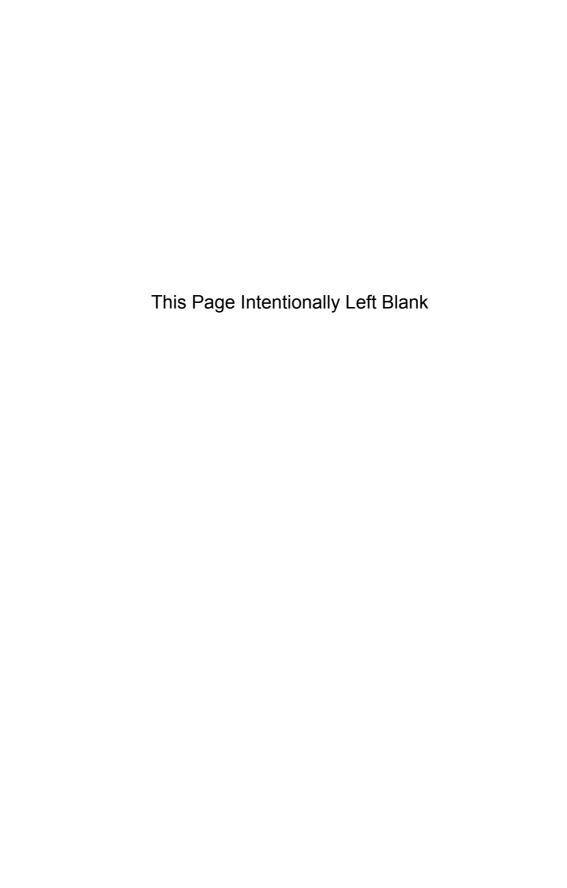
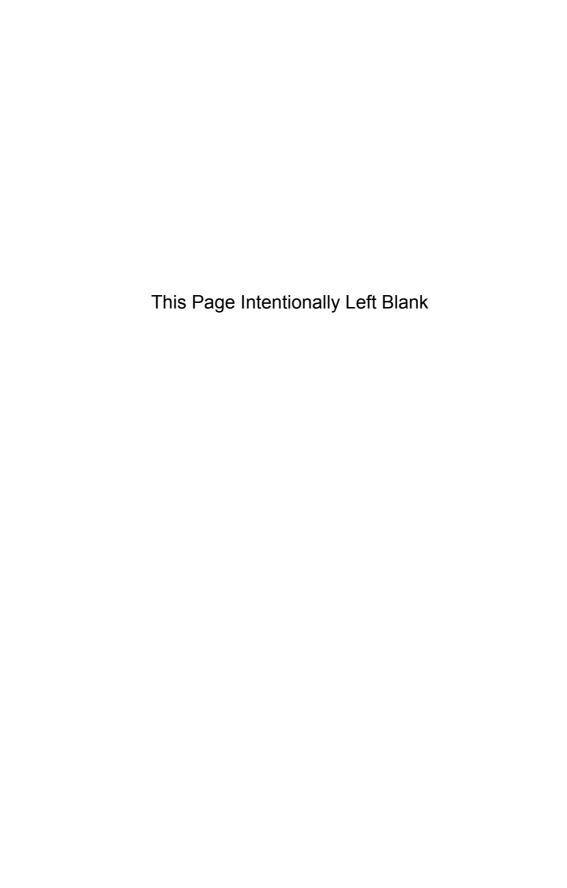
### **Acoustic Echo and Noise Control**

A Practical Approach



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### A Practical Approach

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.

#### xxii PREFACE

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 Hänsler Gerhard Schmidt

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

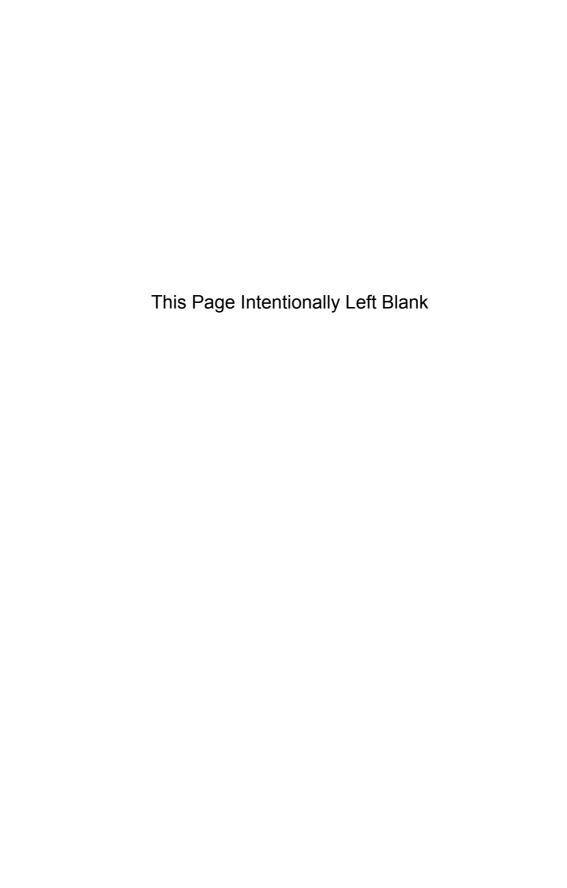
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Eberhard Hänsler Gerhard Schmidt



## 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

#### **XXVI** ABBREVIATIONS AND ACRONYMS

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**

