

Declaration of Catherine Vassilkova

1. I, Catherine Vassilkova, am an employee of BEADOX LLC ("BEA").
2. I am over the age of 21 and have personal knowledge of the facts contained herein unless otherwise indicated.
3. I have worked in the field of information research and document retrieval for law firms since 2005.
4. I have been with BEA since I established the company in 2019.
5. BEA, based in Silver Spring, MD, conducts research at the District of Columbia federal and local courts, archives, government agencies, and libraries around the world. BEA's services include interlibrary loans, specialized research, article location and delivery, and other document retrieval. One of my responsibilities is to scan textbooks for requesters.
6. BEA specializes in the research and delivery of materials from the Library of Congress.
7. On September 26, 2025, I was asked by the requester Morrison and Foerster LLP to obtain a scan from the following book: Springer handbook of speech processing (2008).
8. On September 26, 2025, I obtained Springer handbook of speech processing (2008) from the Library of Congress, and scanned the requested excerpts including book cover, title page, copyright page date-stamped (December 11 2007), table of contents pages, and chapter sections 50.1 (p.1021-1023), 51.2.1 from Springer handbook of speech processing (2008). Those excerpts are included as Exhibit 1009.
9. I confirm that the copyright page of the scanned book has a stamped date of receipt of December 11 2007.
10. I declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true, and that these statements were made with knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under section 1001 of Title 18 of the United States Code.
11. Executed on the 29th of September, 2025, in Silver Spring, MD 20910.

Signed:

C. Vassilkova

Springer Handbook of Speech Processing



From common consumer products such as cell phones and MP3 players to more sophisticated projects such as human-machine interfaces and responsive robots, speech technologies are now everywhere. Many think that it is just a matter of time before more applications of the science of speech become inescapable in our daily life. This **Springer Handbook of Speech Processing** is meant to play a fundamental role for sustainable progress in speech research and development. It targets three categories of readers: graduate students, professors and active researchers in academia and research labs, and engineers in industry who need to understand or implement some specific algorithms for their speech-related products. The handbook could also be used as a sourcebook for one or more graduate courses on signal processing for speech and different aspects of speech processing and applications. A quickly accessible source of application-oriented, authoritative and comprehensive information about these technologies, it combines the established knowledge derived from research in such fast evolving disciplines as Signal Processing and Communications, Acoustics, Computer Science, and Linguistics.

Key Topics

- › Production, Perception, and Modeling of Speech
- › Signal Processing for Speech
- › Speech Coding
- › Text-to-Speech Synthesis
- › Speech and Speaker Recognition
- › Language Recognition
- › Speech Enhancement
- › Multichannel Speech Processing

Features

- › Authoritative desk reference of one of tomorrow's breakthrough technologies
- › Provides quick access to applicable, reliable, and comprehensive knowledge
- › Contains over 450 two-color illustrations
- › Parts and chapters with summaries, detailed index and fully searchable DVD-ROM guarantee quick access to data and links to other sources
- › Delivers a wealth of up-to-date references and further reading

Benesty
Sondhi
Huang
Editors



Springer
Handbook
of
Speech
Processing

ISBN 978-3-540-49125-5



springer.com

DVD containing all contents
best viewed with Adobe Reader 8



System Requirements

- Windows 95/98/ME 32 MB RAM, Windows NT 4/2000/XP 64 MB RAM, Windows Vista 512 MB RAM
- MAC OS 9 or higher, 64 MB RAM, 333 MHz
- LINUX Pentium I /166 MHz, 64 MB RAM

TK 7882
.S65
.S6715
2008
Copy 1

Springer Handbook of

Speech Processing



Benesty
Sondhi
Huang
Editors

Springer

**Springer Handbook
of Speech Processing**

Springer Handbooks provide a concise compilation of approved key information on methods of research, general principles, and functional relationships in physical sciences and engineering. The world's leading experts in the fields of physics and engineering will be assigned by one or several renowned editors to write the chapters comprising each volume. The content is selected by these experts from Springer sources (books, journals, online content) and other systematic and approved recent publications of physical and technical information.

The volumes are designed to be useful as readable desk reference books to give a fast and comprehensive overview and easy retrieval of essential reliable key information, including tables, graphs, and bibliographies. References to extensive sources are provided.

Springer Handbook of Speech Processing

Jacob Benesty, M. Mohan Sondhi, Yiteng Huang
(Eds.)

With DVD-ROM, 456 Figures and 113 Tables

 Springer

Editors:

Jacob Benesty
INRS-EMT, University of Quebec
800 de la Gauchetiere Ouest, Suite 6900
Montreal, Quebec, H5A 1K6, Canada
benesty@emt.inrs.ca

M. Mohan Sondhi
Avayalabs Research
233 Mount Airy Road
Basking Ridge, NJ 07920, USA
mms@research.avayalabs.com

Yiteng Huang
Bell Laboratories, Alcatel-Lucent
600 Mountain Avenue
Murray Hill, NJ 07974, USA
arden_huang@ieee.org

Library of Congress Control Number: 2007931999

ISBN: 978-3-540-49125-5 e-ISBN: 978-3-540-49127-9

This work is subject to copyright. All rights reserved, whether the whole or part of the material is concerned, specifically the rights of translation, reprinting, reuse of illustrations, recitation, broadcasting, reproduction on microfilm or in any other way, and storage in data banks. Duplication of this publication or parts thereof is permitted only under the provisions of the German Copyright Law of September, 9, 1965, in its current version, and permission for use must always be obtained from Springer-Verlag. Violations are liable for prosecution under the German Copyright Law.

Springer is a part of Springer Science+Business Media
springer.com

© Springer-Verlag Berlin Heidelberg 2008

The use of designations, trademarks, etc. in this publication does not imply, even in the absence of a specific statement, that such names are exempt from the relevant protective laws and regulations and therefore free for general use.

Product liability: The publisher cannot guarantee the accuracy of any information about dosage and application contained in this book. In every individual case the user must check such information by consulting the relevant literature.

Typesetting and production:
LE-TeX Jelonek, Schmidt&Vöckler GbR, Leipzig
Senior Manager Springer Handbook: Dr. W. Skolaut, Heidelberg
Typography and layout: schreiberVIS, Seeheim
Illustrations: Hippmann GbR, Schwarzenbruck
Cover design: eStudio Calamar Steinen, Barcelona
Cover production: WMXDesign GmbH, Heidelberg
Printing and binding: Stürtz GmbH, Würzburg

Printed on acid free paper

SPIN 11544036 60/3180/YL 5 4 3 2 1 0

Foreword

Over the past three decades digital signal processing has emerged as a recognized discipline. Much of the impetus for this advance stems from research in representation, coding, transmission, storage and reproduction of speech and image information. In particular, interest in voice communication has stimulated central contributions to digital filtering and discrete-time spectral transforms.

This dynamic development was built upon the convergence of three then-evolving technologies: (i) sampled-data theory and representation of information signals (which led directly to digital telecommunication that provides signal quality independent of transmission distance); (ii) electronic binary computation (aided in early implementation by pulse-circuit techniques from radar design); and, (iii) invention of solid-state devices for exquisite control of electronic current (transistors – which now, through microelectronic materials, scale to systems of enormous size and complexity). This timely convergence was soon followed by optical fiber methods for broadband information transport.

