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Autore	Mann, Michael <1942- >
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2. Record Nr.	UNINA990001168690403321
Autore	Cartan, Elie
Titolo	Le parallelisme absolu et la theorie unitaire du champ. / de Cartan Elie. s
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Collana	Actualités scientifiques et industrielles
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3. Record Nr.	UNISALENT0991001911589707536
Autore	Chiari, Alberto
Titolo	L'autore della Nencia da Barberino : ristampa anastatica dell'edizione del 1848 / Alberto Chiari, Italiano Marchetti
Pubbl/distr/stampa	Firenze : Presso l'Accademia della Crusca, 1991
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Nota di bibliografia	Includes bibliographical references and index.
Nota di contenuto	<p>Foreword -- Preface -- 1 Introduction -- 1.1 Research and Applications in Academia and Industry -- 1.2 Challenges in Distant Speech Recognition -- 1.3 System Evaluation -- 1.4 Fields of Speech Recognition -- 1.5 Robust Perception -- 1.6 Organizations, Conferences and Journals -- 1.7 Useful Tools, Data Resources and Evaluation Campaigns -- 1.8 Organization of this Book -- 1.9 Principal Symbols used Throughout the Book -- 1.10 Units used Throughout the Book -- 2 Acoustics -- 2.1 Physical Aspect of Sound -- 2.2 Speech Signals -- 2.3 Human Perception of Sound -- 2.4 The Acoustic Environment -- 2.5 Recording Techniques and Sensor Configuration -- 2.6 Summary and Further Reading -- 2.7 Principal Symbols -- 3 Signal Processing and Filtering Techniques -- 3.1 Linear Time-Invariant Systems -- 3.2 The Discrete Fourier Transform -- 3.3 Short-Time Fourier Transform -- 3.4 Summary and Further Reading -- 3.5 Principal Symbols -- 4 Bayesian Filters -- 4.1 Sequential Bayesian Estimation -- 4.2 Wiener Filter -- 4.3 Kalman Filter and Variations -- 4.4 Particle Filters -- 4.5 Summary and Further Reading -- 4.6 Principal Symbols -- 5 Speech Feature Extraction -- 5.1 Short-Time Spectral Analysis -- 5.2 Perceptually Motivated Representation -- 5.3 Spectral Estimation and Analysis -- 5.4 Cepstral Processing -- 5.5 Comparison between Mel Frequency, Perceptual LP and warped MVDR Cepstral Coefficient Frontends -- 5.6 Feature Augmentation -- 5.7 Feature Reduction -- 5.8 Feature-Space Minimum Phone Error -- 5.9 Summary and Further Reading -- 5.10 Principal Symbols -- 6 Speech Feature Enhancement -- 6.1 Noise and Reverberation in Various Domains -- 6.2 Two Principal Approaches -- 6.3 Direct Speech Feature Enhancement -- 6.4 Schematics of Indirect Speech Feature Enhancement -- 6.5 Estimating Additive Distortion -- 6.6 Estimating Convolutional Distortion -- 6.7 Distortion Evolution -- 6.8 Distortion Evaluation -- 6.9 Distortion Compensation -- 6.10 Joint Estimation of Additive and Convolutional Distortions.</p> <p>6.11 Observation Uncertainty -- 6.12 Summary and Further Reading -- 6.13 Principal Symbols -- 7 Search: Finding the Best Word Hypothesis -- 7.1 Fundamentals of Search -- 7.2 Weighted Finite-State Transducers -- 7.3 Knowledge Sources -- 7.4 Fast On-the-Fly Composition -- 7.5 Word and Lattice Combination -- 7.6 Summary and Further Reading -- 7.7 Principal Symbols -- 8 Hidden Markov Model Parameter Estimation -- 8.1 Maximum Likelihood Parameter Estimation -- 8.2 Discriminative Parameter Estimation -- 8.3 Summary and Further Reading -- 8.4 Principal Symbols -- 9 Feature and Model Transformation -- 9.1 Feature Transformation Techniques -- 9.2 Model Transformation Techniques -- 9.3 Acoustic Model Combination -- 9.4 Summary and Further Reading -- 9.5 Principal Symbols -- 10 Speaker Localization and Tracking -- 10.1 Conventional Techniques -- 10.2 Speaker Tracking with the Kalman Filter -- 10.3 Tracking Multiple Simultaneous Speakers -- 10.4 Audio-Visual Speaker Tracking -- 10.5 Speaker Tracking with the Particle Filter -- 10.6 Summary and Further Reading -- 10.7 Principal Symbols -- 11 Digital Filter Banks -- 11.1 Uniform Discrete Fourier Transform Filter Banks -- 11.2 Polyphase Implementation -- 11.3 Decimation and Expansion -- 11.4 Noble Identities -- 11.5 Nyquist(M) Filters -- 11.6 Filter Bank Design of De Haan et al -- 11.7 Filter Bank Design with the Nyquist(M) Criterion -- 11.8 Quality Assessment of Filter Bank Prototypes -- 11.9 Summary and Further Reading -- 11.10 Principal Symbols -- 12 Blind Source Separation -- 12.1 Channel Quality and Selection -- 12.2 Independent Component Analysis -- 12.3 BSS Algorithms based on Second-Order</p>

