

CONTENTS

Preface	xv
The Companion Website	xxii
The Cover	xxv
Acknowledgments	xxvi
1 Introduction	1
2 Discrete-Time Signals and Systems	9
2.0 Introduction	9
2.1 Discrete-Time Signals	10
2.2 Discrete-Time Systems	17
2.2.1 Memoryless Systems	18
2.2.2 Linear Systems	19
2.2.3 Time-Invariant Systems	20
2.2.4 Causality	22
2.2.5 Stability	22
2.3 LTI Systems	23
2.4 Properties of Linear Time-Invariant Systems	30
2.5 Linear Constant-Coefficient Difference Equations	35
2.6 Frequency-Domain Representation of Discrete-Time Signals and Systems	40
2.6.1 Eigenfunctions for Linear Time-Invariant Systems	40
2.6.2 Suddenly Applied Complex Exponential Inputs	46
2.7 Representation of Sequences by Fourier Transforms	48
2.8 Symmetry Properties of the Fourier Transform	54
2.9 Fourier Transform Theorems	58
2.9.1 Linearity of the Fourier Transform	59
2.9.2 Time Shifting and Frequency Shifting Theorem	59
2.9.3 Time Reversal Theorem	59

2.9.4	Differentiation in Frequency Theorem	59
2.9.5	Parseval's Theorem	60
2.9.6	The Convolution Theorem	60
2.9.7	The Modulation or Windowing Theorem	61
2.10	Discrete-Time Random Signals	64
2.11	Summary	70
	Problems	70
3	The z-Transform	99
3.0	Introduction	99
3.1	z -Transform	99
3.2	Properties of the ROC for the z -Transform	110
3.3	The Inverse z -Transform	115
3.3.1	Inspection Method	116
3.3.2	Partial Fraction Expansion	116
3.3.3	Power Series Expansion	122
3.4	z -Transform Properties	124
3.4.1	Linearity	124
3.4.2	Time Shifting	125
3.4.3	Multiplication by an Exponential Sequence	126
3.4.4	Differentiation of $X(z)$	127
3.4.5	Conjugation of a Complex Sequence	129
3.4.6	Time Reversal	129
3.4.7	Convolution of Sequences	130
3.4.8	Summary of Some z -Transform Properties	131
3.5	z -Transforms and LTI Systems	131
3.6	The Unilateral z -Transform	135
3.7	Summary	137
	Problems	138
4	Sampling of Continuous-Time Signals	153
4.0	Introduction	153
4.1	Periodic Sampling	153
4.2	Frequency-Domain Representation of Sampling	156
4.3	Reconstruction of a Bandlimited Signal from Its Samples	163
4.4	Discrete-Time Processing of Continuous-Time Signals	167
4.4.1	Discrete-Time LTI Processing of Continuous-Time Signals	168
4.4.2	Impulse Invariance	173
4.5	Continuous-Time Processing of Discrete-Time Signals	175
4.6	Changing the Sampling Rate Using Discrete-Time Processing	179
4.6.1	Sampling Rate Reduction by an Integer Factor	180
4.6.2	Increasing the Sampling Rate by an Integer Factor	184
4.6.3	Simple and Practical Interpolation Filters	187
4.6.4	Changing the Sampling Rate by a Noninteger Factor	190
4.7	Multirate Signal Processing	194
4.7.1	Interchange of Filtering with Compressor/Expander	194
4.7.2	Multistage Decimation and Interpolation	195

4.7.3	Polyphase Decompositions	197
4.7.4	Polyphase Implementation of Decimation Filters	199
4.7.5	Polyphase Implementation of Interpolation Filters	200
4.7.6	Multirate Filter Banks	201
4.8	Digital Processing of Analog Signals	205
4.8.1	Prefiltering to Avoid Aliasing	206
4.8.2	A/D Conversion	209
4.8.3	Analysis of Quantization Errors	214
4.8.4	D/A Conversion	221
4.9	Oversampling and Noise Shaping in A/D and D/A Conversion	224
4.9.1	Oversampled A/D Conversion with Direct Quantization	225
4.9.2	Oversampled A/D Conversion with Noise Shaping	229
4.9.3	Oversampling and Noise Shaping in D/A Conversion	234
4.10	Summary	236
	Problems	237
5	Transform Analysis of Linear Time-Invariant Systems	274
5.0	Introduction	274
5.1	The Frequency Response of LTI Systems	275
5.1.1	Frequency Response Phase and Group Delay	275
5.1.2	Illustration of Effects of Group Delay and Attenuation	278
5.2	System Functions—Linear Constant-Coefficient Difference Equations	283
5.2.1	Stability and Causality	285
5.2.2	Inverse Systems	286
5.2.3	Impulse Response for Rational System Functions	288
5.3	Frequency Response for Rational System Functions	290
5.3.1	Frequency Response of 1 st -Order Systems	292
5.3.2	Examples with Multiple Poles and Zeros	296
5.4	Relationship between Magnitude and Phase	301
5.5	All-Pass Systems	305
5.6	Minimum-Phase Systems	311
5.6.1	Minimum-Phase and All-Pass Decomposition	311
5.6.2	Frequency-Response Compensation of Non-Minimum-Phase Systems	313
5.6.3	Properties of Minimum-Phase Systems	318
5.7	Linear Systems with Generalized Linear Phase	322
5.7.1	Systems with Linear Phase	322
5.7.2	Generalized Linear Phase	326
5.7.3	Causal Generalized Linear-Phase Systems	328
5.7.4	Relation of FIR Linear-Phase Systems to Minimum-Phase Systems	338
5.8	Summary	340
	Problems	341