These advances impact an important aspect of human activity – information exchange. And, over man's existence, speech has played a principal role in human communication. Now, speech is playing an increasing role in human interaction with complex information systems. Automatic services of great variety exploit the comfort of voice exchange, and, in the corporate sector, sophisticated audio/video teleconferencing is reducing the necessity of expensive, time-consuming business travel. In each instance an overarching target is a user environment that captures some of the naturalness and spatial realism of face-to-face communication. Again, speech is a core element, and new understanding from diverse research sectors can be brought to bear.

Editors-in-Chief Benesty, Sondhi and Huang have organized a timely engineering handbook to answer this need. They have assembled a remarkable compendium of current knowledge in speech processing. And, this accumulated understanding can be focused upon enlarging the human capacity to deal with a world ever increasing in complexity. Benesty, Sondhi and Huang are renowned researchers in their own right, and they have attracted an international cadre of over 80 fellow authors and collaborators who constitute a veritable *Who's Who* of world leaders in speech processing research. The resulting book provides under one cover authoritative treatments that commence with the basic physics and psychophysics of speech and hearing, and range through the related topics of computational tools, coding, synthesis, recognition, and signal enhancement, concluding with discussions on capture and projection of sound in enclosures. The book can be expected to become a valuable resource for researchers, engineers and speech scientists throughout the global community. It should equally serve teachers and students in human communication, especially delimiting knowledge frontiers where graduate thesis research may be appropriate.

Warren, New Jersey
October 2007

Jim Flanagan



J. L. Flanagan
Professor Emeritus
Electrical and Computer
Engineering
Rutgers University

Editors:

Jacob Benesty
INRS-EMT, University of Quebec
800 de la Gauchetiere Ouest, Suite 6900
Montreal, Quebec, H5A 1K6, Canada
benesty@emt.inrs.ca

M. Mohan Sondhi
Avayalabs Research
233 Mount Airy Road
Basking Ridge, NJ 07920, USA
mms@research.avayalabs.com

Yiteng Huang
Bell Laboratories, Alcatel-Lucent
600 Mountain Avenue
Murray Hill, NJ 07974, USA
arden_huang@ieee.org

Library of Congress Control Number: 2007931999

ISBN: 978-3-540-49125-5 e-ISBN: 978-3-540-49127-9

This work is subject to copyright. All rights reserved, whether the whole or part of the material is concerned, specifically the rights of translation, reprinting, reuse of illustrations, recitation, broadcasting, reproduction on microfilm or in any other way, and storage in data banks. Duplication of this publication or parts thereof is permitted only under the provisions of the German Copyright Law of September, 9, 1965, in its current version, and permission for use must always be obtained from Springer-Verlag. Violations are liable for prosecution under the German Copyright Law.

Springer is a part of Springer Science+Business Media
springer.com

© Springer-Verlag Berlin Heidelberg 2008

The use of designations, trademarks, etc. in this publication does not imply, even in the absence of a specific statement, that such names are exempt from the relevant protective laws and regulations and therefore free for general use.

Product liability: The publisher cannot guarantee the accuracy of any information about dosage and application contained in this book. In every individual case the user must check such information by consulting the relevant literature.

Typesetting and production:
LE-TeX Jelonek, Schmidt&Vöckler GbR, Leipzig
Senior Manager Springer Handbook: Dr. W. Skolaut, Heidelberg
Typography and layout: schreiberVIS, Seeheim
Illustrations: Hippmann GbR, Schwarzenbruck
Cover design: eStudio Calamar Steinen, Barcelona
Cover production: WMXDesign GmbH, Heidelberg
Printing and binding: Stürtz GmbH, Würzburg

Printed on acid free paper

SPIN 11544036 60/3180/YL 5 4 3 2 1 0



Foreword

Over the past three decades digital signal processing has emerged as a recognized discipline. Much of the impetus for this advance stems from research in representation, coding, transmission, storage and reproduction of speech and image information. In particular, interest in voice communication has stimulated central contributions to digital filtering and discrete-time spectral transforms.

This dynamic development was built upon the convergence of three then-evolving technologies: (i) sampled-data theory and representation of information signals (which led directly to digital telecommunication that provides signal quality independent of transmission distance); (ii) electronic binary computation (aided in early implementation by pulse-circuit techniques from radar design); and, (iii) invention of solid-state devices for exquisite control of electronic current (transistors – which now, through microelectronic materials, scale to systems of enormous size and complexity). This timely convergence was soon followed by optical fiber methods for broadband information transport.

These advances impact an important aspect of human activity – information exchange. And, over man's existence, speech has played a principal role in human communication. Now, speech is playing an increasing role in human interaction with complex information systems. Automatic services of great variety exploit the comfort of voice exchange, and, in the corporate sector, sophisticated audio/video teleconferencing is reducing the necessity of expensive, time-consuming business travel. In each instance an overarching target is a user environment that captures some of the naturalness and spatial realism of face-to-face communication. Again, speech is a core element, and new understanding from diverse research sectors can be brought to bear.

Editors-in-Chief Benesty, Sondhi and Huang have organized a timely engineering handbook to answer this need. They have assembled a remarkable compendium of current knowledge in speech processing. And, this accumulated understanding can be focused upon enlarging the human capacity to deal with a world ever increasing in complexity. Benesty, Sondhi and Huang are renowned researchers in their own right, and they have attracted an international cadre of over 80 fellow authors and collaborators who constitute a veritable *Who's Who* of world leaders in speech processing research. The resulting book provides under one cover authoritative treatments that commence with the basic physics and psychophysics of speech and hearing, and range through the related topics of computational tools, coding, synthesis, recognition, and signal enhancement, concluding with discussions on capture and projection of sound in enclosures. The book can be expected to become a valuable resource for researchers, engineers and speech scientists throughout the global community. It should equally serve teachers and students in human communication, especially delimiting knowledge frontiers where graduate thesis research may be appropriate.

Warren, New Jersey
October 2007

Jim Flanagan



J. L. Flanagan
Professor Emeritus
Electrical and Computer
Engineering
Rutgers University

Stephanie Seneff
Massachusetts Institute of Technology
Computer Science and Artificial
Intelligence Laboratory
32 Vassar Street
Cambridge, MA 02139, USA
e-mail: senef@csail.mit.edu

Wade Shen
Massachusetts Institute of Technology
Communication Systems, Information Systems
Technology, Lincoln Laboratory
244 Wood Street
Lexington, MA 02420-9108, USA
e-mail: swade@ll.mit.edu

Elliot Singer
Massachusetts Institute of Technology
Information Systems Technology Group, Lincoln
Laboratory
244 Wood Street
Lexington, MA 02420-9108, USA
e-mail: es@ll.mit.edu

Jan Skoglund
Global IP Solutions
301 Brannan Street
San Francisco, CA 94107, USA
e-mail: jan.skoglund@gipscorp.com

M. Mohan Sondhi
Avayalabs Research
233 Mount Airy Road
Basking Ridge, NJ 07920, USA
e-mail: mms@research.avayalabs.com

Sascha Spors
Deutsche Telekom AG, Laboratories
Ernst-Reuter-Platz 7
10587 Berlin, Germany
e-mail: Sascha.Spors@telekom.de

