

DIGITAL SIGNAL PROCESSING

Principles, Algorithms, and Applications



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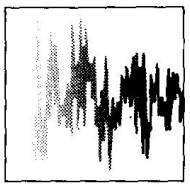

Digital Signal Processing

Principles, Algorithms, and Applications

Third Edition

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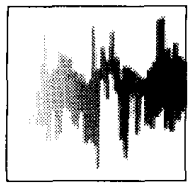

Preface

This book was developed based on our teaching of undergraduate and graduate level courses in digital signal processing over the past several years. In this book we present the fundamentals of discrete-time signals, systems, and modern digital processing algorithms and applications for students in electrical engineering, computer engineering, and computer science. The book is suitable for either a one-semester or a two-semester undergraduate level course in discrete systems and digital signal processing. It is also intended for use in a one-semester first-year graduate-level course in digital signal processing.

It is assumed that the student in electrical and computer engineering has had undergraduate courses in advanced calculus (including ordinary differential equations), and linear systems for continuous-time signals, including an introduction to the Laplace transform. Although the Fourier series and Fourier transforms of periodic and aperiodic signals are described in Chapter 4, we expect that many students may have had this material in a prior course.

A balanced coverage is provided of both theory and practical applications. A large number of well designed problems are provided to help the student in mastering the subject matter. A solutions manual is available for the benefit of the instructor and can be obtained from the publisher.

The third edition of the book covers basically the same material as the second edition, but is organized differently. The major difference is in the order in which the DFT and FFT algorithms are covered. Based on suggestions made by several reviewers, we now introduce the DFT and describe its efficient computation immediately following our treatment of Fourier analysis. This reorganization has also allowed us to eliminate repetition of some topics concerning the DFT and its applications.

In Chapter 1 we describe the operations involved in the analog-to-digital conversion of analog signals. The process of sampling a sinusoid is described in some detail and the problem of aliasing is explained. Signal quantization and digital-to-analog conversion are also described in general terms, but the analysis is presented in subsequent chapters.

Chapter 2 is devoted entirely to the characterization and analysis of linear time-invariant (shift-invariant) discrete-time systems and discrete-time signals in the time domain. The convolution sum is derived and systems are categorized according to the duration of their impulse response as a finite-duration impulse

response (FIR) and as an infinite-duration impulse response (IIR). Linear time-invariant systems characterized by difference equations are presented and the solution of difference equations with initial conditions is obtained. The chapter concludes with a treatment of discrete-time correlation.

The z-transform is introduced in Chapter 3. Both the bilateral and the unilateral z-transforms are presented, and methods for determining the inverse z-transform are described. Use of the z-transform in the analysis of linear time-invariant systems is illustrated, and important properties of systems, such as causality and stability, are related to z-domain characteristics.

Chapter 4 treats the analysis of signals and systems in the frequency domain. Fourier series and the Fourier transform are presented for both continuous-time and discrete-time signals. Linear time-invariant (LTI) discrete systems are characterized in the frequency domain by their frequency response function and their response to periodic and aperiodic signals is determined. A number of important types of discrete-time systems are described, including resonators, notch filters, comb filters, all-pass filters, and oscillators. The design of a number of simple FIR and IIR filters is also considered. In addition, the student is introduced to the concepts of minimum-phase, mixed-phase, and maximum-phase systems and to the problem of deconvolution.

The DFT, its properties and its applications, are the topics covered in Chapter 5. Two methods are described for using the DFT to perform linear filtering. The use of the DFT to perform frequency analysis of signals is also described.

Chapter 6 covers the efficient computation of the DFT. Included in this chapter are descriptions of radix-2, radix-4, and split-radix fast Fourier transform (FFT) algorithms, and applications of the FFT algorithms to the computation of convolution and correlation. The Goertzel algorithm and the chirp-z transform are introduced as two methods for computing the DFT using linear filtering.

Chapter 7 treats the realization of IIR and FIR systems. This treatment includes direct-form, cascade, parallel, lattice, and lattice-ladder realizations. The chapter includes a treatment of state-space analysis and structures for discrete-time systems, and examines quantization effects in a digital implementation of FIR and IIR systems.

Techniques for design of digital FIR and IIR filters are presented in Chapter 8. The design techniques include both direct design methods in discrete time and methods involving the conversion of analog filters into digital filters by various transformations. Also treated in this chapter is the design of FIR and IIR filters by least-squares methods.

