

Contents

Part I. Speech Enhancement

1 Constant Directivity Beamforming

<i>Darren B. Ward, Rodney A. Kennedy, Robert C. Williamson</i>	3
1.1 Introduction	3
1.2 Problem Formulation	6
1.3 Theoretical Solution	7
1.3.1 Continuous sensor	7
1.3.2 Beam-shaping function	8
1.4 Practical Implementation	9
1.4.1 Dimension-reducing parameterization	9
1.4.2 Reference beam-shaping filter	11
1.4.3 Sensor placement	12
1.4.4 Summary of implementation	12
1.5 Examples	13
1.6 Conclusions	16
References	16

2 Superdirective Microphone Arrays

<i>Joerg Bitzer, K. Uwe Simmer</i>	19
2.1 Introduction	19
2.2 Evaluation of Beamformers	20
2.2.1 Array-Gain	21
2.2.2 Beampattern	22
2.2.3 Directivity	23
2.2.4 Front-to-Back Ratio	24
2.2.5 White Noise Gain	24
2.3 Design of Superdirective Beamformers	24
2.3.1 Delay-and-Sum Beamformer	26
2.3.2 Design for spherical isotropic noise	26
2.3.3 Design for Cylindrical Isotropic Noise	30
2.3.4 Design for an Optimal Front-to-Back Ratio	30
2.3.5 Design for Measured Noise Fields	32
2.4 Extensions and Details	33
2.4.1 Alternative Form	33

2.4.2 Comparison with Gradient Microphones	35
2.5 Conclusion	36
References	37
3 Post-Filtering Techniques	
<i>K. Uwe Simmer, Joerg Bitzer, Claude Marro</i>	39
3.1 Introduction	39
3.2 Multi-channel Wiener Filtering in Subbands	41
3.2.1 Derivation of the Optimum Solution	41
3.2.2 Factorization of the Wiener Solution	42
3.2.3 Interpretation	45
3.3 Algorithms for Post-Filter Estimation	46
3.3.1 Analysis of Post-Filter Algorithms	47
3.3.2 Properties of Post-Filter Algorithms	49
3.3.3 A New Post-Filter Algorithm	50
3.4 Performance Evaluation	51
3.4.1 Simulation System	52
3.4.2 Objective Measures	52
3.4.3 Simulation Results	54
3.5 Conclusion	57
4 Spatial Coherence Functions for Differential Microphones in Isotropic Noise Fields	
<i>Gary W. Elko</i>	61
4.1 Introduction	61
4.2 Adaptive Noise Cancellation	61
4.3 Spherically Isotropic Coherence	65
4.4 Cylindrically Isotropic Fields	73
4.5 Conclusions	77
References	84
5 Robust Adaptive Beamforming	
<i>Osamu Hoshuyama, Akihiko Sugiyama</i>	87
5.1 Introduction	87
5.2 Adaptive Beamformers	88
5.3 Robustness Problem in the GJBF	90
5.4 Robust Adaptive Microphone Arrays — Solutions to Steering- Vector Errors	92
5.4.1 LAF-LAF Structure	92
5.4.2 CCAF-LAF Structure	94
5.4.3 CCAF-NCAF Structure	95
5.4.4 CCAF-NCAF Structure with an AMC	97
5.5 Software Evaluation of a Robust Adaptive Microphone Array	99
5.5.1 Simulated Anechoic Environment	99
5.5.2 Reverberant Environment	101

5.6	Hardware Evaluation of a Robust Adaptive Microphone Array . . .	104
5.6.1	Implementation	104
5.6.2	Evaluation in a Real Environment	104
5.7	Conclusion	106
	References	106

6 GSVD-Based Optimal Filtering for Multi-Microphone Speech Enhancement

	<i>Simon Doclo, Marc Moonen</i>	111
6.1	Introduction	111
6.2	GSVD-Based Optimal Filtering Technique	113
6.2.1	Optimal Filter Theory	114
6.2.2	General Class of Estimators	116
6.2.3	Symmetry Properties for Time-Series Filtering	117
6.3	Performance of GSVD-Based Optimal Filtering	118
6.3.1	Simulation Environment	118
6.3.2	Spatial Directivity Pattern	119
6.3.3	Noise Reduction Performance	121
6.3.4	Robustness Issues	121
6.4	Complexity Reduction	122
6.4.1	Linear Algebra Techniques for Computing GSVD	122
6.4.2	Recursive and Approximate GSVD-Updating Algorithms . . .	123
6.4.3	Downsampling Techniques	125
6.4.4	Simulations	125
6.4.5	Computational Complexity	126
6.5	Combination with ANC Postprocessing Stage	127
6.5.1	Creation of Speech and Noise References	127
6.5.2	Noise Reduction Performance of ANC Postprocessing Stage .	128
6.5.3	Comparison with Standard Beamforming Techniques	129
6.6	Conclusion	129
	References	130