Ann Spriet
ESAT-SCD/SISTA, K.U. Leuven
Department of Electrical Engineering
Kasteelpark Arenberg 10
3001 Leuven, Belgium
e-mail: ann.spriet@esat.kuleuven.be

Richard Sproat
University of Illinois at Urbana-Champaign
Department of Linguistics
Urbana, IL 61801, USA
e-mail: rws@uiuc.edu

Yannis Stylianou
Institute of Computer Science
Heraklion, Crete 700 13, Greece
e-mail: yannis@csd.uoc.gr

Jes Thyssen
Broadcom Corporation
5300 California Avenue
Irvine, CA 92617, USA
e-mail: jthyssen@broadcom.com

Jay Wilpon
Research AT&T Labs
Voice and IP Services
Florham Park, NJ 07932, USA
e-mail: jgw@research.att.com

Jan Wouters
ExpORL, Department of Neurosciences, K.U. Leuven
O.& N2, Herestraat 49
3000 Leuven, Belgium
e-mail: jan.wouters@med.kuleuven.be

Arie Yeredor
Tel-Aviv University
Electrical Engineering - Systems
Tel-Aviv 69978, Israel
e-mail: arie@eng.tau.ac.il

Steve Young
Cambridge University Engineering Dept
Cambridge, CB21PZ, UK
e-mail: sjy@eng.cam.ac.uk

Victor Zue
Massachusetts Institute of Technology
CSAI Laboratory
32 Vassar Street
Cambridge, MA 02139, USA
e-mail: zue@csail.mit.edu

Contents

List of Abbreviations	XXXI
1 Introduction to Speech Processing	
<i>J. Benesty, M. M. Sondhi, Y. Huang</i>	1
1.1 A Brief History of Speech Processing	1
1.2 Applications of Speech Processing	2
1.3 Organization of the Handbook	4
References	4
Part A Production, Perception, and Modeling of Speech	
2 Physiological Processes of Speech Production	
<i>K. Honda</i>	7
2.1 Overview of Speech Apparatus	7
2.2 Voice Production Mechanisms	8
2.3 Articulatory Mechanisms	14
2.4 Summary	24
References	25
3 Nonlinear Cochlear Signal Processing and Masking in Speech Perception	
<i>J. B. Allen</i>	27
3.1 Basics	27
3.2 The Nonlinear Cochlea	35
3.3 Neural Masking	45
3.4 Discussion and Summary	55
References	56
4 Perception of Speech and Sound	
<i>B. Kollmeier, T. Brand, B. Meyer</i>	61
4.1 Basic Psychoacoustic Quantities	62
4.2 Acoustical Information Required for Speech Perception	70
4.3 Speech Feature Perception	74
References	81
5 Speech Quality Assessment	
<i>V. Grancharov, W. B. Kleijn</i>	83
5.1 Degradation Factors Affecting Speech Quality	84
5.2 Subjective Tests	85
5.3 Objective Measures	90
5.4 Conclusions	95
References	96

Part B Signal Processing for Speech**6 Wiener and Adaptive Filters**

<i>J. Benesty, Y. Huang, J. Chen</i>	103
6.1 Overview.....	103
6.2 Signal Models.....	104
6.3 Derivation of the Wiener Filter.....	106
6.4 Impulse Response Tail Effect.....	107
6.5 Condition Number.....	108
6.6 Adaptive Algorithms.....	110
6.7 MIMO Wiener Filter.....	116
6.8 Conclusions.....	119
References	120

7 Linear Prediction

<i>J. Benesty, J. Chen, Y. Huang</i>	121
7.1 Fundamentals.....	121
7.2 Forward Linear Prediction.....	122
7.3 Backward Linear Prediction.....	123
7.4 Levinson–Durbin Algorithm.....	124
7.5 Lattice Predictor.....	126
7.6 Spectral Representation.....	127
7.7 Linear Interpolation.....	128
7.8 Line Spectrum Pair Representation.....	129
7.9 Multichannel Linear Prediction.....	130
7.10 Conclusions.....	133
References	133

8 The Kalman Filter

<i>S. Gannot, A. Yeredor</i>	135
8.1 Derivation of the Kalman Filter.....	136
8.2 Examples: Estimation of Parametric Stochastic Process from Noisy Observations.....	141
8.3 Extensions of the Kalman Filter.....	144
8.4 The Application of the Kalman Filter to Speech Processing.....	149
8.5 Summary.....	157
References	157

9 Homomorphic Systems and Cepstrum Analysis of Speech

<i>R. W. Schafer</i>	161
9.1 Definitions.....	161
9.2 Z-Transform Analysis.....	164
9.3 Discrete-Time Model for Speech Production.....	165
9.4 The Cepstrum of Speech.....	166
9.5 Relation to LPC.....	169
9.6 Application to Pitch Detection.....	171

9.7 Applications to Analysis/Synthesis Coding.....	172
9.8 Applications to Speech Pattern Recognition.....	176
9.9 Summary.....	180
References	180

**10 Pitch and Voicing Determination of Speech
with an Extension Toward Music Signals**

<i>W. J. Hess</i>	181
10.1 Pitch in Time-Variant Quasiperiodic Acoustic Signals.....	182
10.2 Short-Term Analysis PDAs.....	185
10.3 Selected Time-Domain Methods.....	192
10.4 A Short Look into Voicing Determination.....	195
10.5 Evaluation and Postprocessing.....	197
10.6 Applications in Speech and Music.....	201
10.7 Some New Challenges and Developments.....	203
10.8 Concluding Remarks.....	207
References	208

11 Formant Estimation and Tracking

<i>D. O'Shaughnessy</i>	213
11.1 Historical.....	213
11.2 Vocal Tract Resonances.....	215
11.3 Speech Production.....	216
11.4 Acoustics of the Vocal Tract.....	218
11.5 Short-Time Speech Analysis.....	221
11.6 Formant Estimation.....	223
11.7 Summary.....	226
References	226

12 The STFT, Sinusoidal Models, and Speech Modification

<i>M. M. Goodwin</i>	229
12.1 The Short-Time Fourier Transform.....	230
12.2 Sinusoidal Models.....	242
12.3 Speech Modification.....	253
References	256

13 Adaptive Blind Multichannel Identification

<i>Y. Huang, J. Benesty, J. Chen</i>	259
13.1 Overview.....	259
13.2 Signal Model and Problem Formulation.....	260
13.3 Identifiability and Principle.....	261
13.4 Constrained Time-Domain Multichannel LMS and Newton Algorithms.....	262
13.5 Unconstrained Multichannel LMS Algorithm with Optimal Step-Size Control.....	266
13.6 Frequency-Domain Blind Multichannel Identification Algorithms.....	268
13.7 Adaptive Multichannel Exponentiated Gradient Algorithm.....	276

13.8 Summary	279
References	279

Part C Speech Coding

14 Principles of Speech Coding

<i>W. B. Kleijn</i>	283
14.1 The Objective of Speech Coding	283
14.2 Speech Coder Attributes	284
14.3 A Universal Coder for Speech	286
14.4 Coding with Autoregressive Models	293
14.5 Distortion Measures and Coding Architecture	296
14.6 Summary	302
References	303

