

1. Record Nr.	UNINA9910811945903321
Titolo	Advances in digital speech transmission / / edited by Rainer Martin, Ulrich Heute, Christiane Antweiler
Pubbl/distr/stampa	Chichester, West Sussex, England ; ; Hoboken, NJ, : John Wiley & Sons, c2008
ISBN	9786611321987 9781281321985 1281321982 9780470727188 0470727187 9780470727171 0470727179
Edizione	[1st ed.]
Descrizione fisica	1 online resource (573 p.)
Altri autori (Persone)	MartinRainer HeuteUlrich AntweilerChristiane
Disciplina	621.39/9
Soggetti	Speech processing systems Signal processing - Digital techniques
Lingua di pubblicazione	Inglese
Formato	Materiale a stampa
Livello bibliografico	Monografia
Note generali	Description based upon print version of record.
Nota di bibliografia	Includes bibliographical references (p. 525-528) and index.
Nota di contenuto	-- List of Contributors xxi -- Preface xxvii -- 1 Introduction 1 /Rainer Martin, Ulrich Heute, Christiane Antweiler -- I Speech Quality Assessment 7 -- 2 Speech-Transmission Quality: Aspects and Assessment for Wideband vs. Narrowband Signals 9 /Ulrich Heute -- 2.1 Introduction 9 -- 2.2 Speech Signals . 10 -- 2.3 Telephone-Band Speech Signals 11 -- 2.4 Wideband Speech Signals 14 -- 2.5 Speech-Quality Assessment 25 -- 2.6 Wideband Speech-Quality Assessment 30 -- 2.7 Concluding Remarks 43 -- Bibliography 44 -- 3 Parametric Quality Assessment of Narrowband Speech in Mobile Communication Systems 51 /Marc Werner -- 3.1 Introduction 51 -- 3.2 Simulations of GSM and UMTS Speech Transmissions 58 -- 3.3 Speech Quality Measures based on Transmission Parameters 65 -- 3.4 Discussion and Conclusions 73 -- Bibliography 73 -- II Adaptive Algorithms in

Acoustic Signal Processing	77
4 Kalman Filtering in Acoustic Echo Control: A Smooth Ride on a Rocky Road	79 /Gerald Enzner
4.1 Introduction	79
4.2 A Comprehensive Theory of Acoustic Echo Control	85
4.3 The Kalman Filter for Conditional Mean and Covariance Estimation	90
4.4 AEC Performance of the Frequency-Domain Adaptive Kalman Filter	100
4.5 Discussion and Conclusions	102
Bibliography	103
5 Noise Reduction - Statistical Analysis and Control of Musical Noise	107 /Colin Breithaupt, Rainer Martin
5.1 Introduction	107
5.2 Speech Enhancement in the DFT Domain	109
5.3 Measurement and Assessment of Unnatural Fluctuations	115
5.4 Avoidance of Processing Artifacts	120
5.5 Control of Spectral Fluctuations in the Cepstral Domain	123
5.6 Discussion and Conclusions	128
5.7 Acknowledgements	129
5.8 Appendix	129
Bibliography	131
6 Acoustic Source Localization with Microphone Arrays	135 /Nilesh Madhu, Rainer Martin
6.1 Introduction	135
6.2 SignalModel	136
6.3 Localization Approach Taxonomy	141
6.4 Indirect Localization Approaches	141
6.5 Direct Localization Approaches	148
6.6 Evaluation of Localization Algorithms	156
6.7 Conclusions	166
Bibliography	166
7 Multi-Channel System Identification with Perfect Sequences / Theory and Applications /	171
/Christiane Antweiler	
7.1 Introduction	171
7.2 System Identification with Perfect Sequences	174
7.3 Multi-Channel System Identification	185
7.4 Applications	191
7.5 Discussion and Conclusions	195
Bibliography	195
III Speech Coding for Heterogeneous Networks	199
8 Embedded Speech Coding: From G.711 to G.729.1	201 /Bernd Geiser, Stephane Ragot, Herve Taddei
8.1 Introduction	201
8.2 Theory and Tools of Embedded Speech Coding	203
8.3 Embedded Speech Coding Methods	212
8.4 Standardized Embedded Speech Coders	219
8.5 Network Aspects of Embedded Speech Coding	232
8.6 Conclusions and Perspectives	237
Bibliography	238
9 Backwards Compatible Wideband Telephony	249 /Peter Jax
9.1 Introduction	249
9.2 From Narrowband Telephony to Wideband Telephony	250
9.3 Stand-Alone Bandwidth Extension	254
9.4 Embedded Wideband Coding Using Bandwidth Extension Techniques	257
9.5 Combination of Bandwidth Extension with Watermarking	262
9.6 Advanced Transmission of Highband Parameters	267
9.7 Conclusions	274
Bibliography	274
IV Joint Source-Channel Coding	279
10 Parameter Models and Estimators in Soft Decision Source Decoding	281 /Tim Fingscheidt
10.1 Introduction	281
10.2 Overview to Soft Decision Source Decoding	283
10.3 The Markovian Parameter Model	287
10.4 Basic Extrapolative Estimators	290
10.5 Joint Extrapolative Estimation of Two Different Parameters	298
10.6 Extrapolative Estimation with Repeated Parameter Transmission	301
10.7 Interpolative Estimation of a Parameter	304
10.8 Discussion and Conclusions	307
Bibliography	307
11 Optimal MMSE Estimation for Vector Sources with Spatially and Temporally Correlated Elements	311 /Stefan Heinen, Marc Adrat
11.1 Introduction	311
11.2 Source Model	312
11.3 Transmission Channel	316
11.4 Optimal MMSE Parameter Estimator	316
11.5 Near-Optimal MMSE Parameter Estimator	320
11.6 Illustrative Comparison	323
11.7 Simulation Results	325
11.8 Conclusions	327
Bibliography	327
12 Source Optimized Channel Codes & Source Controlled Channel Decoding	329 /Stefan Heinen, Thomas Hindelang
12.1 Introduction	329
12.2 The Transmission System Used as Reference	330
12.3 Source Optimized Channel Coding (SOCC)	332
12.4 Source Controlled Channel Decoding (SCCD)	341
12.5 Comparison of SOCC versus SCCD	357
12.6 Conclusions	

