This Page Intentionally Left Blank
Acoustic Echo
and Noise Control
Acoustic Echo and Noise Control
A Practical Approach

Eberhard Hänsler
Gerhard Schmidt
Contents

List of Figures xi
List of Tables xix
Preface xxix
Acknowledgments xxiii
Abbreviations and Acronyms xxv

Part I Basics

1 Introduction 1
   1.1 Some History 1
   1.2 Overview of the Book 4

2 Acoustic Echo and Noise Control Systems 7
   2.1 Notation 7
   2.2 Applications 7
   2.2.1 Hands-Free Telephone Systems 9
   2.2.2 Car Interior Communication Systems 10
   2.2.3 Public Address Systems 11
CONTENTS

2.2.4 Hearing Aids 12

3 Fundamentals 15
  3.1 Signals 15
    3.1.1 Speech 15
    3.1.2 Noise 22
    3.1.3 Probability Density of Spectral Amplitudes of Car Noise 28
  3.2 Acoustic Echoes 31
    3.2.1 Origin of Acoustic Echoes 31
    3.2.2 Electronic Replica of LEM Systems 35
  3.3 Standards 37
    3.3.1 Standards by ITU and ETSI 37

Part II Algorithms

4 Error Criteria and Cost Functions 43
  4.1 Error Criteria for Adaptive Filters 43
  4.2 Error Criteria for Filter Design 46
  4.3 Error Criteria for Speech Processing and Control Purposes 48

5 Wiener Filter 53
  5.1 Time-Domain Solution 53
  5.2 Frequency-Domain Solution 58

6 Linear Prediction 61
  6.1 Normal Equations 62
  6.2 Levinson–Durbin Recursion 64

7 Algorithms for Adaptive Filters 73
  7.1 The Normalized Least Mean Square Algorithm 77
    7.1.1 Control Structures 79
    7.1.2 Stability Analysis 80
  7.2 The Affine Projection Algorithm 95
    7.2.1 Derivation of the Algorithm 96
    7.2.2 Fast Versions of the AP Algorithm 99
  7.3 The Recursive Least Squares Algorithm 105
    7.3.1 Derivation of the Algorithm 105
CONTENTS

7.3.2 Fast Versions of the RLS Algorithm 110
7.4 The Kalman Algorithm 111
7.4.1 Initialization 114
7.4.2 Prediction 116
7.4.3 Correction 117
7.4.4 Colored Noise 120

Part III Acoustic Echo and Noise Control

8 Traditional Methods for Stabilization of Electroacoustic Loops 129
  8.1 Adaptive Line Enhancement 130
    8.1.1 FIR Structure 132
    8.1.2 IIR Structure 133
    8.1.3 Comparison of Both Structures 137
  8.2 Frequency Shift 137
    8.2.1 Basic Idea 138
    8.2.2 Hilbert Filter 141
    8.2.3 Results 144
  8.3 Controlled Attenuation 146
    8.3.1 Automatic Gain Control 148
    8.3.2 Loss Control 151
    8.3.3 Limiter 158

9 Echo Cancellation 163
  9.1 Processing Structures 164
    9.1.1 Fullband Cancellation 165
    9.1.2 Block Processing Algorithms 175
    9.1.3 Subband Cancellation 185
  9.2 Stereophonic and Multichannel Echo Cancellation 212
    9.2.1 Stereophonic Acoustic Echo Cancellation 213
    9.2.2 Multichannel Systems 218

10 Residual Echo and Noise Suppression 221
  10.1 Basics 222
    10.1.1 Echo and Noise Suppression in the Frequency Domain 223
    10.1.2 Filter Characteristics 235
  10.2 Suppression of Residual Echoes 244
    10.2.1 Comfort Noise 248
CONTENTS

10.3 Suppression of Background Noise  251
   10.3.1 Postprocessing  255
10.4 Combining Background Noise and Residual Echo Suppression  262

11  Beamforming  267
   11.1 Basics  269
      11.1.1 Spatial Sampling of a Sound Field  270
   11.2 Characteristics of Microphone Arrays  272
      11.2.1 Directional Pattern  276
      11.2.2 Array Gain  278
      11.2.3 Further Performance Criteria  278
   11.3 Fixed Beamforming  279
      11.3.1 Delay-and-Sum Beamforming  279
      11.3.2 Filter-and-Sum Beamforming  287
   11.4 Adaptive Beamforming  291
      11.4.1 Generalized Sidelobe Canceler  293