15 Voice over IP: Speech Transmission over Packet Networks

<i>J. Skoglund, E. Kozica, J. Linden, R. Hagen, W. B. Kleijn</i>	307
15.1 Voice Communication	307
15.2 Properties of the Network	308
15.3 Outline of a VoIP System	313
15.4 Robust Encoding	317
15.5 Packet Loss Concealment	326
15.6 Conclusion	327
References	328

16 Low-Bit-Rate Speech Coding

<i>A. V. McCree</i>	331
16.1 Speech Coding	331
16.2 Fundamentals: Parametric Modeling of Speech Signals	332
16.3 Flexible Parametric Models	337
16.4 Efficient Quantization of Model Parameters	344
16.5 Low-Rate Speech Coding Standards	345
16.6 Summary	347
References	347

17 Analysis-by-Synthesis Speech Coding

<i>J.-H. Chen, J. Thyssen</i>	351
17.1 Overview	352
17.2 Basic Concepts of Analysis-by-Synthesis Coding	353
17.3 Overview of Prominent Analysis-by-Synthesis Speech Coders	357
17.4 Multipulse Linear Predictive Coding (MPLPC)	360
17.5 Regular-Pulse Excitation with Long-Term Prediction (RPE-LTP)	362
17.6 The Original Code Excited Linear Prediction (CELP) Coder	363
17.7 US Federal Standard FS1016 CELP	367
17.8 Vector Sum Excited Linear Prediction (VSELP)	368
17.9 Low-Delay CELP (LD-CELP)	370

17.10 Pitch Synchronous Innovation CELP (PSI-CELP)	371
17.11 Algebraic CELP (ACELP)	371
17.12 Conjugate Structure CELP (CS-CELP) and CS-ACELP	377
17.13 Relaxed CELP (RCELP) – Generalized Analysis by Synthesis	378
17.14 eX-CELP	381
17.15 iLBC	382
17.16 TSNFC	383
17.17 Embedded CELP	386
17.18 Summary of Analysis-by-Synthesis Speech Coders	388
17.19 Conclusion	390
References	390

18 Perceptual Audio Coding of Speech Signals

<i>J. Herre, M. Lutzky</i>	393
18.1 History of Audio Coding	393
18.2 Fundamentals of Perceptual Audio Coding	394
18.3 Some Successful Standardized Audio Coders	396
18.4 Perceptual Audio Coding for Real-Time Communication	398
18.5 Hybrid/Crossover Coders	403
18.6 Summary	409
References	409

Part D Text-to-Speech Synthesis

19 Basic Principles of Speech Synthesis

<i>J. Schroeter</i>	413
19.1 The Basic Components of a TTS System	413
19.2 Speech Representations and Signal Processing for Concatenative Synthesis	421
19.3 Speech Signal Transformation Principles	423
19.4 Speech Synthesis Evaluation	425
19.5 Conclusions	426
References	426

20 Rule-Based Speech Synthesis

<i>R. Carlson, B. Granström</i>	429
20.1 Background	429
20.2 Terminal Analog	429
20.3 Controlling the Synthesizer	432
20.4 Special Applications of Rule-Based Parametric Synthesis	434
20.5 Concluding Remarks	434
References	434

21 Corpus-Based Speech Synthesis

<i>T. Dutoit</i>	437
21.1 Basics	437

21.2	Concatenative Synthesis with a Fixed Inventory	438
21.3	Unit-Selection-Based Synthesis	447
21.4	Statistical Parametric Synthesis	450
21.5	Conclusion	453
	References	453
22	Linguistic Processing for Speech Synthesis	
	<i>R. Sproat</i>	457
22.1	Why Linguistic Processing is Hard	457
22.2	Fundamentals: Writing Systems and the Graphical Representation of Language	457
22.3	Problems to be Solved and Methods to Solve Them	458
22.4	Architectures for Multilingual Linguistic Processing	465
22.5	Document-Level Processing	465
22.6	Future Prospects	466
	References	467
23	Prosodic Processing	
	<i>J. van Santen, T. Mishra, E. Klabbers</i>	471
23.1	Overview	471
23.2	Historical Overview	475
23.3	Fundamental Challenges	476
23.4	A Survey of Current Approaches	477
23.5	Future Approaches	484
23.6	Conclusions	485
	References	485
24	Voice Transformation	
	<i>Y. Stylianou</i>	489
24.1	Background	489
24.2	Source-Filter Theory and Harmonic Models	490
24.3	Definitions	492
24.4	Source Modifications	494
24.5	Filter Modifications	498
24.6	Conversion Functions	499
24.7	Voice Conversion	500
24.8	Quality Issues in Voice Transformations	501
24.9	Summary	502
	References	502
25	Expressive/Affective Speech Synthesis	
	<i>N. Campbell</i>	505
25.1	Overview	505
25.2	Characteristics of Affective Speech	506
25.3	The Communicative Functionality of Speech	508
25.4	Approaches to Synthesizing Expressive Speech	510
25.5	Modeling Human Speech	512

25.6	Conclusion	515
	References	515

Part E Speech Recognition

26	Historical Perspective of the Field of ASR/NLU	
	<i>L. Rabiner, B.-H. Juang</i>	521
26.1	ASR Methodologies	521
26.2	Important Milestones in Speech Recognition History	523
26.3	Generation 1 – The Early History of Speech Recognition	524
26.4	Generation 2 – The First Working Systems for Speech Recognition	524
26.5	Generation 3 – The Pattern Recognition Approach to Speech Recognition	525
26.6	Generation 4 – The Era of the Statistical Model	530
26.7	Generation 5 – The Future	534
26.8	Summary	534
	References	535
27	HMMs and Related Speech Recognition Technologies	
	<i>S. Young</i>	539
27.1	Basic Framework	539
27.2	Architecture of an HMM-Based Recognizer	540
27.3	HMM-Based Acoustic Modeling	547
27.4	Normalization	550
27.5	Adaptation	551
27.6	Multipass Recognition Architectures	554
27.7	Conclusions	554
	References	555
28	Speech Recognition with Weighted Finite-State Transducers	
	<i>M. Mohri, F. Pereira, M. Riley</i>	559
28.1	Definitions	559
28.2	Overview	560
28.3	Algorithms	567
28.4	Applications to Speech Recognition	574
28.5	Conclusion	582
	References	582
29	A Machine Learning Framework for Spoken-Dialog Classification	
	<i>C. Cortes, P. Haffner, M. Mohri</i>	585
29.1	Motivation	585
29.2	Introduction to Kernel Methods	586
29.3	Rational Kernels	587
29.4	Algorithms	589
29.5	Experiments	591
29.6	Theoretical Results for Rational Kernels	593