Chapter 9 focuses on the sampling of continuous-time signals and the reconstruction of such signals from their samples. In this chapter, we derive the sampling theorem for bandpass continuous-time-signals and then cover the A/D and D/A conversion techniques, including oversampling A/D and D/A converters.

Chapter 10 provides an indepth treatment of sampling-rate conversion and its applications to multirate digital signal processing. In addition to describing decimation and interpolation by integer factors, we present a method of sampling-rate

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conversion by an arbitrary factor. Several applications to multirate signal processing are presented, including the implementation of digital filters, subband coding of speech signals, transmultiplexing, and oversampling A/D and D/A converters.

Linear prediction and optimum linear (Wiener) filters are treated in Chapter 11. Also included in this chapter are descriptions of the Levinson–Durbin algorithm and Schür algorithm for solving the normal equations, as well as the AR lattice and ARMA lattice-ladder filters.

Power spectrum estimation is the main topic of Chapter 12. Our coverage includes a description of nonparametric and model-based (parametric) methods. Also described are eigen-decomposition-based methods, including MUSIC and ESPRIT.

At Northeastern University, we have used the first six chapters of this book for a one-semester (junior level) course in discrete systems and digital signal processing.

A one-semester senior level course for students who have had prior exposure to discrete systems can use the material in Chapters 1 through 4 for a quick review and then proceed to cover Chapter 5 through 8.

In a first-year graduate level course in digital signal processing, the first five chapters provide the student with a good review of discrete-time systems. The instructor can move quickly through most of this material and then cover Chapters 6 through 9, followed by either Chapters 10 and 11 or by Chapters 11 and 12.

We have included many examples throughout the book and approximately 500 homework problems. Many of the homework problems can be solved numerically on a computer, using a software package such as MATLAB©. These problems are identified by an asterisk. Appendix D contains a list of MATLAB functions that the student can use in solving these problems. The instructor may also wish to consider the use of a supplementary book that contains computer based exercises, such as the books *Digital Signal Processing Using MATLAB* (P.W.S. Kent, 1996) by V. K. Ingle and J. G. Proakis and *Computer-Based Exercises for Signal Processing Using MATLAB* (Prentice Hall, 1994) by C. S. Burrus et al.

The authors are indebted to their many faculty colleagues who have provided valuable suggestions through reviews of the first and second editions of this book. These include Drs. W. E. Alexander, Y. Bresler, J. Deller, V. Ingle, C. Keller, H. Lev-Ari, L. Merakos, W. Mikhael, P. Monticciolo, C. Nikias, M. Schetzen, H. Trussell, S. Wilson, and M. Zoltowski. We are also indebted to Dr. R. Price for recommending the inclusion of split-radix FFT algorithms and related suggestions. Finally, we wish to acknowledge the suggestions and comments of many former graduate students, and especially those by A. L. Kok, J. Lin and S. Srinidhi who assisted in the preparation of several illustrations and the solutions manual.

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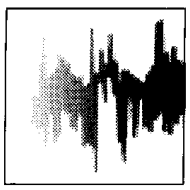

Contents

PREFACE ***xiii***

1 INTRODUCTION ***1***

1.1 Signals, Systems, and Signal Processing 2
- 1.1.1 Basic Elements of a Digital Signal Processing System, 4
- 1.1.2 Advantages of Digital over Analog Signal Processing, 5

1.2 Classification of Signals 6
- 1.2.1 Multichannel and Multidimensional Signals, 7
- 1.2.2 Continuous-Time Versus Discrete-Time Signals, 8
- 1.2.3 Continuous-Valued Versus Discrete-Valued Signals, 10
- 1.2.4 Deterministic Versus Random Signals, 11

1.3 The Concept of Frequency in Continuous-Time and Discrete-Time Signals 14
- 1.3.1 Continuous-Time Sinusoidal Signals, 14
- 1.3.2 Discrete-Time Sinusoidal Signals, 16
- 1.3.3 Harmonically Related Complex Exponentials, 19

1.4 Analog-to-Digital and Digital-to-Analog Conversion 21
- 1.4.1 Sampling of Analog Signals, 23
- 1.4.2 The Sampling Theorem, 29
- 1.4.3 Quantization of Continuous-Amplitude Signals, 33
- 1.4.4 Quantization of Sinusoidal Signals, 36
- 1.4.5 Coding of Quantized Samples, 38
- 1.4.6 Digital-to-Analog Conversion, 38
- 1.4.7 Analysis of Digital Signals and Systems Versus Discrete-Time Signals and Systems, 39

