

Acoustic Echo and Noise Control

A Practical Approach

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Eberhard Hänsler
Gerhard Schmidt



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Preface

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

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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.

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*Eberhard Hänsler
Gerhard Schmidt*

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Abbreviations and Acronyms

AD	Analog/digital
AGC	Automatic gain control
AP	Affine projection
AR	Autoregressive
DA	Digital/analog
DFT	Discrete Fourier transform
DSP	Digital signal processor
ERLE	Echo-return loss enhancement
ES	Exponentially weighted stepsize
ETSI	European Telecommunication Standards Institute
FAP	Fast affine projection
FFT	Fast Fourier transform
FIR	Finite impulse response
FTF	Fast transversal filter
GSM	Global system for mobile communications
IDFT	Inverse discrete Fourier transform
IEEE	Institute of Electrical and Electronics Engineers

IFFT	Inverse fast Fourier transform
IIR	Infinite impulse response
INR	Input-to-noise ratio
IP	Internet Protocol
ITU	International Telecommunication Union
LCMV	Linearly constraint minimum variance
LEM	Loudspeaker–enclosure–microphone (system)
LMS	Least mean square
MAC	Multiply and accumulate
MELP	Mixed excitation linear predictor
MFLOPS	Million floating-point operations per second
MIPS	Million instructions per second
MSE	Mean square error
NLMS	Normalized least mean square
PARCOR	Partial correlation (coefficient)
QMF	Quadrature mirror filterbank
RLS	Recursive least squares
SFTF	Short-time Fourier transform
SNR	Signal-to-noise ratio
VAD	Voice activity detection
XLMS	Extended least mean square

Part I

Basics

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