

Contents

Abbreviations and Acronyms	1
1 Introduction	
E. Hänsler, G. Schmidt	
1.1 Overview about the Book	8

Part I Speech Enhancement

2 Low Delay Filter-Banks for Speech and Audio Processing

H. W. Löllmann, P. Vary	
2.1 Introduction	13
2.2 Analysis-Synthesis Filter-Banks	15
2.2.1 General Structure	15
2.2.2 Tree-Structured Filter-Banks	16
2.2.3 Modulated Filter-Banks	17
2.2.4 Frequency Warped Filter-Banks	20
2.2.5 Low Delay Filter-Banks	26
2.3 The Filter-Bank Equalizer	29
2.3.1 Concept	29
2.3.2 Prototype Filter Design	31
2.3.3 Relation between GDFT and GDCT	33
2.3.4 Realization for Different Filter Structures	35
2.3.5 Polyphase Network Implementation	37
2.3.6 The Non-Uniform Filter-Bank Equalizer	41
2.3.7 Comparison between FBE and AS FB	43
2.3.8 Algorithmic Complexity	43
2.4 Further Measures for Signal Delay Reduction	44
2.4.1 Concept	45

2.4.2	Approximation by a Moving-Average Filter	45
2.4.3	Approximation by an Auto-Regressive Filter	46
2.4.4	Algorithmic Complexity	47
2.4.5	Warped Filter Approximation	48
2.5	Application to Noise Reduction	49
2.5.1	System Configurations	49
2.5.2	Instrumental Quality Measures	50
2.5.3	Simulation Results for the Uniform Filter-Banks	51
2.5.4	Simulation Results for the Warped Filter-Banks	53
2.6	Conclusions	55
	References	56

3 A Pre-Filter for Hands-Free Car Phone Noise Reduction: Suppression of Harmonic Engine Noise Components

H. Puder

3.1	Introduction	63
3.2	Analysis of the Different Car Noise Components	64
3.2.1	Wind Noise	65
3.2.2	Tire Noise	65
3.2.3	Engine Noise	66
3.3	Engine Noise Removal Based on Notch Filters	68
3.4	Compensation of Engine Harmonics with Adaptive Filters	73
3.4.1	Step-Size Control	75
3.4.2	Calculating the Optimal Step-Size	78
3.4.3	Results of the Compensation Approach	80
3.5	Evaluation and Comparison of the Results Obtained by the Notch Filter and the Compensation Approach	84
3.6	Conclusions and Summary	85
3.6.1	Conclusion	85
3.6.2	Summary	86
	References	87

4 Model-Based Speech Enhancement

M. Krini, G. Schmidt

4.1	Introduction	89
4.2	Conventional Speech Enhancement Schemes	91
4.3	Speech Enhancement Schemes Based on Nonlinearities	93
4.4	Speech Enhancement Schemes Based on Speech Reconstruction	97
4.4.1	Feature Extraction and Control	99
4.4.2	Reconstruction of Speech Signals	110
4.5	Combining the Reconstructed and the Noise Suppressed Signal	124
4.5.1	Adding the Fully Reconstructed Signal	125
4.5.2	Adding only the Voiced Part of the Reconstructed Signal	129

4.6	Summary and Outlook	133
	References	133

5 Bandwidth Extension of Telephony Speech

B. Iser, G. Schmidt

5.1	Introduction	135
5.2	Organization of the Chapter	137
5.3	Basics	138
5.3.1	Human Speech Generation	139
5.3.2	Source-Filter Model	141
5.3.3	Parametric Representations of the Spectral Envelope	143
5.3.4	Distance Measures	147
5.4	Non-Model-Based Algorithms for Bandwidth Extension	149
5.4.1	Oversampling with Imaging	149
5.4.2	Spectral Shifting	151
5.4.3	Application of Non-Linear Characteristics	153
5.5	Model-Based Algorithms for Bandwidth Extension	153
5.5.1	Generation of the Excitation Signal	155
5.5.2	Vocal Tract Transfer Function Estimation	159
5.6	Evaluation of Bandwidth Extension Algorithms	176
5.6.1	Objective Distance Measures	177
5.6.2	Subjective Measures	180
5.7	Conclusions	181
	References	182

6 Dereverberation and Residual Echo Suppression in Noisy Environments

E. A. P. Habets, S. Gannot, I. Cohen

6.1	Introduction	186
6.2	Problem Formulation	188
6.3	OM-LSA Estimator for Multiple Interferences	191
6.3.1	OM-LSA Estimator	191
6.3.2	A priori SIR Estimator	193
6.4	Dereverberation of Noisy Speech Signals	195
6.4.1	Short Introduction to Speech Dereverberation	195
6.4.2	Problem Formulation	197
6.4.3	Statistical Reverberation Model	199
6.4.4	Late Reverberant Spectral Variance Estimator	200
6.4.5	Summary and Discussion	203
6.5	Residual Echo Suppression	203
6.5.1	Problem Formulation	204
6.5.2	Late Residual Echo Spectral Variance Estimator	206
6.5.3	Parameter Estimation	208
6.5.4	Summary	210