1.5 Summary and References 39

Problems 40

2 DISCRETE-TIME SIGNALS AND SYSTEMS 43

2.1 Discrete-Time Signals 43
2.1.1 Some Elementary Discrete-Time Signals, 45
2.1.2 Classification of Discrete-Time Signals, 47
2.1.3 Simple Manipulations of Discrete-Time Signals, 52

2.2 Discrete-Time Systems 56
2.2.1 Input–Output Description of Systems, 56
2.2.2 Block Diagram Representation of Discrete-Time Systems, 59
2.2.3 Classification of Discrete-Time Systems, 62
2.2.4 Interconnection of Discrete-Time Systems, 70

2.3 Analysis of Discrete-Time Linear Time-Invariant Systems 72
2.3.1 Techniques for the Analysis of Linear Systems, 72
2.3.2 Resolution of a Discrete-Time Signal into Impulses, 74
2.3.3 Response of LTI Systems to Arbitrary Inputs: The Convolution Sum, 75
2.3.4 Properties of Convolution and the Interconnection of LTI Systems, 82
2.3.5 Causal Linear Time-Invariant Systems, 86
2.3.6 Stability of Linear Time-Invariant Systems, 87
2.3.7 Systems with Finite-Duration and Infinite-Duration Impulse Response, 90

2.4 Discrete-Time Systems Described by Difference Equations 91
2.4.1 Recursive and Nonrecursive Discrete-Time Systems, 92
2.4.2 Linear Time-Invariant Systems Characterized by Constant-Coefficient Difference Equations, 95
2.4.3 Solution of Linear Constant-Coefficient Difference Equations, 100
2.4.4 The Impulse Response of a Linear Time-Invariant Recursive System, 108

2.5 Implementation of Discrete-Time Systems 111
2.5.1 Structures for the Realization of Linear Time-Invariant Systems, 111
2.5.2 Recursive and Nonrecursive Realizations of FIR Systems, 116

2.6 Correlation of Discrete-Time Signals 118
2.6.1 Crosscorrelation and Autocorrelation Sequences, 120
2.6.2 Properties of the Autocorrelation and Crosscorrelation Sequences, 122
2.6.3 Correlation of Periodic Sequences, 124
2.6.4 Computation of Correlation Sequences, 130
2.6.5 Input–Output Correlation Sequences, 131

2.7 Summary and References 134

Problems 135

3 THE Z-TRANSFORM AND ITS APPLICATION TO THE ANALYSIS OF LTI SYSTEMS 151

3.1 The z-Transform 151
3.1.1 The Direct z-Transform, 152
3.1.2 The Inverse z-Transform, 160

3.2 Properties of the z-Transform 161

3.3 Rational z-Transforms 172
3.3.1 Poles and Zeros, 172
3.3.2 Pole Location and Time-Domain Behavior for Causal Signals, 178
3.3.3 The System Function of a Linear Time-Invariant System, 181

3.4 Inversion of the z-Transform 184
3.4.1 The Inverse z-Transform by Contour Integration, 184
3.4.2 The Inverse z-Transform by Power Series Expansion, 186
3.4.3 The Inverse z-Transform by Partial-Fraction Expansion, 188
3.4.4 Decomposition of Rational z-Transforms, 195

3.5 The One-sided z-Transform 197
3.5.1 Definition and Properties, 197
3.5.2 Solution of Difference Equations, 201

3.6 Analysis of Linear Time-Invariant Systems in the z-Domain 203
3.6.1 Response of Systems with Rational System Functions, 203
3.6.2 Response of Pole–Zero Systems with Nonzero Initial Conditions, 204
3.6.3 Transient and Steady-State Responses, 206
3.6.4 Causality and Stability, 208
3.6.5 Pole–Zero Cancellations, 210
3.6.6 Multiple-Order Poles and Stability, 211
3.6.7 The Schür–Cohn Stability Test, 213
3.6.8 Stability of Second-Order Systems, 215

