AUDIO SIGNAL PROCESSING FOR NEXT-GENERATION MULTIMEDIA COMMUNICATION SYSTEMS

Edited by
YITENG (ARDEN) HUANG
Bell Laboratories, Lucent Technologies

JACOB BENESTY
Université du Québec, INRS-EMT

Kluwer Academic Publishers
Boston/Dordrecht/London
## Contents

Preface xi  
Contributing Authors xiii  

1 
Introduction 1  

*Yiteng (Arden) Huang Jacob Benesty*

1. Multimedia Communications 1  
2. Challenges and Opportunities 3  
3. Organization of the Book 4  

**Part I Speech Acquisition and Enhancement**

2  
Differential Microphone Arrays 11  

*Gary W. Elko*

1. Introduction 11  
2. Differential Microphone Arrays 12  
3. Array Directional Gain 22  
4. Optimal Arrays for Isotropic Fields 24  
   4.1 Maximum Directional Gain 24  
   4.2 Maximum Directivity Index for Differential Microphones 28  
   4.3 Maximum Front-to-Back Ratio 32  
   4.4 Minimum Peak Directional Response 37  
   4.5 Beamwidth 39  
5. Design Examples 39  
   5.1 First-Order Designs 40  
   5.2 Second-Order Designs 44  
   5.3 Third-Order Designs 52  
   5.4 Higher-Order designs 58  
6. Sensitivity to Microphone Mismatch and Noise 60  
7. Conclusions 64
Spherical Microphone Arrays for 3D Sound Recording

Jens Meyer Gary W. Elko

1. Introduction
2. Fundamental Concept
3. The Eigenbeamformer
   3.1 Discrete Orthonormality
   3.2 The Eigenbeams
   3.3 The Modal Coefficients
4. Modal-Beamformer
   4.1 Combining Unit
   4.2 Steering Unit
5. Robustness Measure
6. Beampattern Design
   6.1 Arbitrary Beampattern Design
   6.2 Optimum Beampattern Design
7. Measurements
8. Summary
9. Appendix A

Subband Noise Reduction Methods for Speech Enhancement

Eric J. Diethorn

1. Introduction
2. Wiener Filtering
3. Speech Enhancement by Short-Time Spectral Modification
   3.1 Short-Time Fourier Analysis and Synthesis
   3.2 Short-Time Wiener Filter
   3.3 Power Subtraction
   3.4 Magnitude Subtraction
   3.5 Parametric Wiener Filtering
   3.6 Review and Discussion
4. Averaging Techniques for Envelope Estimation
   4.1 Moving Average
   4.2 Single-Pole Recursion
   4.3 Two-Sided Single-Pole Recursion
   4.4 Nonlinear Data Processing
5. Example Implementation
   5.1 Subband Filter Bank Architecture
   5.2 A-Posteriori-SNR Voice Activity Detector
   5.3 Example
6. Conclusion

Part II Acoustic Echo Cancellation

Adaptive Algorithms for MIMO Acoustic Echo Cancellation

Jacob Benesty Tomas Gänslер Yiteng (Arden) Huang Markus Rupp

1. Introduction
2. Normal Equations and Identification of a MIMO System
   2.1 Normal Equations
Contents

2.2 The Nonuniqueness Problem 124
2.3 The Impulse Response Tail Effect 125
2.4 Some Different Solutions for Decorrelation 126
3. The Classical and Factorized Multichannel RLS 128
4. The Multichannel Fast RLS 130
5. The Multichannel LMS Algorithm 132
5.1 Classical Derivation 132
5.2 Improved Version 133
6. The Multichannel APA 134
6.1 The Straightforward Multichannel APA 134
6.2 The Improved Two-Channel APA 135
6.3 The Improved Multichannel APA 136
7. The Multichannel Exponentiated Gradient Algorithm 137
8. The Multichannel Frequency-domain Adaptive Algorithm 142
9. Conclusions 145

6
Double-Talk Detectors for Acoustic Echo Cancelers 149
Tomas Gänsler Jacob Benesty
1. Introduction 149
2. Basics of AEC and DTD 152
2.1 AEC Notations 152
2.2 The Generic DTD 152
2.3 A Suggestion to Performance Evaluation of DTDs 153
3. Double-Talk Detection Algorithms 154
3.1 The Geigel Algorithm 154
3.2 The Cross-Correlation Method 154
3.3 The Normalized Cross-Correlation Method 155
3.4 The Coherence Method 157
3.5 The Normalized Cross-correlation Matrix 159
3.6 The Two-Path Model 161
3.7 DTD Combinations with Robust Statistics 163
4. Comparison of DTDs by Means of the ROC 165
5. Discussion 167

7
The WinEC: A Real-Time Hands-Free Stereo Communication System 171
Tomas Gänsler Volker Fischer Eric J. Diethorn Jacob Benesty
1. Introduction 172
1.1 Signal model 173
2. System Description 173
2.1 The Audio Module 173
2.2 The Network Module 176
2.3 The Echo Canceler Module 177
3. Algorithms of the Echo Canceler Module 177
3.1 Adaptive Filter Algorithm 178
4. Residual Echo and Noise Suppression 181
4.1 Masking Threshold for Residual Echo in Noise 183
4.2 Analysis of Echo Suppression Requirements 184
4.3 Noise and Residual Echo Suppression 186
5. Simulations 186
6. Real-Time Tests with Different Modes of Operation 189
6.1 Point-to-Point Communication 189
6.2 Multi-Point Communication 189
6.3 Transatlantic Teleconference in Stereo 190
7. Discussion 191

Part III Sound Source Tracking and Separation

8
Time Delay Estimation 197
Jingdong Chen Yiteng (Arden) Huang Jacob Benesty
1. Introduction 198
2. Signal Models 200
   2.1 Ideal Propagation Model 200
   2.2 Multipath Model 201
   2.3 Reverberant Model 202
4. The Multichannel Cross-Correlation Algorithm 204
   4.1 Spatial Prediction Technique 204
   4.2 Time Delay Estimation Using Spatial Prediction 207
   4.3 Other Information from the Spatial Correlation Matrix 208
5. Adaptive Eigenvalue Decomposition Algorithm 211
6. Adaptive Multichannel Time Delay Estimation 213
   6.1 Principle 213
   6.2 Time-Domain Multichannel LMS Approach 214
   6.3 Frequency-Domain Adaptive Algorithms 215
7. Experiments 219
   7.1 Experimental Setup 219
   7.2 Performance Measure 220
   7.3 Experimental Results 221
8. Conclusions 223

9
Source Localization 229
Yiteng (Arden) Huang Jacob Benesty Gary W. Elko
1. Introduction 230
2. Source Localization Problem 232
   zation 234
4. Maximum Likelihood Estimator 235
5. Least Squares Estimators 236
   5.1 The Least Squares Error Criteria 237
   5.2 Spherical Intersection (SX) Estimator 239
   5.3 Spherical Interpolation (SI) Estimator 239
   5.4 Linear-Correction Least Squares Estimator 240
6. Example System Implementation 246
7. Source Localization Examples 247
8. Conclusions 249

10
Blind Source Separation for Convolutive Mixtures: A Unified Treatment 255
Herbert Buchner Robert Aichner Walter Kellermann