6.6	Joint Suppression of Reverberation, Residual Echo, and Noise	210
6.7	Experimental Results	212
6.7.1	Experimental Setup	214
6.7.2	Joint Suppression of Reverberation and Noise	214
6.7.3	Suppression of Residual Echo	216
6.7.4	Joint Suppression of Reverberation, Residual Echo, and Noise	221
6.8	Summary and Outlook	223
	References	224

7 Low Distortion Noise Cancellers – Revival of a Classical Technique

A. Sugiyama

7.1	Introduction	229
7.2	Distortions in Widrow’s Adaptive Noise Canceller	230
7.2.1	Distortion by Interference	230
7.2.2	Distortion by Crosstalk	232
7.3	Paired Filter (PF) Structure	233
7.3.1	Algorithm	233
7.3.2	Evaluations	235
7.4	Crosstalk Resistant ANC and Cross-Coupled Structure	239
7.4.1	Crosstalk Resistant ANC	240
7.4.2	Cross-Coupled Structure	241
7.5	Cross-Coupled Paired Filter (CCPF) Structure	242
7.5.1	Algorithm	242
7.5.2	Evaluations	245
7.6	Generalized Cross-Coupled Paired Filter (GCCPF) Structure	247
7.6.1	Algorithm	250
7.6.2	Evaluation by Recorded Signals	251
7.7	Demonstration in a Personal Robot	261
7.8	Conclusions	261
	References	263

Part II Echo Cancellation

8 Nonlinear Echo Cancellation Based on Spectral Shaping

O. Hoshuyama, A. Sugiyama

8.1	Introduction	267
8.2	Frequency-Domain Model of Highly Nonlinear Residual Echo	268
8.2.1	Spectral Correlation Between Residual Echo and Echo Replica	269

8.2.2	Model of Residual Echo Based on Spectral Correlation	273
8.3	Echo Canceller Based on the New Residual Echo Model	274
8.3.1	Overall Structure	274
8.3.2	Estimation of Near-End Speech	275
8.3.3	Spectral Gain Control	276
8.4	Evaluations	277
8.4.1	Objective Evaluations	277
8.4.2	Subjective Evaluation	279
8.5	DSP Implementation and Real-Time Evaluation	280
8.6	Conclusions	280
	References	281

Part III Signal and System Quality Evaluation

9 Telephone-Speech Quality

U. Heute

9.1	Telephone-Speech Signals	287
9.1.1	Telephone Scenario	287
9.1.2	Telephone-Scenario Model	287
9.2	Speech-Signal Quality	289
9.2.1	Intelligibility	289
9.2.2	Speech-Sound Quality	290
9.3	Speech-Quality Assessment	292
9.3.1	Auditory Quality Assessment	292
9.3.2	Aims	292
9.3.3	Instrumental Quality Assessment	293
9.4	Compound-System Quality Prediction	293
9.4.1	The System-Planning Task	293
9.4.2	ETSI Network-Planning Model (E-Model)	293
9.5	Auditory Total-Quality Assessment	294
9.5.1	Conversation Tests	294
9.5.2	Listening Tests	296
9.5.3	LOTs with Pair Comparisons	296
9.5.4	Absolute-Category Rating (ACR) LOTs	297
9.6	Auditory Quality-Attribute Analysis	298
9.6.1	Quality Attributes	298
9.6.2	Attribute-Oriented LOTs	298
9.6.3	Search for Suitable Attributes	302
9.6.4	Integral-Quality Estimation from Attributes	305
9.7	Instrumental Total-Quality Measurement	306
9.7.1	Signal Comparisons	306
9.7.2	Evaluation Approaches	306
9.7.3	Psychoacoustically Motivated Measures	312