29.7 Conclusion	594
References	595
30 Towards Superhuman Speech Recognition	
<i>M. Picheny, D. Nahamoo</i>	597
30.1 Current Status	597
30.2 A Multidomain Conversational Test Set	598
30.3 Listening Experiments	599
30.4 Recognition Experiments	601
30.5 Speculation	607
References	614
31 Natural Language Understanding	
<i>S. Roukos</i>	617
31.1 Overview of NLU Applications	618
31.2 Natural Language Parsing	620
31.3 Practical Implementation	623
31.4 Speech Mining	623
31.5 Conclusion	625
References	626
32 Transcription and Distillation of Spontaneous Speech	
<i>S. Furui, T. Kawahara</i>	627
32.1 Background	627
32.2 Overview of Research Activities on Spontaneous Speech	628
32.3 Analysis for Spontaneous Speech Recognition	632
32.4 Approaches to Spontaneous Speech Recognition	635
32.5 Metadata and Structure Extraction of Spontaneous Speech	640
32.6 Speech Summarization	644
32.7 Conclusions	647
References	647
33 Environmental Robustness	
<i>J. Droppo, A. Acero</i>	653
33.1 Noise Robust Speech Recognition	653
33.2 Model Retraining and Adaptation	656
33.3 Feature Transformation and Normalization	657
33.4 A Model of the Environment	664
33.5 Structured Model Adaptation	667
33.6 Structured Feature Enhancement	671
33.7 Unifying Model and Feature Techniques	675
33.8 Conclusion	677
References	677
34 The Business of Speech Technologies	
<i>J. Wilpon, M. E. Gilbert, J. Cohen</i>	681
34.1 Introduction	682

34.2 Network-Based Speech Services	686
34.3 Device-Based Speech Applications	692
34.4 Vision/Predictions of Future Services – Fueling the Trends	697
34.5 Conclusion	701
References	702
35 Spoken Dialogue Systems	
<i>V. Zue, S. Seneff</i>	705
35.1 Technology Components and System Development	707
35.2 Development Issues	712
35.3 Historical Perspectives	714
35.4 New Directions	715
35.5 Concluding Remarks	718
References	718
Part F Speaker Recognition	
36 Overview of Speaker Recognition	
<i>A. E. Rosenberg, F. Bimbot, S. Parthasarathy</i>	725
36.1 Speaker Recognition	725
36.2 Measuring Speaker Features	729
36.3 Constructing Speaker Models	731
36.4 Adaptation	735
36.5 Decision and Performance	735
36.6 Selected Applications for Automatic Speaker Recognition	737
36.7 Summary	739
References	739
37 Text-Dependent Speaker Recognition	
<i>M. Hébert</i>	743
37.1 Brief Overview	743
37.2 Text-Dependent Challenges	747
37.3 Selected Results	750
37.4 Concluding Remarks	760
References	760
38 Text-Independent Speaker Recognition	
<i>D. A. Reynolds, W. M. Campbell</i>	763
38.1 Introduction	763
38.2 Likelihood Ratio Detector	764
38.3 Features	766
38.4 Classifiers	767
38.5 Performance Assessment	776
38.6 Summary	778
References	779

Part G Language Recognition

39 Principles of Spoken Language Recognition
C.-H. Lee 785
39.1 Spoken Language 785
39.2 Language Recognition Principles 786
39.3 Phone Recognition Followed by Language Modeling (PRLM) 788
39.4 Vector-Space Characterization (VSC) 789
39.5 Spoken Language Verification 790
39.6 Discriminative Classifier Design 791
39.7 Summary 793
References 793

40 Spoken Language Characterization
M. P. Harper, M. Maxwell 797
40.1 Language versus Dialect 798
40.2 Spoken Language Collections 800
40.3 Spoken Language Characteristics 800
40.4 Human Language Identification 804
40.5 Text as a Source of Information on Spoken Languages 806
40.6 Summary 807
References 807

41 Automatic Language Recognition Via Spectral and Token Based Approaches
D. A. Reynolds, W. M. Campbell, W. Shen, E. Singer 811
41.1 Automatic Language Recognition 811
41.2 Spectral Based Methods 812
41.3 Token-Based Methods 815
41.4 System Fusion 818
41.5 Performance Assessment 820
41.6 Summary 823
References 823

42 Vector-Based Spoken Language Classification
H. Li, B. Ma, C.-H. Lee 825
42.1 Vector Space Characterization 826
42.2 Unit Selection and Modeling 827
42.3 Front-End: Voice Tokenization and Spoken Document Vectorization 830
42.4 Back-End: Vector-Based Classifier Design 831
42.5 Language Classification Experiments and Discussion 835
42.6 Summary 838
References 839

Part H Speech Enhancement

43 Fundamentals of Noise Reduction
J. Chen, J. Benesty, Y. Huang, E. J. Diethorn 843
43.1 Noise 843
43.2 Signal Model and Problem Formulation 845
43.3 Evaluation of Noise Reduction 846
43.4 Noise Reduction via Filtering Techniques 847
43.5 Noise Reduction via Spectral Restoration 857
43.6 Speech-Model-Based Noise Reduction 863
43.7 Summary 868
References 869

44 Spectral Enhancement Methods
I. Cohen, S. Gannot 873
44.1 Spectral Enhancement 874
44.2 Problem Formulation 875
44.3 Statistical Models 876
44.4 Signal Estimation 879
44.5 Signal Presence Probability Estimation 881
44.6 A Priori SNR Estimation 882
44.7 Noise Spectrum Estimation 888
44.8 Summary of a Spectral Enhancement Algorithm 891
44.9 Selection of Spectral Enhancement Algorithms 896
44.10 Conclusions 898
References 899

45 Adaptive Echo Cancellation for Voice Signals
M. M. Sondhi 903
45.1 Network Echoes 904
45.2 Single-Channel Acoustic Echo Cancellation 915
45.3 Multichannel Acoustic Echo Cancellation 921
45.4 Summary 925
References 926

46 Dereverberation
Y. Huang, J. Benesty, J. Chen 929
46.1 Background and Overview 929
46.2 Signal Model and Problem Formulation 931
46.3 Source Model-Based Speech Dereverberation 932
46.4 Separation of Speech and Reverberation via Homomorphic Transformation 936
46.5 Channel Inversion and Equalization 937
46.6 Summary 941
References 942

47 Adaptive Beamforming and Postfiltering
S. Gannot, I. Cohen..... 945
47.1 Problem Formulation..... 947
47.2 Adaptive Beamforming..... 948
47.3 Fixed Beamformer and Blocking Matrix..... 953
47.4 Identification of the Acoustical Transfer Function..... 955
47.5 Robustness and Distortion Weighting..... 960
47.6 Multichannel Postfiltering..... 962
47.7 Performance Analysis..... 967
47.8 Experimental Results..... 972
47.9 Summary..... 972
47.A Appendix: Derivation of the Expected Noise Reduction
for a Coherent Noise Field..... 973
47.B Appendix: Equivalence Between Maximum SNR
and LCMV Beamformers..... 974
References..... 975

48 Feedback Control in Hearing Aids
A. Spriet, S. Doclo, M. Moonen, J. Wouters..... 979
48.1 Problem Statement..... 980
48.2 Standard Adaptive Feedback Canceller..... 982
48.3 Feedback Cancellation Based on Prior Knowledge
of the Acoustic Feedback Path..... 986
48.4 Feedback Cancellation Based on Closed-Loop System Identification..... 990
48.5 Comparison..... 995
48.6 Conclusions..... 997
References..... 997

49 Active Noise Control
S. M. Kuo, D. R. Morgan..... 1001
49.1 Broadband Feedforward Active Noise Control..... 1002
49.2 Narrowband Feedforward Active Noise Control..... 1006
49.3 Feedback Active Noise Control..... 1010
49.4 Multichannel ANC..... 1011
49.5 Summary..... 1015
References..... 1015

