

ADAPTIVE FILTERS

C.F.N. COWAN P.M. GRANT

## **SUMMARY**

In this up-to-date state of the art book, the authors provide a coherent and Comprehensive introduction to adaptive filtering. They coyer basic theory, practical realizations, and applications, such as adaptive equalizers for telecommunications data transmission systems. Practical engineers find this book a good source of information on the practical possibilities of these processors.

- This book's key features include ; Chapter 2 estimation theory discusses and is followed by two chapters on.
- adaptive finite impulse response and infinite impulse response.
- Chapter 5 covers the theory, design, and application of adaptive lattice
- Chapter 6 deals with signal transformation techniques for adaptive filtering.
- Chapter 7 covers adaptive filter implementations.

Chapter 8 includes main applications in communications equalization and echo cancellation

Chapter 9 describes such application areas as fast tracking filters for HF and microwave digital radion, linear predictive coding, and maximumentropy and maximum-likelihood analysis techniques.

filters.

## **CONTENTS**

PREFACE ACKNOWLEDGMENTS ABBREVIATIONS SYMBOLS					
1	INTRODUCTION TO ADAPTIVE FILTERS				
	Peter 1.1	r M Grant and Colin F. N Cowan Adaptive Processing 1.1.1 Adaptive Filters 1.1.2 Adaptive Filter Operation	1 2 3		
	1.2	Programmable Filter Designs1.2.1Recursive Filters1.2.2Nonrecursive Filters1.2.3Transform-Based Filters	4 4 6 8		
	1.3	Optimum Linear Estimation	10		
	1.4	Adaptive Filters1.4.1Adaptive Infinite Impulse Response Filters1.4.2Adaptive Finite Impulse Response Filters1.4.3Transform-Based Adaptive Filters1.4.4Hardware Designs	11 11 11 12 14		
2	OPTIMUM ESTIMATION TECHNIQUES				
	Colin F. N. Cowan				
	2.1	Introduction	15		
	2.2	Optimum Nonrecursive (Wiener) Estimation 2.2.1 Practical Example of a Wiener Estimator	16 18		
	2.3	Optimum Recursive (Kalman) Estimation 2.3.1 Scalar Kalman Filter 2.3.2 Derivation of the Kalman Gain	21 21 24		
	2.4	Vector Kalman Filter 2.4.1 Vector Kalman Filter as a Channel Equalizer	25 26		
	2.5	Conclusions	27		

3	ADAPTIVE ALGORITHMS FOR FINITE IMPULSE RESPONSE FILTERS 29				
	Benjamin Friedlander 3.1 Introduction				
	3.2	Recursive Least-Squares Algorithm3.2.1Derivation of the RLS Algorithm3.2.2Exponentially Weighted RLS3.2.3Computational Complexity3.2.4Stochastic Interpretation3.2.5Asymptotic Accuracy of Least-Squares Estimates3.2.6Asymptotic Properties of the Adaptive Filter3.2.7Square-Root Implementation3.2.8Sliding Window Form of the RLS	30 32 33 34 35 37 38 39 40		
	3.3	Least-Mean-Squares Adaptive Algorithm3.3.1Iterative Computation of the Optimal Coefficient Vector3.3.2LMS Algorithm3.3.3Convergence of the LMS Algorithm3.3.4Learning Curve3.3.5Recent Convergence Results3.3.6LMS Algorithm as a Stochastic Approximation Method	41 42 44 46 48 49		
	3.4	Adaptive Finite Impulse Response Filters with Linear-Phase Characteristics3.4.1Stochastic Case3.4.2RLS Algorithm3.4.3LMS Algorithm	55 55 57 58		
4	ADAPT	TIVE ALGORITHMS FOR INFINITE IMPULSE RESPONSE FILTERS	60		
	John R. 4.1	<ul> <li>Treichler</li> <li>Introduction</li> <li>4.1 1 General Scope</li> <li>4.1.2 Why Use UR Adaptive Filters?</li> <li>4.1.3 Problem Formulation</li> <li>4.1.4 Implications of Feedback</li> </ul>	60 60 61 62 63		
	4.2	Minimum Mean-Square-Error Techniques4.2.1Developing Necessary Conditions for a Solution4.2.2Solution Techniques4.2.3Historical Perspective	64 64 65 67		
	4.3	<ul> <li>Techniques Based on Nonlinear Stability Theory</li> <li>4.3.1 Problem Formulation</li> <li>4.3.2 Hyperstable Adaptive Recursive Filter</li> <li>4.3.3 Hyperstability and Adaptive Filtering</li> <li>4.3.4 Simple Hyperstable Recursive Filter</li> </ul>	68 68 69 72 78		
	4.4	Convergence Analysis4.4.1Goals of Convergence Analysis4.4.2Approaches4.4.3General Conclusions	81 81 83 88		
	4.5	Limitations in the Use of UR Adaptive Filters 4.5.1 Coefficient Sensitivity 4.5.2 Inverse Modeling of Non-Minimum-Phase Filters 4.5.3 Order Matching	89 89 89 90		
		4.5.4 Conversion of Stability-Based Techniques to Inverse Modeling	90		
5	4.6 RECUE	CONCIUSION	90 91		
5	Tohn M Turner 5.1 Introduction				
	5.2 5.3	General Lattice Digital Filter StructureProperties of the Lattice Structure5.3.1Orthogonalizing Properties5.3.2Physical Interpretation	93 98 99 102		
	5.4	Sample Data Estimates of Reflection Coefficients 5.4.1 Gradient Estimates of Reflection Coefficients	105 106		
	5.5	Recursive Least-Squares Lattice Algorithm5.5.1Formulation of Recursive Estimates5.5.2Order-Update Equations5.5.3Time-Update Equations5.5.4Exact Least-Squares Lattice Recursions5.5.5Likelihood Variable	108 109 111 113 114 116		