Part IV  Control and Implementation Issues

12  System Control—Basic Aspects  299
   12.1 Convergence versus Divergence Speed  299
   12.2 System Levels for Control Design  301

13  Control of Echo Cancellation Systems  303
   13.1 Pseudooptimal Control Parameters for the NLMS Algorithm  304
      13.1.1 Pseudooptimal Stepsize  304
      13.1.2 Pseudooptimal Regularization  308
      13.1.3 “Relationship” between Both Control Methods  310
   13.2 Pseudooptimal Control Parameters for the Affine Projection Algorithm  312
      13.2.1 Pseudooptimal Regularization  314
      13.2.2 Pseudooptimal Stepsize  316
   13.3 Summary of Pseudooptimal Control Parameters  317
   13.4 Detection and Estimation Methods  318
      13.4.1 Short-Term Power Estimation  319
      13.4.2 Estimating the System Distance  321
      13.4.3 Detection of Remote Single Talk  325
      13.4.4 Rescue Detectors  333
13.4.5 Concluding Remarks 338
13.5 Detector Overview and Combined Control Methods 338
  13.5.1 Stepsize Control for the NLMS Algorithm 338
  13.5.2 Combined Stepsize and Regularization Control for the
        NLMS Algorithm 341
  13.5.3 Regularization Control for AP Algorithms 344

14 Control of Noise and Echo Suppression Systems 349
  14.1 Estimation of Spectral Power Density of Background Noise 349
    14.1.1 Schemes with Speech Pause Detection 350
    14.1.2 Minimum Statistics 355
    14.1.3 Scheme with Fast and Slow Estimators 358
    14.1.4 Extensions to Simple Estimation Schemes 359
    14.1.5 Concluding Remarks 363
  14.2 Musical Noise 364
  14.3 Control of Filter Characteristics 365
    14.3.1 Overestimation Factor 366
    14.3.2 Spectral Floor 367

15 Control for Beamforming 371
  15.1 Practical Problems 373
    15.1.1 Far-Field Assumptions 373
    15.1.2 Sensor Imperfections 374
  15.2 Stepsize Control 377

16 Implementation Issues 381
  16.1 Quantization Errors 381
  16.2 Number Representation Errors 382
  16.3 Arithmetical Errors 382
  16.4 Fixed Point versus Floating Point 382
  16.5 Quantization of Filter Taps 383

Part V Outlook and Appendixes

17 Outlook 391

Appendix A Subband Impulse Responses 393
  A.1 Consequences for Subband Echo Cancellation 393
A.2 Transformation 395
A.3 Concluding Remarks 397

Appendix B **Filterbank Design** 401
- B.1 Conditions for Approximately Perfect Reconstruction 402
- B.2 Filter Design Using a Product Approach 407
- B.3 Design of Prototype Lowpass Filters 414
- B.4 Analysis of Prototype Filters and the Filterbank System 416

References 423

Index 441
# List of Figures

## 1 Introduction
- 1.1 Bell’s new telephone  
- 1.2 The hands-busy telephone  
- 1.3 Wallmounted telephone from 1881  
- 1.4 Desk telephone from 1897

## 2 Acoustic Echo and Noise Control Systems
- 2.1 Structure of a monophonic hands-free telephone system  
- 2.2 Frequency responses of different communication directions within a car  
- 2.3 Structure of a car interior communication system  
- 2.4 Behind-the-ear hearing aid  
- 2.5 Architecture of a hearing aid

## 3 Fundamentals
- 3.1 Example of a speech sequence  
- 3.2 Voiced and unvoiced speech segments  
- 3.3 Normalized Gaussian probability density function and normalized approximation for speech signals  
- 3.4 Approximations of the probability density function