7 Explicit Speech Modeling for Microphone Array Speech Acquisition

	<i>Michael Brandstein, Scott Griebel</i>	133
7.1	Introduction	133
7.2	Model-Based Strategies	136
7.2.1	Example 1: A Frequency-Domain Model-Based Algorithm . .	137
7.2.2	Example 2: A Time-Domain Model-Based Algorithm	140
7.3	Conclusion	148
	References	151

8 Robust Localization in Reverberant Rooms

<i>Joseph H. DiBiase, Harvey F. Silverman, Michael S. Brandstein</i>	157
8.1 Introduction	157
8.2 Source Localization Strategies	158
8.2.1 Steered-Beamformer-Based Locators	159
8.2.2 High-Resolution Spectral-Estimation-Based Locators	160
8.2.3 TDOA-Based Locators	161
8.3 A Robust Localization Algorithm	164
8.3.1 The Impulse Response Model	164
8.3.2 The GCC and PHAT Weighting Function	166
8.3.3 ML TDOA-Based Source Localization	167
8.3.4 SRP-Based Source Localization	169
8.3.5 The SRP-PHAT Algorithm	170
8.4 Experimental Comparison	172
References	178

9 Multi-Source Localization Strategies

<i>Elio D. Di Claudio, Raffaele Parisi</i>	181
9.1 Introduction	181
9.2 Background	184
9.2.1 Array Signal Model	184
9.2.2 Incoherent Approach	185
9.2.3 Coherent Signal Subspace Method (CSSM)	185
9.2.4 Wideband Weighted Subspace Fitting (WB-WSF)	186
9.3 The Issue of Coherent Multipath in Array Processing	187
9.4 Implementation Issues	188
9.5 Linear Prediction-ROOT-MUSIC TDOA Estimation	189
9.5.1 Signal Pre-Whitening	189
9.5.2 An Approximate Model for Multiple Sources in Reverberant Environments	191
9.5.3 Robust TDOA Estimation via ROOT-MUSIC	192
9.5.4 Estimation of the Number of Relevant Reflections	194
9.5.5 Source Clustering	195
9.5.6 Experimental Results	196
References	198

10 Joint Audio-Video Signal Processing for Object Localization and Tracking

<i>Norbert Strobel, Sascha Spors, Rudolf Rabenstein</i>	203
10.1 Introduction	203
10.2 Recursive State Estimation	205
10.2.1 Linear Kalman Filter	206
10.2.2 Extended Kalman Filter due to a Measurement Nonlinearity	210
10.2.3 Decentralized Kalman Filter	212
10.3 Implementation	218

10.3.1 System description	218
10.3.2 Results	219
10.4 Discussion and Conclusions	221
References	222

Part III. Applications

11 Microphone-Array Hearing Aids

<i>Julie E. Greenberg, Patrick M. Zurek</i>	229
11.1 Introduction	229
11.2 Implications for Design and Evaluation	230
11.2.1 Assumptions Regarding Sound Sources	230
11.2.2 Implementation Issues	231
11.2.3 Assessing Performance	232
11.3 Hearing Aids with Directional Microphones	233
11.4 Fixed-Beamforming Hearing Aids	234
11.5 Adaptive-Beamforming Hearing Aids	235
11.5.1 Generalized Sidelobe Canceler with Modifications	236
11.5.2 Scaled Projection Algorithm	242
11.5.3 Direction of Arrival Estimation	243
11.5.4 Other Adaptive Approaches and Devices	243
11.6 Physiologically-Motivated Algorithms	244
11.7 Beamformers with Binaural Outputs	245
11.8 Discussion	246
References	249

12 Small Microphone Arrays with Postfilters for Noise and Acoustic Echo Reduction

<i>Rainer Martin</i>	255
12.1 Introduction	255
12.2 Coherence of Speech and Noise	257
12.2.1 The Magnitude Squared Coherence	257
12.2.2 The Reverberation Distance	258
12.2.3 Coherence of Noise and Speech in Reverberant Enclosures ..	259
12.3 Analysis of the Wiener Filter with Symmetric Input Signals	263
12.3.1 No Near End Speech	265
12.3.2 High Signal to Noise Ratio	265
12.4 A Noise Reduction Application	266
12.4.1 An Implementation Based on the NLMS Algorithm	266
12.4.2 Processing in the 800 – 3600 Hz Band	268
12.4.3 Processing in the 240 – 800 Hz Band	269
12.4.4 Evaluation	269
12.4.5 Alternative Implementations of the Coherence Based Postfilter	271
12.5 Combined Noise and Acoustic Echo Reduction	271

