

Contents

Preface xiii

CHAPTER 1 Introduction to Digital Signal Processing..... 1

1.1 Basic Concepts of Digital Signal Processing 1

1.2 Basic Digital Signal Processing Examples in Block Diagrams 2

1.2.1 Digital Filtering 3

1.2.2 Signal Frequency (Spectrum) Analysis..... 4

1.3 Overview of Typical Digital Signal Processing in Real-World Applications 5

1.3.1 Digital Crossover Audio System..... 5

1.3.2 Interference Cancellation in Electrocardiography 6

1.3.3 Speech Coding and Compression 6

1.3.4 Compact-Disc Recording System 7

1.3.5 Vibration Signature Analysis for Defected Gear Tooth..... 9

1.3.6 Digital Image Enhancement..... 10

1.4 Digital Signal Processing Applications 12

1.5 Summary..... 12

CHAPTER 2 Signal Sampling and Quantization..... 13

2.1 Sampling of Continuous Signal 13

2.2 Signal Reconstruction..... 20

2.2.1 Practical Considerations for Signal Sampling: Anti-Aliasing Filtering..... 24

2.2.2 Practical Considerations for Signal Reconstruction: Anti-Image Filter
and Equalizer..... 28

2.3 Analog-to-Digital Conversion, Digital-to-Analog Conversion, and
Quantization..... 35

2.4 Summary..... 47

2.5 MATLAB Programs 48

2.6 Problems 49

CHAPTER 3 Digital Signals and Systems 59

3.1 Digital Signals 59

3.1.1 Common Digital Sequences..... 60

3.1.2 Generation of Digital Signals..... 63

3.2 Linear Time-Invariant, Causal Systems..... 65

3.2.1 Linearity..... 66

3.2.2 Time Invariance..... 67

3.2.3 Causality 68

3.3	Difference Equations and Impulse Responses.....	69
3.3.1	Format of Difference Equation	69
3.3.2	System Representation Using Its Impulse Response.....	70
3.4	Digital Convolution.....	73
3.5	Bounded-Input and Bounded-Output Stability	81
3.6	Summary.....	82
3.7	Problems	83
CHAPTER 4	Discrete Fourier Transform and Signal Spectrum.....	91
4.1	Discrete Fourier Transform.....	91
4.1.1	Fourier Series Coefficients of Periodic Digital Signals	92
4.1.2	Discrete Fourier Transform Formulas.....	96
4.2	Amplitude Spectrum and Power Spectrum.....	102
4.3	Spectral Estimation Using Window Functions.....	112
4.4	Application to Signal Spectral Estimation.....	121
4.5	Fast Fourier Transform.....	127
4.5.1	Method of Decimation-in-Frequency.....	128
4.5.2	Method of Decimation-in-Time	133
4.6	Summary.....	137
4.7	Problems	137
CHAPTER 5	The z-Transform.....	143
5.1	Definition.....	143
5.2	Properties of the z-Transform	146
5.3	Inverse z-Transform.....	150
5.3.1	Partial Fraction Expansion and Look-Up Table	151
5.3.2	Partial Fraction Expansion Using MATLAB	155
5.3.3	Power Series Method	158
5.3.4	Inversion Formula Method.....	159
5.4	Solution of Difference Equations Using the z-Transform.....	162
5.5	Two-Sided z-Transform	165
5.6	Summary.....	167
5.7	Problems	168
CHAPTER 6	Digital Signal Processing Systems, Basic Filtering Types, and Digital Filter Realizations	173
6.1	Difference Equation and Digital Filtering	173
6.2	Difference Equation and Transfer Function	178
6.2.1	Impulse Response, Step Response, and System Response	181
6.3	The z-Plane Pole-Zero Plot and Stability	184

6.4	Digital Filter Frequency Response.....	190
6.5	Basic Types of Filtering.....	197
6.6	Realization of Digital Filters.....	204
6.6.1	Direct-Form I Realization	204
6.6.2	Direct-Form II Realization.....	205
6.6.3	Cascade (Series) Realization.....	207
6.6.4	Parallel Realization	207
6.7	Application: Signal Enhancement and Filtering.....	211
6.7.1	Preemphasis of Speech.....	211
6.7.2	Bandpass Filtering of Speech.....	214
6.7.3	Enhancement of ECG Signal Using Notch Filtering	217
6.8	Summary.....	219
6.9	Problems	219
CHAPTER 7	Finite Impulse Response Filter Design	229
7.1	Finite Impulse Response Filter Format.....	229
7.2	Fourier Transform Design.....	231
7.3	Window Method	242
7.4	Applications: Noise Reduction and Two-Band Digital Crossover	265
7.4.1	Noise Reduction	265
7.4.2	Speech Noise Reduction	267
7.4.3	Noise Reduction in Vibration Signal	269
7.4.4	Two-Band Digital Crossover	270
7.5	Frequency Sampling Design Method.....	272
7.6	Optimal Design Method.....	282
7.7	Design of FIR Differentiator and Hilbert Transformer.....	293
7.8	Realization Structures of Finite Impulse Response Filters	295
7.8.1	Transversal Form.....	296
7.8.2	Linear Phase Form	297
7.9	Coefficient Accuracy Effects on Finite Impulse Response Filters	298
7.10	Summary of FIR Design Procedures and Selection of the FIR Filter Design Methods in Practice	301
7.11	Summary.....	304
7.12	MATLAB Programs.....	304
7.13	Problems	306
CHAPTER 8	Infinite Impulse Response Filter Design	315
8.1	Infinite Impulse Response Filter Format	316
8.2	Bilinear Transformation Design Method.....	317
8.2.1	Analog Filters Using Lowpass Prototype Transformation	318