362 -- Bibliography	363 -- 13 Iterative Source-Channel Decoding & Turbo DeCodulation
365 /Marc Adrat, Thorsten Clevorn, Laurent Schmalen	-- 13.1 Introduction
365 -- 13.2 The Key of the Turbo Principle: Extrinsic Information	366 -- 13.3 Iterative Source-Channel Decoding (ISCD)
379 -- 13.4 Turbo DeCodulation (TDeC)	387 -- 13.5 Conclusions
394 -- Bibliography	395 -- V Speech Processing in Hearing Instruments
399 -- 14 Binaural Signal Processing in Hearing Aids	401 /Volkmar Hamacher, Ulrich Kornagel, Thomas Lotter, Henning Puder
-- 14.1 Introduction	401 -- 14.2 Wireless System for Hearing Aids
405 -- 14.3 Binaural Classification Systems	410 -- 14.4 Binaural Beamformer
415 -- 14.5 Blind Source Separation (BSS): An Application for a Binaural Directional Microphone Array in Hearing Aids	422 -- 14.6 Conclusions
427 -- Bibliography	428 -- 15 Auditory-profile-based Physical Evaluation of Multi-microphone Noise Reduction Techniques in Hearing Instruments
431 /Koen Eneman, Arne Leijon, Simon Doclo, Ann Spriet, Marc Moonen, Jan Wouters	-- 15.1 Introduction
431 -- 15.2 Multi-microphone Noise Reduction in Hearing Instruments	434 -- 15.3 Auditory-profile-based Physical Evaluation
441 -- 15.4 Test Conditions	449 -- 15.5 Simulation Results
450 -- 15.6 Discussion	452 -- 15.7 Conclusions
455 -- Bibliography	456 -- VI Speech Processing for Human / Machine Interfaces
459 -- 16 Automatic Speech Recognition in Adverse Acoustic Conditions	461 /Hans-Gunter Hirsch
-- 16.1 Introduction	461.
16.2 Structure of Speech Recognition Systems	462 -- 16.3 Acoustic Scenarios during Speech Input
468 -- 16.4 Improving the Recognition Performance in Adverse Conditions	476 -- 16.5 Conclusions
493 -- Bibliography	494 -- 17 Speaker Classification for Next-Generation Voice-Dialog Systems
497 /Felix Burkhardt, Florian Metze, Joachim Stegmann	-- 17.1 Introduction
497 -- 17.2 Speaker Classification	498 -- 17.3 Detection of Age and Gender
505 -- 17.4 Detection of Anger	510 -- 17.5 Applications in IVR Systems
517 -- 17.6 Discussion and Conclusion	523 -- Bibliography
525 -- Index	529 -- Permissions List
541.	

Sommario/riassunto

Speech processing and speech transmission technology are expanding fields of active research. New challenges arise from the 'anywhere, anytime' paradigm of mobile communications, the ubiquitous use of voice communication systems in noisy environments and the convergence of communication networks toward Internet based transmission protocols (such as Voice over IP). As a consequence, new speech coding, new enhancement and error concealment, and new quality assessment methods are emerging. Advances in Digital Speech Transmission provides an up-to-date overview of the field, including topics such as speech coding in heterogeneous communication networks, wideband coding, and the quality assessment of modern speech transmission systems. Provides an insight into the latest developments in speech processing and speech transmission, making it an essential reference to those working in these fields Offers a balanced overview of technology and applications Presents a unique in-depth discussion of soft-decision source decoding, joint source-channel coding and Turbo DeCodulation techniques Explains speech signal processing in hearing instruments and human-machine interfaces from applications point of view Covers speech coding for Voice over IP, source localization, digital hearing aids and speech processing for automatic speech recognition Advances in Digital Speech Transmission serves as an essential link between the basics and the type of technology and applications (prospective) engineers work on in industry labs and academia. The book will also be of interest to advanced students, researchers, and other professionals who need to

brush up their knowledge in this field.