Part I Multichannel Speech Processing

50 Microphone Arrays
G. W. Elko, J. Meyer..... 1021
50.1 Microphone Array Beamforming..... 1021
50.2 Constant-Beamwidth Microphone Array System..... 1029
50.3 Constrained Optimization of the Directional Gain..... 1030
50.4 Differential Microphone Arrays..... 1031
50.5 Eigenbeamforming Arrays..... 1034

50.6 Adaptive Array Systems..... 1037
50.7 Conclusions..... 1040
References..... 1040

51 Time Delay Estimation and Source Localization
Y. Huang, J. Benesty, J. Chen..... 1043
51.1 Technology Taxonomy..... 1043
51.2 Time Delay Estimation..... 1044
51.3 Source Localization..... 1054
51.4 Summary..... 1061
References..... 1062

52 Convolutional Blind Source Separation Methods
M. S. Pedersen, J. Larsen, U. Kjems, L. C. Parra..... 1065
52.1 The Mixing Model..... 1066
52.2 The Separation Model..... 1068
52.3 Identification..... 1071
52.4 Separation Principle..... 1071
52.5 Time Versus Frequency Domain..... 1076
52.6 The Permutation Ambiguity..... 1078
52.7 Results..... 1084
52.8 Conclusion..... 1084
References..... 1084

53 Sound Field Reproduction
R. Rabenstein, S. Spors..... 1095
53.1 Sound Field Synthesis..... 1095
53.2 Mathematical Representation of Sound Fields..... 1096
53.3 Stereophony..... 1100
53.4 Vector-Based Amplitude Panning..... 1103
53.5 Ambisonics..... 1104
53.6 Wave Field Synthesis..... 1109
References..... 1113

Acknowledgements..... 1115
About the Authors..... 1117
Detailed Contents..... 1133
Subject Index..... 1161

50. Microphone Arrays

G. W. Elko, J. Meyer

This chapter introduces various types of microphone array beamforming systems and discusses some of the fundamental theory of their operation, design, implementation, and limitations. It is shown that microphone arrays have the ability to offer directional gains that can significantly improve the quality of signal pickup in reverberant and noisy environments.

Hands-free audio communication is now a major feature in mobile communication systems as well as audio and video conferencing systems. One problem that becomes evident to users of these systems is the decrease in communication quality due to the pickup of room reverberation and background noise. In the past, this problem was dealt with by using microphones placed close to the desired talker or source. Although this simple solution has proven to be quite effective, it also has its drawbacks. First, it is not always possible or desirable to place the microphone very close to the talker's mouth. Second, by placing the microphone close to the talker's mouth, one has to deal with rapid level variation as the talker moves his or her mouth relative to the microphone. Third is the negative impact of speech plosives (air-flow transients generated by plosive sounds) and forth, microphone structure-borne handling noise has a detrimental effect. Finally, for directional microphone elements, there is also a nearfield

50.1 Microphone Array Beamforming.....	1021
50.1.1 Delay-and-Sum Beamforming.....	1023
50.1.2 Filter-and-Sum Beamforming	1028
50.1.3 Arrays with Directional Elements.....	1028
50.2 Constant-Beamwidth Microphone Array System	1029
50.3 Constrained Optimization of the Directional Gain	1030
50.4 Differential Microphone Arrays	1031
50.5 Eigenbeamforming Arrays	1034
50.5.1 Spherical Array.....	1034
50.5.2 Eigenbeamformer	1035
50.5.3 Modal Beamformer.....	1037
50.6 Adaptive Array Systems	1037
50.6.1 Constrained Broadband Arrays.....	1038
50.7 Conclusions	1040
References	1040

proximity effect (where the frequency response of the microphone is modulated by the relative position of the microphone to the mouth). With these issues in mind, it is of interest to investigate other potential solutions. One solution is to use beamforming microphone arrays, which can offer significant directional gain so as to result in similar audio performance to that of closely placed microphones.

50.1 Microphone Array Beamforming

The propagation of acoustic waves in space is a function of both space and time coordinates. In general, the mathematical representation of a propagating acoustic wave is a four-dimensional function: three spatial dimensions and one time variable. Acoustic wave propagation can be modeled by a linearized scalar acoustic wave equation that describes the relationships between the physical quantities of acoustic pressure and density variation of the medium [50.1],

$$\nabla^2 p = \frac{1}{c^2} \frac{\partial^2 p}{\partial t^2}, \quad (50.1)$$

where p is the instantaneous acoustic pressure fluctuation of the sound and c is the propagation speed of sound in the medium. There are a few major underlying assumptions in the derivation of (50.1), but this simplified linear model is adequate for laying the foundation for array beamforming of acoustic signals. The dimen-

sionalities of the wave equation can easily be seen from (50.1), where p is a function of three space variables and one time variable. Therefore the scalar acoustic pressure field in space can be represented as $p(\mathbf{r}, t)$, where $\mathbf{r}$ is the measurement position in the acoustic field and t is the time dependence. The scalar acoustic pressure field must satisfy (50.1) at all points in space. By using the spatial Fourier transform, the acoustic field can also equivalently be represented in the wavevector–frequency domain. This representation has been widely used in the fields of geophysics, structural, underwater and aeroacoustics, and can be quite useful in the analysis and design of microphone array beamformers.

Applying the four-dimensional Fourier transform to the time–space scalar acoustic field yields,

$$P(\mathbf{k}, \omega) = \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} p(\mathbf{r}, t) e^{-i(\omega t - \mathbf{k}^T \mathbf{r})} d\mathbf{r} dt, \quad (50.2)$$

where the acoustic pressure P is capitalized to indicate a transform to the frequency domain, $\mathbf{k}$ is the wavevector and the superscript ‘T’ represents the transpose operator. It may seem that this transformation has unnecessarily complicated the description of the scalar acoustic pressure field, but as will be seen later, it has transformed the field description into a form that makes the analysis of acoustic field space–time functions analogous to the field of multidimensional signal processing. Since the Fourier transform is a linear transformation, one can write an inverse relationship,

$$p(\mathbf{r}, t) = \frac{1}{(2\pi)^4} \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} P(\mathbf{k}, \omega) e^{i(\omega t - \mathbf{k}^T \mathbf{r})} d\mathbf{k} d\omega, \quad (50.3)$$

Equation (50.3) directly shows that any acoustic field $p(\mathbf{r}, t)$ can be represented as an infinite number of propagating plane waves with appropriate complex weighting. This interpretation can easily be seen from the equation for a single propagating plane wave with frequency ω_0 and wavevector $\mathbf{k}_0$,

$$p(\mathbf{r}, t) = A_0 e^{-i(\omega_0 t - \mathbf{k}_0^T \mathbf{r})}, \quad (50.4)$$

where A_0 is the plane-wave amplitude. Note that for generality, one can allow A_0 to be complex, but for this discussion it is assumed that A_0 is real. The wavenumber–frequency spectrum of this single propagating plane wave is simply,

$$P(\mathbf{k}, \omega) = A_0 \delta(\mathbf{k} - \mathbf{k}_0) \delta(\omega - \omega_0). \quad (50.5)$$

Equation (50.5) shows the Fourier mapping of infinite continuous functions in both time and space to be a point in the wavevector–frequency space.