\( x_i \)
<table>
<thead>
<tr>
<th>List of Figures</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>3.5 Probability density function of the logarithmic normalized power spectral density of speech and histograms</td>
<td>22</td>
</tr>
<tr>
<td>3.6 Estimates of the power spectral densities of noises measured in a car and in offices</td>
<td>23</td>
</tr>
<tr>
<td>3.7 Noise produced by a PC ventilator</td>
<td>25</td>
</tr>
<tr>
<td>3.8 Examples of engine noise</td>
<td>26</td>
</tr>
<tr>
<td>3.9 Time–frequency analyses and power spectral densities of wind noise</td>
<td>27</td>
</tr>
<tr>
<td>3.10 Noise during a change in road surface</td>
<td>29</td>
</tr>
<tr>
<td>3.11 Noise caused by a ventilator</td>
<td>30</td>
</tr>
<tr>
<td>3.12 Probability density functions of the logarithmic normalized power spectral density of car noise and histograms</td>
<td>32</td>
</tr>
<tr>
<td>3.13 Model of the loudspeaker–enclosure–microphone system</td>
<td>33</td>
</tr>
<tr>
<td>3.14 Floorplan of an office</td>
<td>34</td>
</tr>
<tr>
<td>3.15 Impulse response and absolute value of the frequency response of an office</td>
<td>36</td>
</tr>
<tr>
<td>3.16 Impulse responses measured in different environments</td>
<td>38</td>
</tr>
<tr>
<td>3.17 Relationship between echo delay and required attenuation</td>
<td>39</td>
</tr>
<tr>
<td>3.18 Composite source signal</td>
<td>40</td>
</tr>
</tbody>
</table>

4 Error Criteria and Cost Functions
4.1 Half-band filter designed with the Remez exchange algorithm | 48 |
4.2 Cepstral transformation as a preprocessor for cost functions | 50 |
4.3 Structure of a single-channel hands-free telephone system | 51 |
4.4 Double-talk detection in hands-free telephone systems | 52 |

5 Wiener Filter
5.1 Wiener filter | 54 |
5.2 Example of a Wiener filter | 57 |
5.3 Error surfaces | 59 |

6 Linear Prediction
6.1 Prediction error filter | 62 |
6.2 Power spectral densities of a speech signal and the related decorrelated signal | 63 |
6.3 Original speech signal and prediction error signals | 65 |
6.4 Predictor of order $k$ | 69 |
6.5 Inverse spectral power density of the predictor input and absolute values of the predictor transfer function | 72 |
7 Algorithms for Adaptive Filters
7.1 Adaptive filter for system identification 73
7.2 Convergence with colored noise 75
7.3 Adaptive filter for system equalization 76
7.4 Example of a vector stepsize control 81
7.5 Prewhitening structure 84
7.6 Contraction–expansion parameter 85
7.7 Simulation examples and theoretic convergence 87
7.8 Convergence plane in the absence of local noise 89
7.9 Convergences with different filter orders 90
7.10 Time-variant prewhitening structure 92
7.11 Convergence examples—only stepsize control 93
7.12 Convergence examples—only regularization control 95
7.13 Control strategies 96
7.14 NLMS versus AP 99
7.15 Convergences for different algorithms 100
7.16 Kalman filter: state space description of coefficient update 112
7.17 Statespace model of the echo canceling filter and the Kalman filter 122
7.18 Sequence of steps of the Kalman filter algorithm 123

8 Traditional Methods for Stabilization of Electroacoustic Loops
8.1 Basic scheme of adaptive noise cancellation 131
8.2 Structure of an FIR filter for adaptive line enhancement 132
8.3 Simulation example for FIR adaptive line enhancement 134
8.4 Structure of an IIR filter for adaptive line enhancement 135
8.5 Simulation example for IIR adaptive line enhancement 136
8.6 Time–frequency analysis of the estimated distortion signals 138
8.7 Structure of a public address system 139
8.8 Frequency response measured in a small lecture room 140
8.9 Spectra of original and shifted signals 142
8.10 Structure of the frequency shift 143
8.11 Frequency responses of Hilbert filters 145
8.12 Additional gains due to the frequency shift 146
8.13 Dynamic processing in hands-free telephone systems 147
8.14 Basic structure of an automatic gain control circuit 148
8.15 Simulation example for automatic gain control 152
8.16 Structure of a loss control 153
8.17 Simulation example for a loss control circuit (part I) 155
8.18 Simulation example for a loss control circuit (part II) 156
8.19 Hard–soft limiter combination 159
8.20 Simulation example for dynamic limiting 161