6	Structures for Discrete-Time Systems	374
6.0	Introduction	374
6.1	Block Diagram Representation of Linear Constant-Coefficient Difference Equations	375
6.2	Signal Flow Graph Representation	382
6.3	Basic Structures for IIR Systems	388
6.3.1	Direct Forms	388
6.3.2	Cascade Form	390
6.3.3	Parallel Form	393
6.3.4	Feedback in IIR Systems	395
6.4	Transposed Forms	397
6.5	Basic Network Structures for FIR Systems	401
6.5.1	Direct Form	401
6.5.2	Cascade Form	402
6.5.3	Structures for Linear-Phase FIR Systems	403
6.6	Lattice Filters	405
6.6.1	FIR Lattice Filters	406
6.6.2	All-Pole Lattice Structure	412
6.6.3	Generalization of Lattice Systems	415
6.7	Overview of Finite-Precision Numerical Effects	415
6.7.1	Number Representations	415
6.7.2	Quantization in Implementing Systems	419
6.8	The Effects of Coefficient Quantization	421
6.8.1	Effects of Coefficient Quantization in IIR Systems	422
6.8.2	Example of Coefficient Quantization in an Elliptic Filter	423
6.8.3	Poles of Quantized 2 nd -Order Sections	427
6.8.4	Effects of Coefficient Quantization in FIR Systems	429
6.8.5	Example of Quantization of an Optimum FIR Filter	431
6.8.6	Maintaining Linear Phase	434
6.9	Effects of Round-off Noise in Digital Filters	436
6.9.1	Analysis of the Direct Form IIR Structures	436
6.9.2	Scaling in Fixed-Point Implementations of IIR Systems	445
6.9.3	Example of Analysis of a Cascade IIR Structure	448
6.9.4	Analysis of Direct-Form FIR Systems	453
6.9.5	Floating-Point Realizations of Discrete-Time Systems	458
6.10	Zero-Input Limit Cycles in Fixed-Point Realizations of IIR Digital Filters	459
6.10.1	Limit Cycles Owing to Round-off and Truncation	459
6.10.2	Limit Cycles Owing to Overflow	462
6.10.3	Avoiding Limit Cycles	463
6.11	Summary	463
	Problems	464
7	Filter Design Techniques	493
7.0	Introduction	493
7.1	Filter Specifications	494

7.2	Design of Discrete-Time IIR Filters from Continuous-Time Filters . . .	496
7.2.1	Filter Design by Impulse Invariance	497
7.2.2	Bilinear Transformation	504
7.3	Discrete-Time Butterworth, Chebyshev and Elliptic Filters	508
7.3.1	Examples of IIR Filter Design	509
7.4	Frequency Transformations of Lowpass IIR Filters	526
7.5	Design of FIR Filters by Windowing	533
7.5.1	Properties of Commonly Used Windows	535
7.5.2	Incorporation of Generalized Linear Phase	538
7.5.3	The Kaiser Window Filter Design Method	541
7.6	Examples of FIR Filter Design by the Kaiser Window Method	545
7.6.1	Lowpass Filter	545
7.6.2	Highpass Filter	547
7.6.3	Discrete-Time Differentiators	550
7.7	Optimum Approximations of FIR Filters	554
7.7.1	Optimal Type I Lowpass Filters	559
7.7.2	Optimal Type II Lowpass Filters	565
7.7.3	The Parks–McClellan Algorithm	566
7.7.4	Characteristics of Optimum FIR Filters	568
7.8	Examples of FIR Equiripple Approximation	570
7.8.1	Lowpass Filter	570
7.8.2	Compensation for Zero-Order Hold	571
7.8.3	Bandpass Filter	576
7.9	Comments on IIR and FIR Discrete-Time Filters	578
7.10	Design of an Upsampling Filter	579
7.11	Summary	582
	Problems	582
8	The Discrete Fourier Transform	623
8.0	Introduction	623
8.1	Representation of Periodic Sequences: The Discrete Fourier Series . .	624
8.2	Properties of the DFS	628
8.2.1	Linearity	629
8.2.2	Shift of a Sequence	629
8.2.3	Duality	629
8.2.4	Symmetry Properties	630
8.2.5	Periodic Convolution	630
8.2.6	Summary of Properties of the DFS Representation of Periodic Sequences	633
8.3	The Fourier Transform of Periodic Signals	633
8.4	Sampling the Fourier Transform	638
8.5	Fourier Representation of Finite-Duration Sequences	642
8.6	Properties of the DFT	647
8.6.1	Linearity	647
8.6.2	Circular Shift of a Sequence	648
8.6.3	Duality	650
8.6.4	Symmetry Properties	653