Now consider a general weighting function of the space–time pressure distribution with a four-dimensional linear shift-invariant filter having an impulse response $h(\mathbf{r}, t)$. Then, the output signal is the convolution of the space–time acoustic pressure signal and the spatial weighting function such that,

$$y(\mathbf{r}, t) = \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} h(\mathbf{r} - \hat{\mathbf{r}}, t - \tau) p(\hat{\mathbf{r}}, \tau) d\hat{\mathbf{r}} d\tau. \quad (50.6)$$

Equation (50.6) is a filtering operation that can equivalently be represented in the wavevector–frequency domain as,

$$Y(\mathbf{k}, \omega) = H(\mathbf{k}, \omega) P(\mathbf{k}, \omega). \quad (50.7)$$

Equation (50.6) and (50.7) show the direct analogy between the space–time wavevector–frequency representation of spatial filtering and the well-known results from multidimensional linear systems theory. Thus, by using specific spatial weighting functions, one can filter the acoustic field to investigate the propagating directions, amplitude, and phases of plane waves traveling in space. Filtering the spatiotemporal acoustic scalar pressure field is known in the field of array signal processing as *beamforming*. Thus, if one wants to look at plane waves propagating from the direction of the unit vector $\mathbf{k}_0/\|\mathbf{k}_0\|$, one would design a filter such that,

$$H(\mathbf{k}, \omega) = \delta(\mathbf{k} - \mathbf{k}_0) G(\omega), \quad (50.8)$$

where $G(\omega)$ is the desired frequency response to waves propagating in the direction of the wavevector $\mathbf{k}_0$.

Although it may seem limiting to use plane waves as the underlying basis functions for the representation of a sound-field, it should be noted that other orthogonal representations such as spherical and cylindrical basis functions can themselves be represented as a series of plane waves.

The representation of the wavevector–frequency domain is a natural framework for analysis of the spatial and frequency filtering of beamforming arrays, as will be seen in the following.

In spatial filtering by beamforming, one deals with spatial apertures. An aperture is a region over which energy is received. Where an aperture can either be continuous, as in parabolic dishes, or discretely realized as in microphone arrays with multiple microphones, mixtures of continuous and discrete apertures are possible.

In fact, even discrete microphone arrays are a mixture since no microphone is a perfect point receiver. Although sampled aperture systems must generally perform more signal processing, they offer several advantages over continuous aperture systems. A main advantage in using sampled apertures is that they allow for the possibility of using digital signal processing on the individual array signals. Arrays can be electronically steered, can form multiple simultaneous beams, and can be made adaptive purely by electronic means.

50.1.1 Delay-and-Sum Beamforming

Delay-and-sum beamforming, also known as *classical beamforming*, is one of the simplest and oldest techniques for realizing directional array systems. Although not fundamentally limited in bandwidth, early delay-and-sum arrays were used in *narrowband* operation to focus arrays onto a particular point or direction. Since a time delay for narrowband applications can be accomplished with a unique phase shift for each element, narrowband beamforming is typically implemented with phase shifts and is therefore commonly referred to as *phased-array* beamforming. One typical application for microphone arrays is to pick up speech or acoustic signals that are wideband (the desired signals cover many octaves in bandwidth). Therefore most microphone arrays for speech acquisition are implemented as delay-sum beamformers since they are inherently wideband.

Both continuous and discrete beamformers can be seen as implementations of wavevector–frequency filtering as discussed in the previous section. It should be noted that these two beamformers are not mutually exclusive; one can, for instance, have a discrete set of continuous transducers so that the beamformer is a mix of continuous and discrete sampling. Of special interest here is the spatial and frequency response of a finite array of microphones that sample the acoustic field. A classic delay-sum beamformer uses the sum of weighted and delayed samples from an array of N discretely sampled points of the pressure field. In general, a delay-sum beamformer output y is formed as,

$$y(t) = \sum_{n=1}^N w_n s(\mathbf{r}_n, t - \tau_n), \quad (50.9)$$

where the weights w_n are real and the time delay τ_n is applied to the measured signals $s(\mathbf{r}_n, t)$, at microphone n . If the incident field is a planewave with wavevector $\mathbf{k}_0$, amplitude A_0 and frequency ω_0 , then,

$$s(\mathbf{r}_n, t) = A_0 e^{i(\omega_0 t - \mathbf{k}_0^T \mathbf{r}_n)}, \quad (50.10)$$

and

$$y(t) = A_0 e^{i\omega_0 t} \sum_{n=1}^N w_n e^{-i(\omega_0 \tau_n + \mathbf{k}_0^T \mathbf{r}_n)}. \quad (50.11)$$

The real weights w_n scale the signals measured at the positions $\mathbf{r}_n$. Thus, from (50.9), the origin of the term *delay-and-sum* can be seen as simply a description of how the classical delay-sum beamformer is realized.

For a causal delay-sum beamformer, the exponent in (50.11) must be less than or equal to zero for all n . Typically, one element position is selected as the spatial origin. This reference element position therefore defines the vectors $\mathbf{r}_n$. Since the array can be steered to any direction in space and one can select any position to define $\mathbf{r}_n$, an additional delay may be required in order to maintain causality. (Recall that $\mathbf{k}_0^T \mathbf{r}_n$ can be either positive or negative depending on the direction of the incident wave relative to the array.) If an end element is selected as the spatial origin, then the causal delay is equal to the maximum time that an acoustic wave takes to transit the array. Smaller causal delays are also possible depending on which microphone position in the array is chosen as the spatial origin reference and the maximum desired steering angle.

If we set the causal time delay equal to T_0 , then the output for an incident plane wave with angular frequency ω_0 and wavevector $\mathbf{k}_0$ can be written as,

$$y(t) = A_0 e^{i\omega_0 t} \sum_{n=1}^N w_n e^{-i[\omega_0(T_0 + \tau_n) + \mathbf{k}_0^T \mathbf{r}_n]}. \quad (50.12)$$

The maximum plane-wave response corresponds to the direction where the delays τ_n compensate for the propagation delay $\mathbf{k}_0^T \mathbf{r}_n$ of a plane wave propagating with wavevector $\mathbf{k}_0$. This is done by setting the values of τ_n such that these delays offset the time lead (or lag) of a desired direction plane wave propagating over the array. Thus, setting the delays to steer the array to the direction of $\mathbf{k}_0$, means that the last two terms in the exponential in (50.12) cancel out. The delayed and summed output $y(t)$ is now such that all microphone signals propagating from the desired direction are added in phase. This direction corresponds to a maximum amplitude output for any selection of τ_n . Plane waves propagating from directions other than $\mathbf{k}_0$ will result in the addition of position-dependent phase variations, and as a result, the output amplitude will be smaller due to destructive interference.

To summarize, classical delay-and-sum beamforming uses delays between each array element that compensate for differences in the propagation delay of

array beam steering [51.1–4] and a good example of the latter is automatic camera pointing for videoconferencing [51.5–8].

The problem of locating radiative point sources using sensor arrays has long been of great research interest given its theoretical as well as practical importance in a great variety of applications, e.g., radar [51.9, 10], underwater sonar [51.11], and seismology [51.12]. In these applications, source localization is more commonly referred to as direction of arrival (DOA) estimation. A class of celebrated approaches is based on high-resolution spectral analysis, including Capon's minimum variance (MV) spectral estimation method [51.13], and eigenanalysis-based techniques, such as the popular multiple signal classification (MUSIC) algorithm [51.14]. These methods perform statistical fit for DOA with respect to a spatio-spectral correlation matrix derived from the signals received at the sensors. By assumption, the source signal needs to be statistically stationary and narrowband, and the source is located in the far field of the sensor array. However, these basic premises are barely met by speech sources. Moreover, the multipath effect is not taken into account in the formulation of DOA estimation. Therefore, the high-resolution DOA estimation algorithms were found to be incompetent for speech source localization, particularly in reverberant acoustic environments.