9 Echo Cancellation

9.1 Structure of a single-channel acoustic echo control system 163
9.2 Placement of decorrelation and inverse decorrelation filters 167
9.3 Power spectral densities and decorrelation filters for different languages 168
9.4 Placement of adaptive decorrelation filters 169
9.5 Convergence improvement due to decorrelation filters 170
9.6 First-order decorrelation, excited with speech sequences from both male and female speakers 171
9.7 System distance for different static update schemes 173
9.8 System distance for different adaptive update schemes 174
9.9 Time-domain block processing 177
9.10 Convergences using fullband NLMS and normalized block LMS 179
9.11 Computational complexities and memory requirements 183
9.12 Normalized fast block LMS algorithm 186
9.13 Structure of an acoustic echo control system 188
9.14 Computational complexity and memory requirements 190
9.15 Downsampling by a factor $r$ 191
9.16 Upsampling by a factor $r$ 192
9.17 Example of a four-band quadrature mirror filterbank 193
9.18 Four-band nonuniform quadrature mirror filterbank 194
9.19 Two-band analysis synthesis quadrature mirror framework 195
9.20 Design example for a 32-tap quadrature mirror filterbank 196
9.21 Basic structure of a polyphase analysis filterbank 198
9.22 Prototype lowpass filter and resulting bandpass filter 199
9.23 Efficient structure of a polyphase analysis filterbank 201
9.24 Basic structure of a polyphase synthesis filterbank 202
9.25 Efficient structure of a polyphase synthesis filterbank 203
9.26 Example of a prototype lowpass filter 205
9.27 Convergences with and without artificially delayed microphone signal 207
9.28 Group delays of the two analysis filterbanks 208
9.29 Convergences with equal and with different analysis filterbanks 209
9.30 Orders of the subband echo cancellation filters 212
10 **Residual Echo and Noise Suppression**

<table>
<thead>
<tr>
<th>Figure</th>
<th>Description</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>10.1</td>
<td>Single-channel system for residual echo and noise suppression</td>
<td>221</td>
</tr>
<tr>
<td>10.2</td>
<td>Basic units of a noise suppression system</td>
<td>222</td>
</tr>
<tr>
<td>10.3</td>
<td>Short-term spectrum of a voiced speech sequence</td>
<td>223</td>
</tr>
<tr>
<td>10.4</td>
<td>Uniform analysis–synthesis system</td>
<td>224</td>
</tr>
<tr>
<td>10.5</td>
<td>Weighted Hanning windows</td>
<td>226</td>
</tr>
<tr>
<td>10.6</td>
<td>Window functions</td>
<td>227</td>
</tr>
<tr>
<td>10.7</td>
<td>Analysis by a DFT and a by polyphase filterbank</td>
<td>228</td>
</tr>
<tr>
<td>10.8</td>
<td>Impulse and frequency responses of a Hanning window and of a prototype lowpass filter</td>
<td>229</td>
</tr>
<tr>
<td>10.9</td>
<td>Critical bandwidth as a function of frequency</td>
<td>231</td>
</tr>
<tr>
<td>10.10</td>
<td>Structure of an allpass-transformed polyphase filterbank</td>
<td>232</td>
</tr>
<tr>
<td>10.11</td>
<td>Transfer functions of allpass-transformed polyphase filterbanks</td>
<td>233</td>
</tr>
<tr>
<td>10.12</td>
<td>Two-stage cascaded filterbank</td>
<td>234</td>
</tr>
<tr>
<td>10.13</td>
<td>Structure of a cascaded filterbank system utilized for echo cancellation and residual echo and noise suppression</td>
<td>236</td>
</tr>
<tr>
<td>10.14</td>
<td>Filter characteristics</td>
<td>238</td>
</tr>
<tr>
<td>10.15</td>
<td>Further filter characteristics</td>
<td>242</td>
</tr>
<tr>
<td>10.16</td>
<td>Suppression of residual echoes</td>
<td>247</td>
</tr>
<tr>
<td>10.17</td>
<td>Structure of a residual echo suppression system with comfort noise injection</td>
<td>248</td>
</tr>
<tr>
<td>10.18</td>
<td>Structure of a binary pseudo–noise generator</td>
<td>249</td>
</tr>
<tr>
<td>10.19</td>
<td>Simulation examples for comfort noise injection</td>
<td>250</td>
</tr>
<tr>
<td>10.20</td>
<td>Example of a background noise suppression</td>
<td>253</td>
</tr>
<tr>
<td>10.21</td>
<td>Results of a survey on the importance of several aspects for evaluation of a noise suppression scheme</td>
<td>254</td>
</tr>
<tr>
<td>10.22</td>
<td>Mismatch between maxima of the postfilter and the pitch structure of speech</td>
<td>258</td>
</tr>
<tr>
<td>10.23</td>
<td>Frequency response of the postfilter designed in the frequency domain</td>
<td>259</td>
</tr>
<tr>
<td>10.24</td>
<td>Time–frequency analysis of the spoken digits “one” to “ten”</td>
<td>261</td>
</tr>
</tbody>
</table>
LIST OF FIGURES