8.6.5	Circular Convolution	654
8.6.6	Summary of Properties of the DFT	659
8.7	Linear Convolution Using the DFT	660
8.7.1	Linear Convolution of Two Finite-Length Sequences	661
8.7.2	Circular Convolution as Linear Convolution with Aliasing	661
8.7.3	Implementing Linear Time-Invariant Systems Using the DFT	667
8.8	The Discrete Cosine Transform (DCT)	673
8.8.1	Definitions of the DCT	673
8.8.2	Definition of the DCT-1 and DCT-2	675
8.8.3	Relationship between the DFT and the DCT-1	676
8.8.4	Relationship between the DFT and the DCT-2	678
8.8.5	Energy Compaction Property of the DCT-2	679
8.8.6	Applications of the DCT	682
8.9	Summary	683
	Problems	684
9	Computation of the Discrete Fourier Transform	716
9.0	Introduction	716
9.1	Direct Computation of the Discrete Fourier Transform	718
9.1.1	Direct Evaluation of the Definition of the DFT	718
9.1.2	The Goertzel Algorithm	719
9.1.3	Exploiting both Symmetry and Periodicity	722
9.2	Decimation-in-Time FFT Algorithms	723
9.2.1	Generalization and Programming the FFT	731
9.2.2	In-Place Computations	731
9.2.3	Alternative Forms	734
9.3	Decimation-in-Frequency FFT Algorithms	737
9.3.1	In-Place Computation	741
9.3.2	Alternative Forms	741
9.4	Practical Considerations	743
9.4.1	Indexing	743
9.4.2	Coefficients	745
9.5	More General FFT Algorithms	745
9.5.1	Algorithms for Composite Values of N	746
9.5.2	Optimized FFT Algorithms	748
9.6	Implementation of the DFT Using Convolution	748
9.6.1	Overview of the Winograd Fourier Transform Algorithm	749
9.6.2	The Chirp Transform Algorithm	749
9.7	Effects of Finite Register Length	754
9.8	Summary	762
	Problems	763
10	Fourier Analysis of Signals Using the Discrete Fourier Transform	792
10.0	Introduction	792
10.1	Fourier Analysis of Signals Using the DFT	793

10.2	DFT Analysis of Sinusoidal Signals	797
10.2.1	The Effect of Windowing	797
10.2.2	Properties of the Windows	800
10.2.3	The Effect of Spectral Sampling	801
10.3	The Time-Dependent Fourier Transform	811
10.3.1	Invertibility of $X[n, \omega]$	815
10.3.2	Filter Bank Interpretation of $X[n, \omega]$	816
10.3.3	The Effect of the Window	817
10.3.4	Sampling in Time and Frequency	819
10.3.5	The Overlap–Add Method of Reconstruction	822
10.3.6	Signal Processing Based on the Time-Dependent Fourier Transform	825
10.3.7	Filter Bank Interpretation of the Time-Dependent Fourier Transform	826
10.4	Examples of Fourier Analysis of Nonstationary Signals	829
10.4.1	Time-Dependent Fourier Analysis of Speech Signals	830
10.4.2	Time-Dependent Fourier Analysis of Radar Signals	834
10.5	Fourier Analysis of Stationary Random Signals: the Periodogram	836
10.5.1	The Periodogram	837
10.5.2	Properties of the Periodogram	839
10.5.3	Periodogram Averaging	843
10.5.4	Computation of Average Periodograms Using the DFT	845
10.5.5	An Example of Periodogram Analysis	845
10.6	Spectrum Analysis of Random Signals	849
10.6.1	Computing Correlation and Power Spectrum Estimates Using the DFT	853
10.6.2	Estimating the Power Spectrum of Quantization Noise	855
10.6.3	Estimating the Power Spectrum of Speech	860
10.7	Summary	862
	Problems	864
11	Parametric Signal Modeling	890
11.0	Introduction	890
11.1	All-Pole Modeling of Signals	891
11.1.1	Least-Squares Approximation	892
11.1.2	Least-Squares Inverse Model	892
11.1.3	Linear Prediction Formulation of All-Pole Modeling	895
11.2	Deterministic and Random Signal Models	896
11.2.1	All-Pole Modeling of Finite-Energy Deterministic Signals	896
11.2.2	Modeling of Random Signals	897
11.2.3	Minimum Mean-Squared Error	898
11.2.4	Autocorrelation Matching Property	898
11.2.5	Determination of the Gain Parameter G	899
11.3	Estimation of the Correlation Functions	900
11.3.1	The Autocorrelation Method	900
11.3.2	The Covariance Method	903
11.3.3	Comparison of Methods	904