For speech source localization, microphone array beam scanning and time delay estimation (TDE)-based localization methods are the two most widely used approaches. A beamformer is a spatial filter that operates on the microphone outputs in order to enhance the signal coming from one direction while suppressing noise and interference from other directions. Therefore, steering a microphone array beamformer and scanning across a room for the highest-energy output leads to an estimate of the direction of an active speech source. Using two arrays and by intersecting the two corresponding direction estimates, the source location

is determined [51.15, 16]. While the beam scanning method has the remarkable advantage that it can easily be extended to the case of simultaneously localizing multiple speech sources [51.17], it has some intrinsic drawbacks. Since speech is a typical broadband signal, ideally a broadband beamformer whose beam pattern would be the same across the entire speech spectrum needs to be developed, which is a technically challenging problem. In addition, most beamforming algorithms assume that the speech sound source is in the far field in order to make the design problem analytically tractable. When the speech source is close to the microphone array, performance degradation can be expected. On the contrary, if the speech source is in the far field, the localization resolution may not be satisfactory as the source direction is only selected from a set of discrete beam scanning angles. The number of beam scanning angles is therefore a trade-off between resolution and computational complexity.

Alternatively, a TDE-based source localization algorithm involves a two-step procedure [51.18]. In the first step, a set of relative time differences of arrival (TDOAs) for different microphone pairs are calculated. In the second step, the acoustic source location is estimated from the TDOAs and using the a priori knowledge about the locations of the microphones. TDE-based source localization algorithms have many merits: they cope well with both narrowband and broadband source signals; their localization resolution can be flexibly adjusted by varying the sampling rate and microphone array size; the effect of room reverberation is only of concern in the first-step TDE and several recently developed robust TDE algorithms are promising for future practical use; and in general they are computationally efficient. Therefore, after continuous investigation over the last two decades, TDE-based speech source localization schemes have become the technique of choice, especially in recent digital systems. In this chapter, an overview of the state of the art of this class of source localization methods is presented.

Depending on the surrounding acoustic environment, there are two signal models for the TDE problem: the ideal free-field model and the real reverberant model. The former assumes no room reverberation while the latter uses an acoustic channel impulse response, usually a finite impulse response (FIR) filter, to describe the effect of room reverberation.

Ideal Free-Field Model

In an anechoic open space, as shown in Fig. 51.1a, the speech source signal $s(k)$ propagates radiatively and the sound level falls off as a function of distance from the source. Then the signal captured by the n -th microphone at time k can be expressed as follows:

$$x_n(k) = \alpha_n s(k - \tau_n) + b_n(k), \quad n = 1, 2, \dots, N, \quad (51.1)$$

where α_n ($0 \leq \alpha_n \leq 1$) is an attenuation factor due to propagation loss, τ_n is the propagation time, and $b_n(k)$ is additive noise. The time difference of arrival (TDOA) between the i -th and j -th microphones is defined as,

$$\tau_{ij} \triangleq \tau_i - \tau_j, \quad i, j = 1, 2, \dots, N. \quad (51.2)$$

The noise signal $b_n(k)$ is presumed to be a zero-mean, white, Gaussian random process, and is uncorrelated with the source signal as well as the noise signals at other microphones.

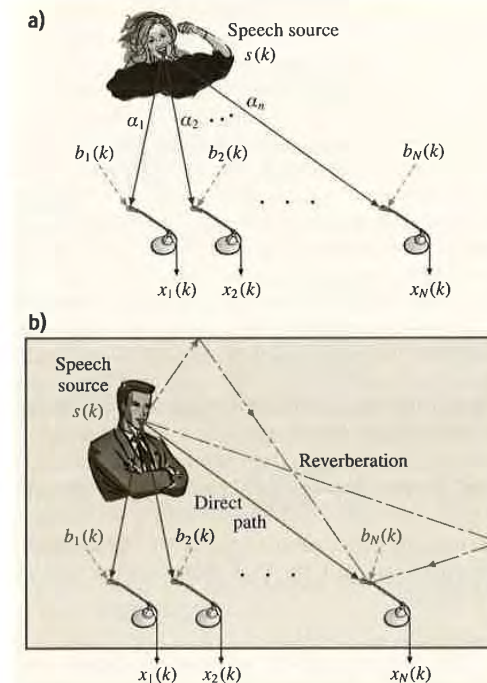


Fig. 51.1a,b Illustration of the two acoustic signal models for time delay estimation: (a) ideal free-field model and (b) real reverberant model

Real Reverberant Model

While the ideal free-field model has the merit of being simple, it does not take into account room reverberation. Therefore, in a real reverberant environment, the ideal free-field model is inadequate and problematic, and we need a more-comprehensive and more-informative alternative to describe the effect of multipath propagation. The real reverberant model treats the acoustic impulse response with an FIR filter, as illustrated by Fig. 51.1b. In such a single-input multiple-output (SIMO) system, the n -th microphone signal is given by

$$x_n(k) = h_n * s(k) + b_n(k), \quad n = 1, 2, \dots, N, \quad (51.3)$$

where h_n is the n -th channel impulse response, and the symbol $*$ denotes the linear convolution operator. In vector/matrix form, (51.3) can be rewritten as

$$\mathbf{x}_n(k) = \mathbf{H}_n \cdot \mathbf{s}(k) + \mathbf{b}_n(k), \quad n = 1, 2, \dots, N, \quad (51.4)$$

where

$$\begin{aligned} \mathbf{x}_n(k) &= [x_n(k) \cdots x_n(k-L+1)]^T, \\ \mathbf{H}_n &= \begin{pmatrix} h_{n,0} & \cdots & h_{n,L-1} & \cdots & 0 \\ \vdots & \ddots & \vdots & \ddots & \vdots \\ 0 & \cdots & h_{n,0} & \cdots & h_{n,L-1} \end{pmatrix}, \\ \mathbf{s}(k) &= [s(k) \quad s(k-1) \quad \cdots \quad s(k-L+1) \\ &\quad \cdots \quad s(k-2L+2)]^T, \\ \mathbf{b}_n(k) &= [b_n(k) \quad \cdots \quad b_n(k-L+1)]^T, \end{aligned}$$

where $[\cdot]^T$ denotes the transpose of a vector or a matrix, and L is the length of the longest channel impulse response in this SIMO system.

In contrast to the ideal free-field model, the time delay τ_n in the real reverberant model is an implicit or hidden parameter. Using such a model, TDE can be performed only after the SIMO system is *blindly* identified (since the source signal is unknown), which looks like a more-difficult problem but is fortunately not insurmountable.

51.2.2 The Family of the Generalized Cross-Correlation Methods

The generalized cross-correlation (GCC) algorithm proposed by Knapp and Carter [51.19] is so far the most widely used approach to TDE. It employs the ideal free-field model (51.1) and considers only two microphones,