10.25 Time-frequency analysis of a signal with and without fricative spreading 263
10.26 Simulation example for a combined residual echo and noise suppression system 265

11 Beamforming
11.1 Influence of beamforming on the recognition rate of speech recognition systems 268
11.2 Beampatterns of two beamformers 271
11.3 Characteristics of a microphone array 273
11.4 Coordinate system 274
11.5 Beampatterns of arrays with different sensor types 277
11.6 Structure of the delay-and-sum beamformer 280
11.7 Geometry of a linear beamformer 280
11.8 Structure of a delay-and-sum beamformer implemented in the frequency domain 282
11.9 Group delays and frequency responses of three delay compensation filters 285
11.10 Comparison of a single-microphone signal and an output signal of a delay-and-sum beamformer 286
11.11 Structure of the filter-and-sum beamformer 287
11.12 Beampatterns of a delay-and-sum beamformer and of a delay-and-filter beamformer 290
11.13 Structure of the generalized sidelobe canceler 293

12 System Control—Basic Aspects
12.1 Convergence and divergence of an adaptive filter 300

13 Control of Echo Cancellation Systems
13.1 Convergence using a time-variant stepsize 307
13.2 Convergence using a time-variant regularization parameter 311
13.3 Regularization versus stepsize control 313
13.4 Convergence using a time-variant regularization parameter 317
13.5 Short-term power estimation 319
13.6 Simple scheme for estimating the local noise power 321
13.7 Simulation example of the background noise estimation 322
13.8 Model of the LEM system and the adaptive filter 323
13.9 Methods for estimating the system distance 324
13.10 Simulation examples for estimation of the system distance 326
13.11 Local speech activity detection 328
13.12 Simulation example for the detection of local speech activity 329
13.13 Remote single-talk detection methods 330
13.14 Simulation example for detecting remote single talk 332
13.15 Detection principles for enclosure dislocations 334
13.16 Simulation examples for the detection of enclosure dislocations 336
13.17 Overview of detector combination possibilities – Part I 339
13.18 Simulation example of a complete stepsize control 342
13.19 Overview of detector combination possibilities – Part II 343
13.20 Simulation example for a combined stepsize and regularization control 345
13.21 Simulation example for a regularization control 347

14 Control of Noise and Echo Suppression Systems
14.1 Noisy speech signal and three different short-term power estimations 352
14.2 Example of background noise-level estimations utilizing a voice activity detector 354
14.3 Basic structure of a minimum statistic 356
14.4 Background noise estimation with minimum statistics 358
14.5 Short-term smoothed subband powers and corresponding frequency-smoothed counterparts 361
14.6 Effects of temporal filtering and smoothing along the frequency direction 362
14.7 Comparison of different noise estimation schemes 363
14.8 Musical noise 365
14.9 Standard Wiener characteristic and its recursive counterpart 367
14.10 Mapping function of the preliminary subband attenuation factors on the subband spectral floor parameters 369
14.11 Simulation example showing the time/frequency-selective adjustment of the subband spectral floor parameters 370

15 Control for Beamforming
15.1 Input and output signals of beamformers with both fixed and adaptive control parameters 372
15.2 Measurement of the diffuse field distance in a car 374
15.3 Tolerances of microphones 375
15.4 Structure of the generalized sidelobe structure with additional microphone calibration stage 376
15.5 Stepsize control for the generalized sidelobe structure within the subband domain 377
15.6 Input signal of the first microphone and normalized power ratio 379

16 Implementation Issues
16.1 Convergence of the system distance as a function of the arithmetic used 383
16.2 Convergence of the system distance for the NLMS algorithm with scalar stepsize and for exponentially weighted stepsize 384
16.3 System mismatch vector at the beginning of the adaptation and after 500, 2000, and 4000 iterations 385
16.4 Impulse response of an LEM system and its corresponding scaling matrix 386