11.4	Model Order	905
11.5	All-Pole Spectrum Analysis	907
11.5.1	All-Pole Analysis of Speech Signals	908
11.5.2	Pole Locations	911
11.5.3	All-Pole Modeling of Sinusoidal Signals	913
11.6	Solution of the Autocorrelation Normal Equations	915
11.6.1	The Levinson–Durbin Recursion	916
11.6.2	Derivation of the Levinson–Durbin Algorithm	917
11.7	Lattice Filters	920
11.7.1	Prediction Error Lattice Network	921
11.7.2	All-Pole Model Lattice Network	923
11.7.3	Direct Computation of the k -Parameters	925
11.8	Summary	926
	Problems	927
12	Discrete Hilbert Transforms	942
12.0	Introduction	942
12.1	Real- and Imaginary-Part Sufficiency of the Fourier Transform	944
12.2	Sufficiency Theorems for Finite-Length Sequences	949
12.3	Relationships Between Magnitude and Phase	955
12.4	Hilbert Transform Relations for Complex Sequences	956
12.4.1	Design of Hilbert Transformers	960
12.4.2	Representation of Bandpass Signals	963
12.4.3	Bandpass Sampling	966
12.5	Summary	969
	Problems	969
13	Cepstrum Analysis and Homomorphic Deconvolution	980
13.0	Introduction	980
13.1	Definition of the Cepstrum	981
13.2	Definition of the Complex Cepstrum	982
13.3	Properties of the Complex Logarithm	984
13.4	Alternative Expressions for the Complex Cepstrum	985
13.5	Properties of the Complex Cepstrum	986
13.5.1	Exponential Sequences	986
13.5.2	Minimum-Phase and Maximum-Phase Sequences	989
13.5.3	Relationship Between the Real Cepstrum and the Complex Cepstrum	990
13.6	Computation of the Complex Cepstrum	992
13.6.1	Phase Unwrapping	993
13.6.2	Computation of the Complex Cepstrum Using the Logarithmic Derivative	996
13.6.3	Minimum-Phase Realizations for Minimum-Phase Sequences	998
13.6.4	Recursive Computation of the Complex Cepstrum for Minimum- and Maximum-Phase Sequences	998
13.6.5	The Use of Exponential Weighting	1000
13.7	Computation of the Complex Cepstrum Using Polynomial Roots	1001

13.8	Deconvolution Using the Complex Cepstrum	1002
13.8.1	Minimum-Phase/Allpass Homomorphic Deconvolution	1003
13.8.2	Minimum-Phase/Maximum-Phase Homomorphic Deconvolution	1004
13.9	The Complex Cepstrum for a Simple Multipath Model	1006
13.9.1	Computation of the Complex Cepstrum by z -Transform Analysis	1009
13.9.2	Computation of the Cepstrum Using the DFT	1013
13.9.3	Homomorphic Deconvolution for the Multipath Model	1016
13.9.4	Minimum-Phase Decomposition	1017
13.9.5	Generalizations	1024
13.10	Applications to Speech Processing	1024
13.10.1	The Speech Model	1024
13.10.2	Example of Homomorphic Deconvolution of Speech	1028
13.10.3	Estimating the Parameters of the Speech Model	1030
13.10.4	Applications	1032
13.11	Summary	1032
	Problems	1034
A	Random Signals	1043
B	Continuous-Time Filters	1056
C	Answers to Selected Basic Problems	1061
	Bibliography	1082
	Index	1091