

1.	Record Nr.	UNINA990010027670403321
	Autore	Mann, Michael <1942- >
	Titolo	The sources of social power / Michel Mann
	Pubbl/distr/stampa	Cambridge : Cambridge University Press, 2012
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	Lingua di pubblicazione	Inglese
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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
	Pubbl/distr/stampa	Paris : Hermann, 1932
	Collana	Actualités scientifiques et industrielles
	Locazione	MA1
	Collocazione	12-I-3
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3.	Record Nr.	UNISALENTO991001911589707536
	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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	Altri autori (Persone)	Marchetti, Italianoauthor
	Soggetti	Giambullari, Bernardo Lorenzo : <de' Medici>
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4.	Record Nr.	UNINA9910808716103321
	Autore	Wolfel Matthias
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Note generali	Description based upon print version of record.
Nota di bibliografia	Includes bibliographical references and index.
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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.