Statistics -- 12.4 Summary and Further Reading -- 12.5 Principal Symbols -- 13 Beamforming -- 13.1 Beamforming Fundamentals -- 13.2 Beamforming Performance Measures -- 13.3 Conventional Beamforming Algorithms -- 13.4 Recursive Algorithms -- 13.5 Nonconventional Beamforming Algorithms -- 13.6 Array Shape Calibration -- 13.7 Summary and Further Reading. 13.8 Principal Symbols -- 14 Hands On -- 14.1 Example Room Configurations -- 14.2 Automatic Speech Recognition Engines -- 14.3 Word Error Rate -- 14.4 Single-Channel Feature Enhancement Experiments -- 14.5 Acoustic Speaker-Tracking Experiments -- 14.6 Audio-Video Speaker-Tracking Experiments -- 14.7 Speaker-Tracking Performance vs Word Error Rate -- 14.8 Single-Speaker Beamforming Experiments -- 14.9 Speech Separation Experiments -- 14.10 Filter Bank Experiments -- 14.11 Summary and Further Reading -- Appendices -- A List of Abbreviations -- B Useful Background -- B.1 Discrete Cosine Transform -- B.2 Matrix Inversion Lemma -- B.3 Cholesky Decomposition -- B.4 Distance Measures -- B.5 Super-Gaussian Probability Density Functions -- B.6 Entropy -- B.7 Relative Entropy -- B.8 Transformation Law of Probabilities -- B.9 Cascade of Warping Stages -- B.10 Taylor Series -- B.11 Correlation and Covariance -- B.12 Bessel Functions -- B.13 Proof of the Nyquist / Shannon Sampling Theorem -- B.14 Proof of Equations (11.31 / 11.32) -- B.15 Givens Rotations -- B.16 Derivatives with Respect to Complex Vectors -- B.17 Perpendicular Projection Operators -- Bibliography -- Index.

Sommario/riassunto

A complete overview of distant automatic speech recognition The performance of conventional Automatic Speech Recognition (ASR) systems degrades dramatically as soon as the microphone is moved away from the mouth of the speaker. This is due to a broad variety of effects such as background noise, overlapping speech from other speakers, and reverberation. While traditional ASR systems underperform for speech captured with far-field sensors, there are a number of novel techniques within the recognition system as well as techniques developed in other areas of signal processing that can mitigate the deleterious effects of noise and reverberation, as well as separating speech from overlapping speakers. Distant Speech Recognition presents a contemporary and comprehensive description of both theoretic abstraction and practical issues inherent in the distant ASR problem. Key Features: *Covers the entire topic of distant ASR and offers practical solutions to overcome the problems related to it *Provides documentation and sample scripts to enable readers to construct state-of-the-art distant speech recognition systems *Gives relevant background information in acoustics and filter techniques, *Explains the extraction and enhancement of classification relevant speech features *Describes maximum likelihood as well as discriminative parameter estimation, and maximum likelihood normalization techniques *Discusses the use of multi-microphone configurations for speaker tracking and channel combination *Presents several applications of the methods and technologies described in this book *Accompanying website with open source software and tools to construct state-of-the-art distant speech recognition systems This reference will be an invaluable resource for researchers, developers, engineers and other professionals, as well as advanced students in speech technology, signal processing, acoustics, statistics and artificial intelligence fields.