A Subband Impulse Responses
A.1 Time–frequency analysis of a room impulse response 394
A.2 Fullband and first subband impulse response 398
A.3 Impulse response of the second subband 399

B Filterbank Design
B.1 Structure of the filter design 403
B.2 Structure of a through-connected filterbank system 405
B.3 Frequency response of a Dolph–Chebyshev lowpass filter 409
B.4 Graph of the first six Chebyshev polynomials 411
B.5 Quadruples of zeros 413
B.6 Properties of the designed prototype lowpass filter 417
B.7 Properties of the through-connected filterbank system 419
B.8 Aliasing components of the through-connected filterbank 421
List of Tables

2 Acoustic Echo and Noise Control Systems
  2.1 Notation .................................................. 8

6 Linear Prediction
  6.1 Levinson–Durbin recursion ................................. 71

7 Algorithms for Adaptive Filters
  7.1 NLMS algorithm ............................................... 79
  7.2 AP algorithm .................................................. 98
  7.3 Fast affine projection algorithm ......................... 104
  7.4 RLS algorithm ................................................ 109
  7.5 Adaptive algorithms and their cost functions .......... 109
  7.6 Kalman algorithm ............................................ 121

9 Echo Cancellation
  9.1 Processing Structures ....................................... 165
  9.2 Normalized fast block LMS algorithm .................... 187
  9.3 Orders of the subband echo cancellation filters ........ 211

xix
### LIST OF TABLES

10 **Residual Echo and Noise Suppression**

10.1 Frequency resolutions of the inner filterbanks  
10.2 Indices of the nonzero feedback coefficients for maximum-length pseudonoise sequences

13 **Control of Echo Cancellation Systems**

13.1 Pseudooptimal control parameters
The motivation to write this book originated after a 20-year long engagement in the problems of acoustic echoes and noise control at the Signal Theory Group at Darmstadt University of Technology. About 20 Ph.D. students were involved in various projects on these topics. The authors now intend to present a concise documentation of the results of this work embedded into the state of the art.

The work of the Signal Theory Group spanned the entire range of scientific and development work: theoretical considerations, computer and hardware simulations, and implementation of realtime demonstrators. Testing ideas in real environments at real time turned out to be an extremely useful tool to judge results gained by formal analysis and computer simulations and to create new ideas.

The organization of this book somewhat reflects this working mode; we start with presenting the basic algorithms for filtering, for linear prediction, and for adaptation of filter coefficients. We then apply these methods to acoustic echo cancellation and residual echo and noise suppression. Considerable space is devoted to the estimation of nonmeasurable quantities that are, however, necessary to control the algorithms. Suitable control structures based on these quantities are derived in some detail.

Worldwide knowledge of problems of echo and noise control has increased enormously. Therefore, it was necessary to limit the contents of the book. The main emphasis is put on single-channel systems where—to the opinion of the authors—a certain completeness has been reached. Multichannel systems provide additional options for improved solutions. They receive, currently, increased attention in research and development laboratories. The book deals with the basic ideas.
Implementation issues of acoustic echo and noise control systems are, beyond doubt, just as important as the topics mentioned above. They are, however, not covered in detail in this text.

The readers of this book should have a basic knowledge of linear system theory and of digital signal processing as it is presented, for example, in undergraduate courses. The authors hope that all—theoreticians and practitioners alike—are able to take advantage of the material of this book and learn more about this exciting area of digital signal processing.

Darmstadt and Ulm, Germany

Eberhard Hänssler
Gerhard Schmidt
Acknowledgments

Hidden behind the authors of a book on such an active area as acoustic echo and noise control are a large number of colleagues who helped with intensive discussions, constructive criticism, and constant encouragement. To mention all by name would inevitably mean to forget some. We express our sincere thanks to all of them.

There is, however, no rule without exception. Special thanks go to all former and current research assistants of the Signal Theory Group of Darmstadt University of Technology. Their work, documented in numerous presentations, papers, and Ph.D. dissertations, considerably contributed to the progress of acoustic echo and noise control. Their names can be found in the references to this book.

Furthermore, we have to offer our thanks to all members of the Temic audio research group.

We also have to thank the German Science Foundation for the support of some of the graduate students within the frame of a Graduate College on Intelligent Systems for Information Technology and Control Engineering.

