

TABLE OF CONTENTS

Preface.....	viii
Use in Courses.....	xi
Interactive Approach Using Java-DSP.....	xi
Acknowledgements	xii
Trademarks.....	xiii
References	xiii
1. Review of Continuous-Time Signals and Systems.....	1
1.1. Introduction	1
1.2. Continuous-Time Signals; Definitions and Classification.....	2
1.3. Spectra of Analog Signals	4
1.3.1. The Fourier Series	6
1.3.2. The Fourier Transform	9
1.4. Review of Convolution And Filtering	13
1.5. Uniform Sampling.....	18
1.6. Aliasing Effects Demonstrated with Sinusoidal Signals	22
1.6.1. Sampling of the Impulse Response	23
1.7. Summary	24
1.7.1. Short Concept Questions	25
1.7.2. Further Reading.....	26
Problems.....	28
J-DSP Laboratory on Digital Filter Basics.....	31
General Information on J-DSP	31
Generating Signals.....	35
Performing Simulations.....	39
Visualizing Results Graphically	41
Spectral Estimation of MIDI and DTMF Signals.....	43

2. Basics of Discrete-Time Signals and Systems	49
2.1. Introduction	49
2.2. The Discrete-Time Fourier Transform	50
2.3. Digital Filters and Difference Equations	53
2.4. Transient and Steady-State Response of Digital Filters	58
2.5. The Frequency Response Function.....	64
2.6. Steady-State Response with a Sinusoidal Input.....	66
2.7. Summary	74
2.7.1. Short Concept Questions	75
2.7.2. Further Reading	76
Problems	78
J-DSP Laboratory Exercises: Digital Filter Simulations	80
3. The Z-Transform	85
3.1. Introduction	85
3.2. Definition	86
3.3. The Transfer Function	92
3.4. Inverse Z-Transform.....	92
3.5. Poles and Zeros and Frequency Response.....	97
3.6. Cascade and Parallel Configurations	103
3.7. Summary	105
3.7.1. Short Concept Questions	106
3.7.2. Further Reading	108
Problems.....	109
J-DSP Laboratory Exercises: Z-Transform and Frequency Response	112
4. FIR Filter Design.....	121
4.1. Introduction	121
4.2. Linear Phase FIR Filter Design	122
4.3. Design of Linear Phase FIR Filters using the Fourier Series.....	126

4.4.	Windows and FIR Design using the Fourier Series.....	131
4.4.1.	Design FIR Filter using the Kaiser Window	135
4.5.	Filter Design using the Parks-McClellan Method	139
4.6.	FIR Filter Design by Frequency Sampling	143
4.7.	Summary	146
4.7.1.	Short Concept Questions	148
4.7.2.	Further Reading.....	150
	Problems.....	151
	J-DSP Laboratory Exercises: Digital Filter Simulations	152
5.	IIR Filter Design	160
5.1.	Introduction	160
5.2.	IIR Filter Design using Analog Filter Approximations	161
5.2.1.	The Impulse Invariance Method.....	161
5.2.2.	The Bilinear Transformation	163
5.3.	The Butterworth Filter Design.....	167
5.4.	Chebyshev Filter Design	172
5.5.	The Elliptic Design.....	173
5.6.	IIR Filter Transformations.....	176
5.7.	Shelving Filters, Peaking Filters and Graphic Equalizers	176
5.7.1.	Shelving Filters.....	177
5.7.2.	Peaking Filters	178
5.7.3.	The Graphic Equalizer.....	180
5.8.	Summary	183
5.8.1.	Short Concept Questions	184
5.8.2.	Further Reading.....	185
	Problems.....	186
	J-DSP Laboratory Exercises: Digital Filter Simulations	188