9.8 Instrumental Attribute-Based Quality Measurements 320

9.8.1 Basic Ideas 320

9.8.2 Loudness 322

9.8.3 Sharpness 323

9.8.4 Roughness 323

9.8.5 Directness/Frequency Content (DFC) 324

9.8.6 Continuity 326

9.8.7 Noisiness 329

9.8.8 Combined Direct and Attribute-Based Total
Quality Determination 331

9.9 Conclusions, Outlook, and Final Remarks 331

References 332

10 Evaluation of Hands-free Terminals

F. Kettler, H.-W. Gierlich

10.1 Introduction 339

10.2 Quality Assessment of Hands-free Terminals 340

10.3 Subjective Methods for Determining
the Communicational Quality 342

10.3.1 General Setup and Opinion Scales
Used for Subjective Performance Evaluation 343

10.3.2 Conversation Tests 345

10.3.3 Double Talk Tests 346

10.3.4 Talking and Listening Tests 347

10.3.5 Listening-only Tests (LOT) and Third Party
Listening Tests 348

10.3.6 Experts Tests for Assessing Real Life Situations 349

10.4 Test Environment 350

10.4.1 The Acoustical Environment 351

10.4.2 Background Noise Simulation Techniques 351

10.4.3 Positioning of the Hands-Free Terminal 352

10.4.4 Positioning of the Artificial Head 352

10.4.5 Influence of the Transmission System 354

10.5 Test Signals and Analysis Methods 354

10.5.1 Speech and Perceptual Speech Quality Measures 356

10.5.2 Speech-like Test Signals 356

10.5.3 Background Noise 360

10.5.4 Applications 363

10.6 Result Representation 365

10.6.1 Interpretation of HFT “Quality Pies” 366

10.6.2 Examples 368

10.7 Related Aspects 368

10.7.1 The Lombard Effect 368

10.7.2 Intelligibility Outside Vehicles 372

References 375

Part IV Multi-Channel Processing

11 Correlation-Based TDOA-Estimation for Multiple Sources in Reverberant Environments

J. Scheuing, B. Yang

11.1 Introduction 381

11.2 Analysis of TDOA Ambiguities 383

 11.2.1 Signal Model 383

 11.2.2 Multipath Ambiguity 384

 11.2.3 Multiple Source Ambiguity 384

 11.2.4 Ambiguity due to Periodic Signals 386

 11.2.5 Principles of TDOA Disambiguation 386

11.3 Estimation of Direct Path TDOAs 390

 11.3.1 Correlation and Extremum Positions 390

 11.3.2 Raster Matching 392

11.4 Consistent TDOA Graphs 397

 11.4.1 TDOA Graph 397

 11.4.2 Strategies of Consistency Check 398

 11.4.3 Properties of TDOA Graphs 399

 11.4.4 Efficient Synthesis Algorithm 402

 11.4.5 Initialization and Termination 404

 11.4.6 Estimating the Number of Active Sources 405

11.5 Experimental Results 406

 11.5.1 Localization System 406

 11.5.2 TDOA Estimation of a Single Signal Block 408

 11.5.3 Source Position Estimation 412

 11.5.4 Evaluation of Continuous Measurements 412

11.6 Summary 414

References 415

12 Microphone Calibration for Multi-Channel Signal Processing

M. Buck, T. Haulick, H.-J. Pfeiderer

12.1 Introduction 417

12.2 Beamforming with Ideal Microphones 418

 12.2.1 Principle of Beamforming 418

 12.2.2 Evaluation of Beamformers 421

 12.2.3 Statistically Optimum Beamformers 424

12.3 Microphone Mismatch and its Effect on Beamforming 427

 12.3.1 Model for Non-Ideal Microphone Characteristics 428

 12.3.2 Effect of Microphone Mismatch on Fixed Beamformers 429

12.3.3	Effect of Microphone Mismatch on Adaptive Beamformers	430
12.3.4	Comparison of Fixed and Adaptive Beamformers	432
12.4	Calibration Techniques and their Limits for Real-World Applications	432
12.4.1	Calibration of Single Microphones	432
12.4.2	Analysis of Fixed Beamformers	440
12.4.3	Analysis of Adaptive Beamformers	444
12.4.4	Comparison of Fixed and Adaptive Beamformers	448
12.5	Self-Calibration Techniques	449
12.5.1	Basic Unit	451
12.5.2	Configurations for Array Processing	452
12.5.3	Recursive Configuration	455
12.5.4	Adaptation Control	457
12.5.5	Experimental Results	458
12.6	Summary	459
12.A	Experimental Determination of the Directivity Index	460
12.A.1	Numerical Integration over a Spherical Surface	460
12.A.2	Definition of the Coordinate System	462
12.A.3	Determination of the Directivity using the Normalized Cross Power Spectral Densities of Microphone Signals	464
References	465

13 Convolutive Blind Source Separation for Noisy Mixtures

R. Aichner, H. Buchner, W. Kellermann

13.1	Introduction	469
13.2	Blind Source Separation for Acoustic Mixtures Based on the TRINICON Framework	473
13.2.1	Matrix Formulation	473
13.2.2	Optimization Criterion and Coefficient Update	474
13.2.3	Approximations Leading to Special Cases	478
13.2.4	Estimation of the Correlation Matrices and an Efficient Normalization Strategy	484
13.2.5	On Broadband and Narrowband BSS Algorithms in the DFT Domain	485
13.2.6	Experimental Results for Reverberant Environments	488
13.3	Extensions for Blind Source Separation in Noisy Environments	490
13.3.1	Model for Background Noise in Realistic Environments	491
13.3.2	Pre-Processing for Noise-Robust Adaptation	493
13.3.3	Post-Processing for Suppression of Residual Crosstalk and Background Noise	498
13.4	Conclusions	518
References	519