3.7 Summary and References 219

Problems 220

4 FREQUENCY ANALYSIS OF SIGNALS AND SYSTEMS 230

4.1 Frequency Analysis of Continuous-Time Signals 230
4.1.1 The Fourier Series for Continuous-Time Periodic Signals, 232
4.1.2 Power Density Spectrum of Periodic Signals, 235
4.1.3 The Fourier Transform for Continuous-Time Aperiodic Signals, 240
4.1.4 Energy Density Spectrum of Aperiodic Signals, 243

4.2 Frequency Analysis of Discrete-Time Signals 247
4.2.1 The Fourier Series for Discrete-Time Periodic Signals, 247

4.2.2 Power Density Spectrum of Periodic Signals, 250
4.2.3 The Fourier Transform of Discrete-Time Aperiodic Signals, 253
4.2.4 Convergence of the Fourier Transform, 256
4.2.5 Energy Density Spectrum of Aperiodic Signals, 260
4.2.6 Relationship of the Fourier Transform to the z-Transform, 264
4.2.7 The Cepstrum, 265
4.2.8 The Fourier Transform of Signals with Poles on the Unit Circle, 267
4.2.9 The Sampling Theorem Revisited, 269
4.2.10 Frequency-Domain Classification of Signals: The Concept of Bandwidth, 279
4.2.11 The Frequency Ranges of Some Natural Signals, 282
4.2.12 Physical and Mathematical Dualities, 282

4.3 Properties of the Fourier Transform for Discrete-Time Signals 286
4.3.1 Symmetry Properties of the Fourier Transform, 287
4.3.2 Fourier Transform Theorems and Properties, 294

4.4 Frequency-Domain Characteristics of Linear Time-Invariant Systems 305
4.4.1 Response to Complex Exponential and Sinusoidal Signals: The Frequency Response Function, 306
4.4.2 Steady-State and Transient Response to Sinusoidal Input Signals, 314
4.4.3 Steady-State Response to Periodic Input Signals, 315
4.4.4 Response to Aperiodic Input Signals, 316
4.4.5 Relationships Between the System Function and the Frequency Response Function, 319
4.4.6 Computation of the Frequency Response Function, 321
4.4.7 Input–Output Correlation Functions and Spectra, 325
4.4.8 Correlation Functions and Power Spectra for Random Input Signals, 327

4.5 Linear Time-Invariant Systems as Frequency-Selective Filters 330
4.5.1 Ideal Filter Characteristics, 331
4.5.2 Lowpass, Highpass, and Bandpass Filters, 333
4.5.3 Digital Resonators, 340
4.5.4 Notch Filters, 343
4.5.5 Comb Filters, 345
4.5.6 All-Pass Filters, 350
4.5.7 Digital Sinusoidal Oscillators, 352

4.6 Inverse Systems and Deconvolution 355
4.6.1 Invertibility of Linear Time-Invariant Systems, 356
4.6.2 Minimum-Phase, Maximum-Phase, and Mixed-Phase Systems, 359
4.6.3 System Identification and Deconvolution, 363
4.6.4 Homomorphic Deconvolution, 365

4.7 Summary and References 367

Problems 368

5 THE DISCRETE FOURIER TRANSFORM: ITS PROPERTIES AND APPLICATIONS 394

5.1 Frequency Domain Sampling: The Discrete Fourier Transform 394
- 5.1.1 Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals, 394
- 5.1.2 The Discrete Fourier Transform (DFT), 399
- 5.1.3 The DFT as a Linear Transformation, 403
- 5.1.4 Relationship of the DFT to Other Transforms, 407

5.2 Properties of the DFT 409
- 5.2.1 Periodicity, Linearity, and Symmetry Properties, 410
- 5.2.2 Multiplication of Two DFTs and Circular Convolution, 415
- 5.2.3 Additional DFT Properties, 421

5.3 Linear Filtering Methods Based on the DFT 425
- 5.3.1 Use of the DFT in Linear Filtering, 426
- 5.3.2 Filtering of Long Data Sequences, 430

5.4 Frequency Analysis of Signals Using the DFT 433

5.5 Summary and References 440

Problems 440

6 EFFICIENT COMPUTATION OF THE DFT: FAST FOURIER TRANSFORM ALGORITHMS 448

6.1 Efficient Computation of the DFT: FFT Algorithms 448
- 6.1.1 Direct Computation of the DFT, 449
- 6.1.2 Divide-and-Conquer Approach to Computation of the DFT, 450
- 6.1.3 Radix-2 FFT Algorithms, 456
- 6.1.4 Radix-4 FFT Algorithms, 465
- 6.1.5 Split-Radix FFT Algorithms, 470
- 6.1.6 Implementation of FFT Algorithms, 473