6. Multirate Signal Processing	192
6.1. Introduction	192
6.2. Downsampling By an Integer	193
6.3. Upsampling By an Integer	196
6.4. Sampling Rate Changes by Noninteger Factors	199
6.5. Practical Considerations	201
6.6. Sampling Rate Changes in A/D and D/A Converters	206
6.7. Quadrature Mirror Filter (QMF) Banks	210
6.8. Analysis-Synthesis Framework for M-Band Filter Banks	216
6.9. Pseudo-QMF Filter Banks and MP3 Algorithms	219
6.10. Summary	220
6.10.1. Short Concept Questions	222
6.10.2. Further Reading	223
Problems	224
J-DSP Laboratory Exercises: Multirate Signal Processing	226
7. The Discrete and the Fast Fourier Transforms	230
7.1. Introduction	230
7.2. The Discrete Fourier Transform	231
7.3. The DFT Properties	236
7.4. Windows and DFTs	238
7.5. Matrix Representation of the DFT	243
7.6. Signal Analysis-Synthesis using the DFT	246
7.7. Running DFT and Filter Banks	249
7.8. The Decimation-In-Time Fast Fourier Transform	252
7.9. Fast Convolutions using the FFT	257
7.10. Summary	262
7.10.1. Short Concept Questions	264
7.10.2. Further Reading	265
Problems	266

J-DSP Laboratory Exercises.....	268
8. Discrete-Time Random Signals.....	272
8.1. Introduction	272
8.2. First and Second-Order Statistics	273
8.3. Random Signals Processed by LTI Digital Filters.....	279
8.4. Crosscorrelation and Deconvolution	285
8.5. Estimating Signal Statistics from Finite-Length Data	288
8.5.1. Correlogram Estimators of the Psd.....	289
8.5.2. Periodogram Estimators	289
8.6. Parametric Spectral Estimates	291
8.7. Effects of Roundoff Noise on Digital Filters.....	295
8.7.1. Uniform Quantization.....	295
8.7.2. Roundoff Noise in Digital Filters	297
8.7.3. Digital Filter Coefficient Quantization.....	300
8.8. Summary	303
8.8.1. Short Concept Questions	304
8.8.2. Further Reading.....	306
Problems.....	307
J-DSP Laboratory Exercises.....	309
9. Adaptive Filtering.....	314
9.1. Introduction	314
9.2. System Identification.....	315
9.3. Iterative Algorithms and the LMS.....	319
9.4. The Normalized LMS Algorithm	326
9.5. Eigenvalue Spread and Newton-Like Algorithms	326
9.5.1. The Recursive Least Squares Algorithm	328
9.6. The Block LMS Algorithm	333
9.7. Frequency-Domain Adaptive Algorithms	334

9.8.	Adaptive Noise Cancellation.....	337
9.9.	The LMS Linear Prediction Algorithm	340
9.10.	Summary.....	342
9.10.1.	Short Concept Questions	342
9.10.2.	Further Reading.....	343
	Problems.....	345
	J-DSP Laboratory Exercises: Digital Filter Simulations	347
10.	Speech Processing Algorithms	350
10.1.	Introduction	350
10.2.	Speech Properties	351
10.3.	Linear Predictive Coding Analysis-Synthesis Algorithms	355
10.4.	A Simple Speech Analysis-Synthesis System	359
10.4.1.	Two State Excitation Models	361
10.4.2.	Quantization of LPC Parameters	363
10.5.	Linear Prediction Parameter Transformations.....	364
10.6.	Long-Term Prediction	366
10.7.	Algorithms Used in Wireless Cellular Telephony.....	367
10.8.	Summary	369
10.8.1.	Short Concept Questions	370
10.8.2.	Further Reading.....	371
	Problems.....	373
	J-DSP Laboratory Exercises: LPC Simulations	374
11.	The MPEG-1 Layer III (MP3) Algorithm.....	387
11.1.	Introduction	387
11.2.	The MP3 Algorithm	388
11.3.	The MP3 Filter Bank	390
11.4.	The MDCT and the Hybrid Filter Bank	392
11.5.	Psychoacoustics and MP3	393

11.6.	Perceptual Entropy	401
11.7.	J-DSP Modules and Psychoacoustic Principles	404
11.8.	J-DSP and the MP3 Psychoacoustic Model	406
11.9.	Psychoacoustics used in DFT-Based Compression	413
11.10.	Summary	417
11.10.1.	Short Concept Questions	418
11.10.2.	Further Reading	420
Problems	421
J-DSP Laboratory Exercises: Perceptual Audio Processing	422
Appendix A – Helpful Formulas	427
A.1	Trigonometric Formulas	427
A.2	Fourier Series of Select Functions	428
A.3	Evaluation of Geometric Series	429
A.4	Matrix Formulas	430
A.5	Eigen-Analysis of Matrices	430
A.6	Special Matrices	431
Appendix B – Answers To Short Questions	433
Index	435