14 Binaural Speech Segregation

N. Roman, D.L. Wang

14.1 Introduction 525
 14.2 T-F Masks for CASA 528
 14.3 Anechoic Binaural Segregation 529
 14.4 Reverberant Binaural Segregation 533
 14.5 Evaluation 536
 14.6 Concluding Remarks 543
 14.7 Acknowledgments 546
 References 546

15 Spatio-Temporal Adaptive Inverse Filtering in the Wave Domain

S. Spors, H. Buchner, R. Rabenstein

15.1 Introduction 551
 15.2 Problem Description 553
 15.2.1 Nomenclature 553
 15.2.2 Massive Multichannel Sound Reproduction 553
 15.2.3 Multichannel Active Listening Room Compensation 556
 15.2.4 Multichannel Active Noise Control 559
 15.2.5 Unified Representation of Spatio-Temporal Adaptive Filtering Problems 561
 15.2.6 Frequency-Domain Notation 563
 15.3 Computation of Compensation Filters 563
 15.3.1 Classification of Algorithms 564
 15.3.2 Ideal Solution 564
 15.3.3 Adaptive Solution 565
 15.3.4 Problems of the Adaptive Solution 567
 15.4 Eigenspace Adaptive Filtering 568
 15.4.1 Generalized Singular Value Decomposition 568
 15.4.2 Eigenspace Adaptive Filtering 569
 15.4.3 Problems 570
 15.5 Wave-Domain Adaptive Filtering 570
 15.5.1 Concept 571
 15.5.2 The Circular Harmonics Decomposition 573
 15.5.3 The Circular Harmonics Expansion Using Boundary Measurements 574
 15.6 Application of WDAF to Adaptive Inverse Filtering Problems 576
 15.6.1 Application of WDAF to Active Listening Room Compensation 577
 15.6.2 Application of WDAF to Active Noise Control 578
 15.7 Conclusions 578
 References 580

Part V Selected Applications

16 Virtual Hearing

K. Wiklund, S. Haykin

16.1 Previous Work.....	588
16.2 VirtualHearing	592
16.3 Room Acoustic Model	593
16.4 HRTF Simulation	597
16.5 Neural Model.....	600
16.6 The Software and Interface.....	605
16.7 Software Testing	609
16.8 Future Work and Conclusions	610
References	613

**17 Dynamic Sound Control Algorithms
in Automobiles**

M. Christoph

17.1 Introduction.....	615
17.1.1 Introduction of Dynamic Volume Control Systems	616
17.1.2 Introduction of Dynamic Equalization Control Systems	618
17.2 Previous Systems – Description and Analysis	619
17.2.1 Speed Dependant Sound systems	619
17.2.2 Microphone Based Dynamic Volume Control Sound Systems	621
17.2.3 Non-Acoustic Sensor Based Sound Systems	643
17.3 Spectrum-Based Dynamic Equalization Control	645
17.3.1 Frequency Domain Adaptive Filter.....	646
17.3.2 Generalized Multidelay Adaptive Filter	649
17.3.3 Step-Size Control	652
17.3.4 Multi-Channel Systems.....	653
17.3.5 Estimating the Power Spectral Density of the Background Noise.....	654
17.3.6 Psychoacoustic Basics	658
17.3.7 The Psychoacoustic Masking Model According to Johnston	661
17.4 Conclusion and Outlook	670
17.5 Acknowledgement	673
References	674

**18 Towards Robust Distant-Talking Automatic Speech
Recognition in Reverberant Environments**

A. Sehr, W. Kellermann

18.1 Introduction.....	679
18.2 The Distant-Talking ASR Scenario	680

18.3	How to Deal with Reverberation in ASR Systems?	683
18.4	Effect of Reverberation in the Feature Domain	691
18.5	Signal Dereverberation and Beamforming	695
18.6	Robust Features	699
18.7	Model Training and Adaptation	700
18.8	Reverberation Modeling for Speech Recognition	702
18.8.1	Feature Production Model	703
18.8.2	Reverberation Model	704
18.8.3	Training of the Reverberation Model	705
18.8.4	Decoding	708
18.8.5	Inner Optimization	713
18.8.6	Solution of the Inner Optimization Problem in the Melspec Domain for Single Gaussian Densities	714
18.8.7	Simulations	717
18.9	Summary and Conclusions	722
	References	723
	Index	729