

Jacob Benesty · Jingdong Chen · Yiteng Huang

Microphone Array Signal Processing

Contents

1	Introduction	1
1.1	Microphone Array Signal Processing	1
1.2	Organization of the Book	5
2	Classical Optimal Filtering	7
2.1	Introduction	7
2.2	Wiener Filter	8
2.3	Frost Filter	16
2.3.1	Algorithm	16
2.3.2	Generalized Sidelobe Canceller Structure	17
2.3.3	Application to Linear Interpolation	19
2.4	Kalman Filter	21
2.5	A Viable Alternative to the MSE	25
2.5.1	Pearson Correlation Coefficient	26
2.5.2	Important Relations with the SPCC	26
2.5.3	Examples of Optimal Filters Derived from the SPCC	29
2.6	Conclusions	37
3	Conventional Beamforming Techniques	39
3.1	Introduction	39
3.2	Problem Description	40
3.3	Delay-and-Sum Technique	41
3.4	Design of a Fixed Beamformer	46
3.5	Maximum Signal-to-Noise Ratio Filter	49
3.6	Minimum Variance Distortionless Response Filter	52
3.7	Approach with a Reference Signal	54
3.8	Response-Invariant Broadband Beamformers	55
3.9	Null-Steering Technique	58
3.10	Microphone Array Pattern Function	61
3.10.1	First Signal Model	62
3.10.2	Second Signal Model	64

3.11	Conclusions	65
4	On the Use of the LCMV Filter in Room Acoustic Environments	67
4.1	Introduction	67
4.2	Signal Models	67
4.2.1	Anechoic Model	68
4.2.2	Reverberant Model	68
4.2.3	Spatio-Temporal Model	69
4.3	The LCMV Filter with the Anechoic Model	69
4.4	The LCMV Filter with the Reverberant Model	73
4.5	The LCMV Filter with the Spatio-Temporal Model	75
4.5.1	Experimental Results	78
4.6	The LCMV Filter in the Frequency Domain	81
4.7	Conclusions	83
5	Noise Reduction with Multiple Microphones: a Unified Treatment	85
5.1	Introduction	85
5.2	Signal Model and Problem Description	86
5.3	Some Useful Definitions	87
5.4	Wiener Filter	89
5.5	Subspace Method	92
5.6	Spatio-Temporal Prediction Approach	95
5.7	Case of Perfectly Coherent Noise	97
5.8	Adaptive Noise Cancellation	99
5.9	Kalman Filter	100
5.10	Simulations	101
5.10.1	Acoustic Environments and Experimental Setup	101
5.10.2	Experimental Results	103
5.11	Conclusions	114
6	Noncausal (Frequency-Domain) Optimal Filters	115
6.1	Introduction	115
6.2	Signal Model and Problem Formulation	116
6.3	Performance Measures	117
6.4	Noncausal Wiener Filter	120
6.5	Parametric Wiener Filtering	124
6.6	Generalization to the Multichannel Case	126
6.6.1	Signal Model	126
6.6.2	Definitions	128
6.6.3	Multichannel Wiener Filter	129
6.6.4	Spatial Maximum SNR Filter	132
6.6.5	Minimum Variance Distortionless Response Filter	134
6.6.6	Distortionless Multichannel Wiener Filter	135

6.7	Conclusions	136
7	Microphone Arrays from a MIMO Perspective	139
7.1	Introduction	139
7.2	Signal Models and Problem Description	140
7.2.1	SISO Model	141
7.2.2	SIMO Model	141
7.2.3	MISO Model	142
7.2.4	MIMO Model	143
7.2.5	Problem Description	144
7.3	Two-Element Microphone Array	144
7.3.1	Least-Squares Approach	145
7.3.2	Frost Algorithm	146
7.3.3	Generalized Sidelobe Canceller Structure	148
7.4	N -Element Microphone Array	150
7.4.1	Least-Squares and MINT Approaches	150
7.4.2	Frost Algorithm	152
7.4.3	Generalized Sidelobe Canceller Structure	154
7.4.4	Minimum Variance Distortionless Response Approach ..	156
7.5	Simulations	156
7.5.1	Acoustic Environments and Experimental Setup	156
7.6	Conclusions	163
8	Sequential Separation and Dereverberation: the	
	Two-Stage Approach	165
8.1	Introduction	165
8.2	Signal Model and Problem Description	165
8.3	Source Separation	168
8.3.1	2×3 MIMO System	168
8.3.2	$M \times N$ MIMO System	172
8.4	Speech Dereverberation	175
8.4.1	Direct Inverse	175
8.4.2	Minimum Mean-Square Error and Least-Squares Methods	177
8.4.3	MINT Method	177
8.5	Conclusions	180
9	Direction-of-Arrival and Time-Difference-of-Arrival	
	Estimation	181
9.1	Introduction	181
9.2	Problem Formulation and Signal Models	184
9.2.1	Single-Source Free-Field Model	184
9.2.2	Multiple-Source Free-Field Model	185
9.2.3	Single-Source Reverberant Model	186
9.2.4	Multiple-Source Reverberant Model	187

9.3	Cross-Correlation Method	188
9.4	The Family of the Generalized Cross-Correlation Methods	190
9.4.1	Classical Cross-Correlation	191
9.4.2	Smoothed Coherence Transform	191
9.4.3	Phase Transform	192
9.5	Spatial Linear Prediction Method	193
9.6	Multichannel Cross-Correlation Coefficient Algorithm	196
9.7	Eigenvector-Based Techniques	200
9.7.1	Narrowband MUSIC	201
9.7.2	Broadband MUSIC	203
9.8	Minimum Entropy Method	205
9.8.1	Gaussian Source Signal	205
9.8.2	Speech Source Signal	206
9.9	Adaptive Eigenvalue Decomposition Algorithm	207
9.10	Adaptive Blind Multichannel Identification Based Methods . . .	209
9.11	TDOA Estimation of Multiple Sources	211
9.12	Conclusions	215
10	Unaddressed Problems	217
10.1	Introduction	217
10.2	Speech Source Number Estimation	217
10.3	Cocktail Party Effect and Blind Source Separation	218
10.4	Blind MIMO Identification	220
10.5	Conclusions	222
	References	223
	Index	237