8.2.2 Bilinear Transformation and Frequency Warping	322
8.2.3 Bilinear Transformation Design Procedure	328
8.3 Digital Butterworth and Chebyshev Filter Designs	332
8.3.1 Lowpass Prototype Function and Its Order	332
8.3.2 Lowpass and Highpass Filter Design Examples	335
8.3.3 Bandpass and Bandstop Filter Design Examples	344
8.4 Higher-Order Infinite Impulse Response Filter Design Using the Cascade Method	352
8.5 Application: Digital Audio Equalizer	355
8.6 Impulse Invariant Design Method	359
8.7 Pole-Zero Placement Method for Simple Infinite Impulse Response Filters	365
8.7.1 Second-Order Bandpass Filter Design	366
8.7.2 Second-Order Bandstop (Notch) Filter Design	368
8.7.3 First-Order Lowpass Filter Design	369
8.7.4 First-Order Highpass Filter Design	372
8.8 Realization Structures of Infinite Impulse Response Filters	373
8.8.1 Realization of Infinite Impulse Response Filters in Direct-Form I and Direct-Form II	373
8.8.2 Realization of Higher-Order Infinite Impulse Response Filters Via the Cascade Form	376
8.9 Application: 60-Hz Hum Eliminator and Heart Rate Detection Using Electrocardiography	377
8.10 Coefficient Accuracy Effects on Infinite Impulse Response Filters	384
8.11 Application: Generation and Detection of DTMF Tones Using the Goertzel Algorithm	388
8.11.1 Single-Tone Generator	389
8.11.2 Dual-Tone Multifrequency Tone Generator	391
8.11.3 Goertzel Algorithm	392
8.11.4 Dual-Tone Multifrequency Tone Detection Using the Modified Goertzel Algorithm	398
8.12 Summary of Infinite Impulse Response (IIR) Design Procedures and Selection of the IIR Filter Design Methods in Practice	404
8.13 Summary	407
8.14 Problems	407

CHAPTER 9 Adaptive Filters and Applications 421

9.1 Introduction to Least Mean Square Adaptive Finite Impulse Response Filters	421
9.2 Basic Wiener Filter Theory and Adaptive Algorithms	425
9.2.1 Wiener Filter Theory and Linear Prediction	425
9.2.2 Steepest Descent Algorithm	433

9.2.3 Least Mean Square Algorithm	436
9.2.4 Recursive Least Squares Algorithm	437
9.3 Applications: Noise Cancellation, System Modeling, and Line Enhancement	440
9.3.1 Noise Cancellation	440
9.3.2 System Modeling	448
9.3.3 Line Enhancement Using Linear Prediction	454
9.4 Other Application Examples	457
9.4.1 Canceling Periodic Interferences Using Linear Prediction	457
9.4.2 Electrocardiography Interference Cancellation	458
9.4.3 Echo Cancellation in Long-Distance Telephone Circuits	460
9.5 Summary	461
9.6 Problems	462

CHAPTER 10 Waveform Quantization and Compression 475

10.1 Linear Midtread Quantization	475
10.2 μ -Law Companding	478
10.2.1 Analog μ -Law Companding	479
10.2.2 Digital μ -Law Companding	483
10.3 Examples of Differential Pulse Code Modulation (DPCM), Delta Modulation, and Adaptive DPCM G.721	486
10.3.1 Examples of DPCM and Delta Modulation	487
10.3.2 Adaptive Differential Pulse Code Modulation G.721	490
10.4 Discrete Cosine Transform, Modified Discrete Cosine Transform, and Transform Coding in MPEG Audio	496
10.4.1 Discrete Cosine Transform	496
10.4.2 Modified Discrete Cosine Transform	499
10.4.3 Transform Coding in MPEG Audio	502
10.5 Summary	505
10.6 MATLAB Programs	506
10.7 Problems	521

CHAPTER 11 Multirate Digital Signal Processing, Oversampling of Analog-to-Digital Conversion, and Undersampling of Bandpass Signals 529

11.1 Multirate Digital Signal Processing Basics	529
11.1.1 Sampling Rate Reduction by an Integer Factor	530
11.1.2 Sampling Rate Increase by an Integer Factor	536
11.1.3 Changing Sampling Rate by a Non-Integer Factor L/M	541
11.1.4 Application: CD Audio Player	545
11.1.5 Multistage Decimation	548