6.2 Applications of FFT Algorithms 475
- 6.2.1 Efficient Computation of the DFT of Two Real Sequences, 475
- 6.2.2 Efficient Computation of the DFT of a $2N$-Point Real Sequence, 476
- 6.2.3 Use of the FFT Algorithm in Linear Filtering and Correlation, 477

6.3 A Linear Filtering Approach to Computation of the DFT 479
- 6.3.1 The Goertzel Algorithm, 480
- 6.3.2 The Chirp-z Transform Algorithm, 482

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6.4 Quantization Effects in the Computation of the DFT 486
6.4.1 Quantization Errors in the Direct Computation of the DFT, 487
6.4.2 Quantization Errors in FFT Algorithms, 489

6.5 Summary and References 493

Problems 494

7 IMPLEMENTATION OF DISCRETE-TIME SYSTEMS 500

7.1 Structures for the Realization of Discrete-Time Systems 500

7.2 Structures for FIR Systems 502
7.2.1 Direct-Form Structure, 503
7.2.2 Cascade-Form Structures, 504
7.2.3 Frequency-Sampling Structures†, 506
7.2.4 Lattice Structure, 511

7.3 Structures for IIR Systems 519
7.3.1 Direct-Form Structures, 519
7.3.2 Signal Flow Graphs and Transposed Structures, 521
7.3.3 Cascade-Form Structures, 526
7.3.4 Parallel-Form Structures, 529
7.3.5 Lattice and Lattice-Ladder Structures for IIR Systems, 531

7.4 State-Space System Analysis and Structures 539
7.4.1 State-Space Descriptions of Systems Characterized by Difference Equations, 540
7.4.2 Solution of the State-Space Equations, 543
7.4.3 Relationships Between Input–Output and State-Space Descriptions, 545
7.4.4 State-Space Analysis in the z-Domain, 550
7.4.5 Additional State-Space Structures, 554

7.5 Representation of Numbers 556
7.5.1 Fixed-Point Representation of Numbers, 557
7.5.2 Binary Floating-Point Representation of Numbers, 561
7.5.3 Errors Resulting from Rounding and Truncation, 564

7.6 Quantization of Filter Coefficients 569
7.6.1 Analysis of Sensitivity to Quantization of Filter Coefficients, 569
7.6.2 Quantization of Coefficients in FIR Filters, 578

7.7 Round-Off Effects in Digital Filters 582
7.7.1 Limit-Cycle Oscillations in Recursive Systems, 583
7.7.2 Scaling to Prevent Overflow, 588
7.7.3 Statistical Characterization of Quantization Effects in Fixed-Point Realizations of Digital Filters, 590

7.8 Summary and References 598

Problems 600

8 DESIGN OF DIGITAL FILTERS 614

8.1 General Considerations 614
- 8.1.1 Causality and Its Implications, 615
- 8.1.2 Characteristics of Practical Frequency-Selective Filters, 619

8.2 Design of FIR Filters 620
- 8.2.1 Symmetric and Antisymmetric FIR Filters, 620
- 8.2.2 Design of Linear-Phase FIR Filters Using Windows, 623
- 8.2.3 Design of Linear-Phase FIR Filters by the Frequency-Sampling Method, 630
- 8.2.4 Design of Optimum Equiripple Linear-Phase FIR Filters, 637
- 8.2.5 Design of FIR Differentiators, 652
- 8.2.6 Design of Hilbert Transformers, 657
- 8.2.7 Comparison of Design Methods for Linear-Phase FIR Filters, 662

8.3 Design of IIR Filters From Analog Filters 666
- 8.3.1 IIR Filter Design by Approximation of Derivatives, 667
- 8.3.2 IIR Filter Design by Impulse Invariance, 671
- 8.3.3 IIR Filter Design by the Bilinear Transformation, 676
- 8.3.4 The Matched-z Transformation, 681
- 8.3.5 Characteristics of Commonly Used Analog Filters, 681
- 8.3.6 Some Examples of Digital Filter Designs Based on the Bilinear Transformation, 692

8.4 Frequency Transformations 692
- 8.4.1 Frequency Transformations in the Analog Domain, 693
- 8.4.2 Frequency Transformations in the Digital Domain, 698