12.5.1 Experimental Results	274
12.6 Conclusions	275
References	276

13 Acoustic Echo Cancellation for Beamforming

Microphone Arrays

<i>Walter L. Kellermann</i>	281
13.1 Introduction	281
13.2 Acoustic Echo Cancellation	282
13.2.1 Adaptation algorithms	284
13.2.2 AEC for multi-channel sound reproduction	287
13.2.3 AEC for multi-channel acquisition	287
13.3 Beamforming	288
13.3.1 General structure	288
13.3.2 Time-invariant beamforming	290
13.3.3 Time-varying beamforming	291
13.3.4 Computational complexity	292
13.4 Generic structures for combining AEC with beamforming	292
13.4.1 Motivation	292
13.4.2 Basic options	293
13.4.3 'AEC first'	293
13.4.4 'Beamforming first'	296
13.5 Integration of AEC into time-varying beamforming	297
13.5.1 Cascading time-invariant and time-varying beamforming ...	297
13.5.2 AEC with GSC-type beamforming structures	301
13.6 Combined AEC and beamforming for multi-channel recording and multi-channel reproduction	302
13.7 Conclusions	303
References	303

14 Optimal and Adaptive Microphone Arrays for Speech Input in Automobiles

<i>Sven Nordholm, Ingvar Claesson, Nedelko Grbić</i>	307
14.1 Introduction: Hands-Free Telephony in Cars	307
14.2 Optimum and Adaptive Beamforming	309
14.2.1 Common Signal Modeling	309
14.2.2 Constrained Minimum Variance Beamforming and the Gen- eralized Sidelobe Canceler	310
14.2.3 <i>In Situ</i> Calibrated Microphone Array (ICMA)	312
14.2.4 Time-Domain Minimum-Mean-Square-Error Solution	313
14.2.5 Frequency-Domain Minimum-Mean-Square-Error Solution ..	314
14.2.6 Optimal Near-Field Signal-to-Noise plus Interference Beam- former	316
14.3 Subband Implementation of the Microphone Array	317
14.3.1 Description of LS-Subband Beamforming	318

14.4 Multi-Resolution Time-Frequency Adaptive Beamforming	319
14.4.1 Memory Saving and Improvements	319
14.5 Evaluation and Examples	320
14.5.1 Car Environment	320
14.5.2 Microphone Configurations	321
14.5.3 Performance Measures	321
14.5.4 Spectral Performance Measures	322
14.5.5 Evaluation on car data	323
14.5.6 Evaluation Results	323
14.6 Summary and Conclusions	324
References	326
15 Speech Recognition with Microphone Arrays	
<i>Maurizio Omologo, Marco Matassoni, Piergiorgio Svaizer</i>	331
15.1 Introduction	331
15.2 State of the Art	332
15.2.1 Automatic Speech Recognition	332
15.2.2 Robustness in ASR	336
15.2.3 Microphone Arrays and Related Processing for ASR	337
15.2.4 Distant-Talker Speech Recognition	339
15.3 A Microphone Array-Based ASR System	342
15.3.1 System Description	342
15.3.2 Speech Corpora and Task	345
15.3.3 Experiments and Results	346
15.4 Discussion and Future Trends	348
References	349
16 Blind Separation of Acoustic Signals	
<i>Scott C. Douglas</i>	355
16.1 Introduction	355
16.1.1 The Cocktail Party Effect	355
16.1.2 Chapter Overview	356
16.2 Blind Signal Separation of Convolutional Mixtures	357
16.2.1 Problem Structure	357
16.2.2 Goal of Convolutional BSS	359
16.2.3 Relationship to Other Problems	360
16.3 Criteria for Blind Signal Separation	362
16.3.1 Overview of BSS Criteria	362
16.3.2 Density Modeling Criteria	362
16.3.3 Contrast Functions	364
16.3.4 Correlation-Based Criteria	366
16.4 Structures and Algorithms for Blind Signal Separation	367
16.4.1 Filter Structures	367
16.4.2 Density Matching BSS Using Natural Gradient Adaptation ..	368
16.4.3 Contrast-Based BSS Under Prewhitening Constraints	370

16.4.4 Temporal Decorrelation BSS for Nonstationary Sources	372
16.5 Numerical Evaluations	373
16.6 Conclusions and Open Issues	375
References	378

Part IV. Open Problems and Future Directions

17 Future Directions for Microphone Arrays	
<i>Gary W. Elko</i>	383
17.1 Introduction	383
17.2 Hands-Free Communication	383
17.3 The "Future" of Microphone Array Processing	385
17.4 Conclusions	387
18 Future Directions in Microphone Array Processing	
<i>Dirk Van Compernelle</i>	389
18.1 Lessons From the Past	389
18.2 A Future Focused on Applications	391
18.2.1 Automotive	391
18.2.2 Desktop	392
18.2.3 Hearing Aids	393
18.2.4 Teleconferencing	393
18.2.5 Very Large Arrays	393
18.2.6 The Signal Subspace Approach - An Alternative to Spatial Filtering ?	393
18.3 Final Remarks	394
Index	395

Microphone Arrays

Signal Processing Techniques and Applications

Brandstein, M.; Ward, D. (Eds.)

2001, XVIII, 398 p., Hardcover

ISBN: 978-3-540-41953-2