11.2 Polyphase Filter Structure and Implementation	552
11.3 Oversampling of Analog-To-Digital Conversion	559
11.3.1 Oversampling and ADC Resolution	560
11.3.2 Sigma-Delta Modulation ADC	565
11.4 Application Example: CD Player.....	574
11.5 Undersampling of Bandpass Signals.....	576
11.6 Summary	583
11.7 Problems	584
11.8 MATLAB Problems	588
MATLAB Project.....	590

CHAPTER 12 Subband and Wavelet-Based Coding..... 591

12.1 Subband Coding Basics.....	591
12.2 Subband Decomposition and Two-Channel Perfect Reconstruction- Quadrature Mirror Filter Bank.....	596
12.3 Subband Coding of Signals.....	604
12.4 Wavelet Basics and Families of Wavelets.....	607
12.5 Multiresolution Equations	619
12.6 Discrete Wavelet Transform	623
12.7 Wavelet Transform Coding of Signals	632
12.8 MATLAB Programs.....	638
12.9 Summary	640
12.10 Problems	641

CHAPTER 13 Image Processing Basics 649

13.1 Image Processing Notation and Data Formats	650
13.1.1 8-Bit Gray Level Images.....	650
13.1.2 24-Bit Color Images	651
13.1.3 8-Bit Color Images.....	652
13.1.4 Intensity Images	653
13.1.5 RGB Components and Grayscale Conversion.....	654
13.1.6 MATLAB Functions for Format Conversion	656
13.2 Image Histogram and Equalization.....	658
13.2.1 Grayscale Histogram and Equalization.....	658
13.2.2 24-Bit Color Image Equalization	663
13.2.3 8-Bit Indexed Color Image Equalization.....	664
13.2.4 MATLAB Functions for Equalization	665
13.3 Image Level Adjustment and Contrast	669
13.3.1 Linear Level Adjustment.....	669
13.3.2 Adjusting the Level for Display.....	670
13.3.3 MATLAB Functions for Image Level Adjustment	672

13.4 Image Filtering Enhancement	673
13.4.1 Lowpass Noise Filtering.....	673
13.4.2 Median Filtering	677
13.4.3 Edge Detection	679
13.4.4 MATLAB Functions for Image Filtering	682
13.5 Image Pseudo-Color Generation and Detection	684
13.6 Image Spectra	687
13.7 Image Compression by Discrete Cosine Transform.....	691
13.7.1 Two-Dimensional Discrete Cosine Transform	692
13.7.2 Two-Dimensional JPEG Grayscale Image Compression Example.....	694
13.7.3 JPEG Color Image Compression	697
13.7.4 Image Compression Using Wavelet Transform Coding	701
13.8 Creating a Video Sequence by Mixing two Images.....	708
13.9 Video Signal Basics	709
13.9.1 Analog Video.....	709
13.9.2 Digital Video	714
13.10 Motion Estimation in Video.....	716
13.11 Summary	719
13.12 Problems	720

CHAPTER 14 Hardware and Software for Digital Signal Processors 727

14.1 Digital Signal Processor Architecture.....	727
14.2 DSP Hardware Units	730
14.2.1 Multiplier and Accumulator	730
14.2.2 Shifters.....	731
14.2.3 Address Generators	731
14.3 DSPs and Manufactures	733
14.4 Fixed-Point and Floating-Point Formats.....	733
14.4.1 Fixed-Point Format.....	734
14.4.2 Floating-Point Format	741
14.4.3 IEEE Floating-Point Formats.....	745
14.4.4 Fixed-Point DSPs	748
14.4.5 Floating-Point DSPs	749
14.5 Finite Impulse Response and Infinite Impulse Response Filter Implementations in Fixed-Point Systems	751
14.6 Digital Signal Processing Programming Examples	756
14.6.1 Overview of TMS320C67x DSK.....	757
14.6.2 Concept of Real-Time Processing	761
14.6.3 Linear Buffering	761
14.6.4 Sample C Programs	764

14.7 Additional Real-Time DSP Examples	768
14.7.1 Adaptive Filtering Using the TMS320C6713 DSK	768
14.7.2 Signal Quantization Using the TMS320C6713 DSK	773
14.7.3 Sampling Rate Conversion Using the TMS320C6713 DSK.....	776
14.8 Summary	781
14.9 Problems	782
Appendix A: Introduction to the MATLAB Environment	785
Appendix B: Review of Analog Signal Processing Basics	795
Appendix C: Normalized Butterworth and Chebyshev Functions	825
Appendix D: Sinusoidal Steady-State Response of Digital Filters	833
Appendix E: Finite Impulse Response Filter Design Equations by Frequency Sampling Design Method.....	837
Appendix F: Wavelet Analysis and Synthesis Equations.....	841
Appendix G: Review of Discrete-Time Random Signals	845
Appendix H: Some Useful Mathematical Formulas.....	853
Answers to Selected Problems	859
References.....	885
Index	889