8.5 Design of Digital Filters Based on Least-Squares Method 701
- 8.5.1 Padé Approximation Method, 701
- 8.5.2 Least-Squares Design Methods, 706
- 8.5.3 FIR Least-Squares Inverse (Wiener) Filters, 711
- 8.5.4 Design of IIR Filters in the Frequency Domain, 719

8.6 Summary and References 724

Problems 726

9 SAMPLING AND RECONSTRUCTION OF SIGNALS 738

9.1 Sampling of Bandpass Signals 738
- 9.1.1 Representation of Bandpass Signals, 738
- 9.1.2 Sampling of Bandpass Signals, 742
- 9.1.3 Discrete-Time Processing of Continuous-Time Signals, 746

9.2 Analog-to-Digital Conversion 748
- 9.2.1 Sample-and-Hold, 748
- 9.2.2 Quantization and Coding, 750
- 9.2.3 Analysis of Quantization Errors, 753
- 9.2.4 Oversampling A/D Converters, 756

9.3 Digital-to-Analog Conversion 763
9.3.1 Sample and Hold, 765
9.3.2 First-Order Hold, 768
9.3.3 Linear Interpolation with Delay, 771
9.3.4 Oversampling D/A Converters, 774

9.4 Summary and References 774

Problems 775

10 MULTIRATE DIGITAL SIGNAL PROCESSING **782**

10.1 Introduction 783

10.2 Decimation by a Factor D 784

10.3 Interpolation by a Factor I 787

10.4 Sampling Rate Conversion by a Rational Factor I/D 790

10.5 Filter Design and Implementation for Sampling-Rate Conversion 792
10.5.1 Direct-Form FIR Filter Structures, 793
10.5.2 Polyphase Filter Structures, 794
10.5.3 Time-Variant Filter Structures, 800

10.6 Multistage Implementation of Sampling-Rate Conversion 806

10.7 Sampling-Rate Conversion of Bandpass Signals 810
10.7.1 Decimation and Interpolation by Frequency Conversion, 812
10.7.2 Modulation-Free Method for Decimation and Interpolation, 814

10.8 Sampling-Rate Conversion by an Arbitrary Factor 815
10.8.1 First-Order Approximation, 816
10.8.2 Second-Order Approximation (Linear Interpolation), 819

10.9 Applications of Multirate Signal Processing 821
10.9.1 Design of Phase Shifters, 821
10.9.2 Interfacing of Digital Systems with Different Sampling Rates, 823
10.9.3 Implementation of Narrowband Lowpass Filters, 824
10.9.4 Implementation of Digital Filter Banks, 825
10.9.5 Subband Coding of Speech Signals, 831
10.9.6 Quadrature Mirror Filters, 833
10.9.7 Transmultiplexers, 841
10.9.8 Oversampling A/D and D/A Conversion, 843

10.10 Summary and References 844

Problems 846

11 LINEAR PREDICTION AND OPTIMUM LINEAR FILTERS 852

11.1 Innovations Representation of a Stationary Random Process 852
- 11.1.1 Rational Power Spectra, 854
- 11.1.2 Relationships Between the Filter Parameters and the Autocorrelation Sequence, 855

11.2 Forward and Backward Linear Prediction 857
- 11.2.1 Forward Linear Prediction, 857
- 11.2.2 Backward Linear Prediction, 860
- 11.2.3 The Optimum Reflection Coefficients for the Lattice Forward and Backward Predictors, 863
- 11.2.4 Relationship of an AR Process to Linear Prediction, 864

11.3 Solution of the Normal Equations 864
- 11.3.1 The Levinson–Durbin Algorithm, 865
- 11.3.2 The Schür Algorithm, 868

11.4 Properties of the Linear Prediction-Error Filters 873

11.5 AR Lattice and ARMA Lattice-Ladder Filters 876
- 11.5.1 AR Lattice Structure, 877
- 11.5.2 ARMA Processes and Lattice-Ladder Filters, 878

11.6 Wiener Filters for Filtering and Prediction 880
- 11.6.1 FIR Wiener Filter, 881
- 11.6.2 Orthogonality Principle in Linear Mean-Square Estimation, 884
- 11.6.3 IIR Wiener Filter, 885
- 11.6.4 Noncausal Wiener Filter, 889