The International Workshop on Acoustic Echo and Noise Control (IWAENC) biennially gathers researchers and developers from many countries to meet their colleagues and to present new ideas. The authors participated in these events and enjoyed the intensive discussions.

We are especially indebted to Professor Simon Haykin for his constant encouragement to write this book and his help in planning it.

Finally, we would like to express our thanks for the friendly assistance of the editors of John Wiley & Sons.

Eberhard Hänssler
Gerhard Schmidt
**Abbreviations and Acronyms**

<table>
<thead>
<tr>
<th>Abbreviation</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>AD</td>
<td>Analog/digital</td>
</tr>
<tr>
<td>AGC</td>
<td>Automatic gain control</td>
</tr>
<tr>
<td>AP</td>
<td>Affine projection</td>
</tr>
<tr>
<td>AR</td>
<td>Autoregressive</td>
</tr>
<tr>
<td>DA</td>
<td>Digital/analog</td>
</tr>
<tr>
<td>DFT</td>
<td>Discrete Fourier transform</td>
</tr>
<tr>
<td>DSP</td>
<td>Digital signal processor</td>
</tr>
<tr>
<td>ERLE</td>
<td>Echo-return loss enhancement</td>
</tr>
<tr>
<td>ES</td>
<td>Exponentially weighted stepsize</td>
</tr>
<tr>
<td>ETSI</td>
<td>European Telecommunication Standards Institute</td>
</tr>
<tr>
<td>FAP</td>
<td>Fast affine projection</td>
</tr>
<tr>
<td>FFT</td>
<td>Fast Fourier transform</td>
</tr>
<tr>
<td>FIR</td>
<td>Finite impulse response</td>
</tr>
<tr>
<td>FTF</td>
<td>Fast transversal filter</td>
</tr>
<tr>
<td>GSM</td>
<td>Global system for mobile communications</td>
</tr>
<tr>
<td>IDFT</td>
<td>Inverse discrete Fourier transform</td>
</tr>
<tr>
<td>IEEE</td>
<td>Institute of Electrical and Electronics Engineers</td>
</tr>
<tr>
<td>Abbreviation</td>
<td>Definition</td>
</tr>
<tr>
<td>--------------</td>
<td>------------</td>
</tr>
<tr>
<td>IFFT</td>
<td>Inverse fast Fourier transform</td>
</tr>
<tr>
<td>IIR</td>
<td>Infinite impulse response</td>
</tr>
<tr>
<td>INR</td>
<td>Input-to-noise ratio</td>
</tr>
<tr>
<td>IP</td>
<td>Internet Protocol</td>
</tr>
<tr>
<td>ITU</td>
<td>International Telecommunication Union</td>
</tr>
<tr>
<td>LCMV</td>
<td>Linearly constraint minimum variance</td>
</tr>
<tr>
<td>LEM</td>
<td>Loudspeaker-enclosure-microphone (system)</td>
</tr>
<tr>
<td>LMS</td>
<td>Least mean square</td>
</tr>
<tr>
<td>MAC</td>
<td>Multiply and accumulate</td>
</tr>
<tr>
<td>MELP</td>
<td>Mixed excitation linear predictor</td>
</tr>
<tr>
<td>MFLOPS</td>
<td>Million floating-point operations per second</td>
</tr>
<tr>
<td>MIPS</td>
<td>Million instructions per second</td>
</tr>
<tr>
<td>MSE</td>
<td>Mean square error</td>
</tr>
<tr>
<td>NLMS</td>
<td>Normalized least mean square</td>
</tr>
<tr>
<td>PARCOR</td>
<td>Partial correlation (coefficient)</td>
</tr>
<tr>
<td>QMF</td>
<td>Quadrature mirror filterbank</td>
</tr>
<tr>
<td>RLS</td>
<td>Recursive least squares</td>
</tr>
<tr>
<td>SFTF</td>
<td>Short-time Fourier transform</td>
</tr>
<tr>
<td>SNR</td>
<td>Signal-to-noise ratio</td>
</tr>
<tr>
<td>VAD</td>
<td>Voice activity detection</td>
</tr>
<tr>
<td>XLMS</td>
<td>Extended least mean square</td>
</tr>
</tbody>
</table>
Part I

Basics
This Page Intentionally Left Blank