	5.0	שוורד וטנכסס במנוונכ ו וונכו	111
	5.7	Square-Root Normalized Least-Squares Lattice	120
	5.8	Computational Complexity and CORDIC Arithmetic 5.8.1 CORDIC Arithmetic 5.8.2 Lattice Filtering by Rotations	123 124 126
	5.9 5.10	Simulations and Applications Comments and Conclusions	129 143
6	FREQ	QUENCY-DOMAIN ADAPTIVE FILTERING	145
	Farl F	R Ferrara Ir	
	6.1	Introduction	145
	6.2	Frequency-Domain Adaptive Filter Based on Circular Convolution	146
	6.3	Algorithms for General Adaptive Filtering 6.3.1 Fast LMS Adaptive Filter 6.3.2 Unconstrained Frequency-Domain LMS Adaptive Filter	152 152 157
	6.4	Channel Equalization 6.4.1 Isolated Pulse Equalization 6.4.2 Random Data Sequence Equalization	160 160 164
	6.5	Transmultiplexer Adaptive Filter	164
	6.6	Convergence Rate Improvement	172
	6.7	Summary	176
	6.8	Appendix: Linear versus Circular Convolution	177
7	SUR\	VEY OF ANALOG AND DIGITAL ADAPTIVE FILTER REALIZATIONS	180
	Colin 7.1	F. N. Cowan and Peter M Grant Introduction	180
	7.2	Digital Implementations	181
		7.2.1 Classical Digital Design	181
		<ul> <li>7.2.2 Digital Adaptive Filters Using Simplified Algorithms</li> <li>7.2.3 Digital Adaptive Filters Using Memory Access Techniques</li> </ul>	187 190
		7.2.4 Distributed Arithmetic Adaptive Filters 7.2.5 Residue Number Systems	190 195
	7 2	Analog Sampled Data Adaptive Eilters	100
	7.5	7.3.1 Charge-Coupled-Device Implementations	199
		7.3.2 Monolithic CCD Adaptive Filter	206
	7.4	High-Bandwidth Adaptive Filters Using Surface Acoustic Wave Devices	210
	7.5	Future Designs Using VLSI Technology	214
8	ADAF Peter	PTIVE FILTERS IN TELECOMMUNICATIONS	216
	8.1	Introduction	216
	8.2	Data Transmission	217
		8.2.1 Linear Distortions in Telephony Networks	218
		8.2.3 Echo Cancellation for Speech-Band Data Transmission	232
	8.3	Digital Transmission over Local Networks	241
		8.3.1 Echo Cancellation for WAL2 Transmission	244
	8.4	Echo Cancellation for Telephony	240
		8.4.1 Network Echo Cancelers	252
	0 5	8.4.2 Terminal Echo Cancelers	254
0	0.5		200
9	Data		257
	Peter 9.1	Introduction	257
	9.2	Adaptive Estimation	258
		9.2.1 Inverse System Modeling	258
		9.2.2 Direct System modeling	264
	9.3	Spectral Estimation	266
		9.3.1 Introduction	266
		9.3.2 Spectral Line Ennancement 9.3.3 Speech Processing	270 272
			212

	9.4	Adaptive Array Processing 9.4.1 Bearing Estimation	276 277
	9.5	Summary	282
REFERENCES			283
INDEX			303

TOP

The term adaptive filter implies changing the characteristic of a filter in some automated fashion to obtain the best possible signal quality in spite of changing signal/system conditions. Adaptive filters are usually associated with the broader topic of statistical signal processing. The operation of signal filtering by definition implies extracting something desired from a signal containing both desired and undesired components. 28 chapter 3. applications of adaptive filters. Echo canceller. Figure 3.11. Hopefully, the adapting Iter will restore that property by removing the distortion or interference! Example 4.1: the Constant-Modulus Algorithm (CMA). Certain communication modulation schemes, such as PSK and FSK, transmit a sinusoid of a constant analytic magnitude. Only the frequency or phase change with time. The constant modulus algorithm tries to drive the output signal to one having a constant amplitude