11.7 Summary and References 890

Problems 892

12 POWER SPECTRUM ESTIMATION 896

12.1 Estimation of Spectra from Finite-Duration Observations of Signals 896
- 12.1.1 Computation of the Energy Density Spectrum, 897
- 12.1.2 Estimation of the Autocorrelation and Power Spectrum of Random Signals: The Periodogram, 902
- 12.1.3 The Use of the DFT in Power Spectrum Estimation, 906

12.2 Nonparametric Methods for Power Spectrum Estimation 908
- 12.2.1 The Bartlett Method: Averaging Periodograms, 910
- 12.2.2 The Welch Method: Averaging Modified Periodograms, 911
- 12.2.3 The Blackman and Tukey Method: Smoothing the Periodogram, 913
- 12.2.4 Performance Characteristics of Nonparametric Power Spectrum Estimators, 916

12.2.5 Computational Requirements of Nonparametric Power Spectrum Estimates, 919

12.3 Parametric Methods for Power Spectrum Estimation 920
12.3.1 Relationships Between the Autocorrelation and the Model Parameters, 923
12.3.2 The Yule–Walker Method for the AR Model Parameters, 925
12.3.3 The Burg Method for the AR Model Parameters, 925
12.3.4 Unconstrained Least-Squares Method for the AR Model Parameters, 929
12.3.5 Sequential Estimation Methods for the AR Model Parameters, 930
12.3.6 Selection of AR Model Order, 931
12.3.7 MA Model for Power Spectrum Estimation, 933
12.3.8 ARMA Model for Power Spectrum Estimation, 934
12.3.9 Some Experimental Results, 936

12.4 Minimum Variance Spectral Estimation 942

12.5 Eigenanalysis Algorithms for Spectrum Estimation 946
12.5.1 Pisarenko Harmonic Decomposition Method, 948
12.5.2 Eigen-decomposition of the Autocorrelation Matrix for Sinusoids in White Noise, 950
12.5.3 MUSIC Algorithm, 952
12.5.4 ESPRIT Algorithm, 953
12.5.5 Order Selection Criteria, 955
12.5.6 Experimental Results, 956

12.6 Summary and References 959

Problems 960

A RANDOM SIGNALS, CORRELATION FUNCTIONS, AND POWER SPECTRA ***A1***

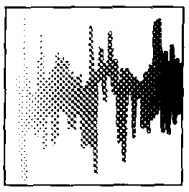
B RANDOM NUMBER GENERATORS ***B1***

C TABLES OF TRANSITION COEFFICIENTS FOR THE DESIGN OF LINEAR-PHASE FIR FILTERS ***C1***

D LIST OF MATLAB FUNCTIONS ***D1***

REFERENCES AND BIBLIOGRAPHY ***R1***

INDEX ***I1***



1 Introduction

Digital signal processing is an area of science and engineering that has developed rapidly over the past 30 years. This rapid development is a result of the significant advances in digital computer technology and integrated-circuit fabrication. The digital computers and associated digital hardware of three decades ago were relatively large and expensive and, as a consequence, their use was limited to general-purpose non-real-time (off-line) scientific computations and business applications. The rapid developments in integrated-circuit technology, starting with medium-scale integration (MSI) and progressing to large-scale integration (LSI), and now, very-large-scale integration (VLSI) of electronic circuits has spurred the development of powerful, smaller, faster, and cheaper digital computers and special-purpose digital hardware. These inexpensive and relatively fast digital circuits have made it possible to construct highly sophisticated digital systems capable of performing complex digital signal processing functions and tasks, which are usually too difficult and/or too expensive to be performed by analog circuitry or analog signal processing systems. Hence many of the signal processing tasks that were conventionally performed by analog means are realized today by less expensive and often more reliable digital hardware.

We do not wish to imply that digital signal processing is the proper solution for all signal processing problems. Indeed, for many signals with extremely wide bandwidths, real-time processing is a requirement. For such signals, analog or, perhaps, optical signal processing is the only possible solution. However, where digital circuits are available and have sufficient speed to perform the signal processing, they are usually preferable.

Not only do digital circuits yield cheaper and more reliable systems for signal processing, they have other advantages as well. In particular, digital processing hardware allows programmable operations. Through software, one can more easily modify the signal processing functions to be performed by the hardware. Thus digital hardware and associated software provide a greater degree of flexibility in system design. Also, there is often a higher order of precision achievable with digital hardware and software compared with analog circuits and analog signal processing systems. For all these reasons, there has been an explosive growth in digital signal processing theory and applications over the past three decades.

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