# Microphone Arrays Signal Processing Techniques And Applications Digital Signal Processing By Michael Brandstein Editor Darren Ward Editor 2 May 2001 Hardcover

When Speech and Audio Signal Processing published in 1999,it stood out from its competition in its breadth of coverage andits accessible, intutiont-based style. This book was aimed atindividual students and engineers excited about the broad span ofaudio processing and curious to understand the availabletechniques. Since then, with the advent of the iPod in 2001,the field of digital audio and music has exploded, leading to amuch greater interest in the technical aspects of audioprocessing. This Second Edition will update and revise the originalbook to augment it with new material describing both the enablingtechnologies of digital music distribution (most significantly theMP3) and a range of exciting new research areas in automatic musiccontent processing (such as automatic transcription, musicsimilarity, etc.) that have emerged in the past five years, drivenby the digital music revolution. New chapter topics include: Psychoacoustic Audio Coding, describing MP3 and relatedaudio coding schemes based on psychoacoustic masking ofquantization noise Music Transcription, including automatically derivingnotes, beats, and chords from music signals. Music Information Retrieval, primarily focusing onaudio-based genre classification, artist/style identification, andsimilarity estimation. Audio Source Separation, including multi-microphonebeamforming, blind source separation, and the perception-inspiredtechniques usually referred to as Computational Auditory SceneAnalysis (CASA).

Single-channel signal enhancement in the time domain -- Single-channel signal enhancement in the frequency domain -- Multichannel signal enhancement in the time domain -- Multichannel signal enhancement in the frequency domain -- An exhaustive class of linear filters -- Fixed beamforming -- Adaptive beamforming -- Differential beamforming -- Beampattern design -- Beamforming in the time domain

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 Recognitionpresents 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.

A comprehensive guide that addresses the theory and practice of spatial audio This book provides readers with the principles and best practices in spatial audio signal processing. It describes how sound fields and their perceptual attributes are captured and analyzed within the time-frequency domain, how essential representation parameters are coded, and how such signals are efficiently reproduced for practical applications. The book is split into four parts starting with an overview of the fundamentals. It then goes on to explain the reproduction of spatial sound before offering an examination of signal-dependent spatial filtering. The book finishes with coverage of both current and future applications and the direction that spatial audio research is heading in. Parametric Time-frequency Domain Spatial Audio focuses on applications in entertainment audio, including music, home cinema, and gaming—covering the capturing and reproduction of spatial sound as well as its generation, transduction, representation, transmission, and perception. This book will teach readers the tools needed for such processing, and provides an overview to existing research. It also shows recent up-to-date projects and commercial applications built on top of the systems. Provides an in-depth presentation of the principles, past developments, state-of-the-art methods, and future research directions of spatial audio technologies Includes contributions from leading researchers in the field Offers MATLAB codes with selected chapters An advanced book aimed at readers who are capable of digesting mathematical expressions about digital signal processing and sound field analysis, Parametric Time-frequency Domain Spatial Audio is best suited for researchers in academia and in the audio industry.

A handbook on recent advancements and the state of the art in array processing and sensor Networks Handbook on Array Processing and Sensor Networks provides readers with a collection of tutorial articles contributed by world-renowned experts on recent advancements and the state of the art in array processing and sensor networks. Focusing on fundamental principles as well as applications, the handbook provides exhaustive coverage of: wavelets; spatial spectrum estimation; MIMO radio propagation; robustness issues in sensor array processing; wireless communications and sensing in multi-path environments using multi-antenna transceivers; implicit training and array processing for digital communications systems; unitary design of radar waveform diversity sets; acoustic array processing for speech enhancement; acoustic beamforming for hearing aid applications; undetermined blind source separation using acoustic arrays; array processing in astronomy; digital 3D/4D ultrasound imaging technology; self-localization of sensor networks; multi-target tracking and classification in collaborative sensor networks via sequential Monte Carlo; energy-efficient decentralized estimation; sensor data fusion with application to multi-target tracking; distributed algorithms in sensor networks; cooperative communications; distributed source coding; network coding for sensor networks; information-theoretic studies of wireless networks; distributed adaptive learning mechanisms; routing for statistical inference in sensor networks; spectrum estimation in cognitive radios; nonparametric techniques for pedestrian tracking in wireless local area networks; signal processing and networking via the theory of global games; biochemical transport modeling, estimation, and detection in realistic environments; and security and privacy for sensor networks. Handbook on Array Processing and Sensor Networks is the first book of its kind and will appeal to researchers, professors, and graduate students in array processing, sensor networks, advanced signal processing, and networking.

This book constitutes of the major results of the EU COST (European Cooperation in the field of Scientific and Technical Research) Action 277: NSP, Nonlinear Speech Processing, running from April 2001 to June 2005. Coverage includes such areas as speech analysis for speech synthesis, speech recognition, speech-non speech discrimination and voice quality assessment, speech enhancement, and emotional state detection.

The volume Automation Control Theory Perspectives in Intelligent Systems presents new approaches and methods to real-world problems, and in particular, exploratory research that describes novel approaches in the field of cybernetics and automation control theory. Particular emphasis is laid on modern trends in intelligent information technology, system monitoring and proactive management of complex objects

The 5th Computer Science On-line Conference (CSOC2016) is intended to provide an international forum for discussions on the latest high-quality research results in all areas related to Computer Science. The addressed topics are the theoretical aspects and applications of Computer Science, Artificial Intelligences, Cybernetics, Automation Control Theory and Software Engineering.

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This book deals with the problem of detecting and localizing multiple simultaneously active wideband acoustic sources by applying the notion of wavefield decomposition using circular and spherical microphone arrays. A rigorous derivation of modal array signal processing algorithms for unambiguous source detection and localization, as well as performance evaluations by means of measurements using an actual real-time capable implementation, are discussed.

Modern communication devices, such as mobile phones, teleconferencing systems, VoIP, etc., are often used in noisy and reverberant environments. Therefore, signals picked up by the microphones from telecommunication devices contain not only the desired near-end speech signal, but also interferences such as the background noise, far-end echoes produced by the loudspeaker, and reverberations of the desired source. These interferences degrade the fidelity and intelligibility of the near-end speech in human-to-human telecommunications and decrease the performance of human-to-machine interfaces (i.e., automatic speech recognition systems). The proposed book deals with the fundamental challenges of speech processing in modern communication, including speech enhancement, interference suppression, acoustic echo cancellation, relative transfer function identification, source localization, dereverberation, and beamforming in reverberant environments. Enhancement of speech signals is necessary whenever the source signal is corrupted by noise. In highly non-stationary noise environments, noise transients, and interferences may be extremely annoying. Acoustic echo cancellation is used to eliminate the acoustic coupling between the loudspeaker and the microphone of a communication device. Identification of the relative transfer function between sensors in response to a desired speech signal enables to derive a reference noise signal for suppressing directional or coherent noise sources. Source localization, dereverberation, and beamforming in reverberant environments further enable to increase the intelligibility of the near-end speech signal.

This book presents the signal processing algorithms that have been developed to process the signals acquired by a spherical microphone array. Spherical microphone arrays can be used to capture the sound field in three dimensions and have received significant interest from researchers and audio engineers. Algorithms for spherical array processing are different to corresponding algorithms already known in the literature of linear and planar arrays because the spherical geometry can be exploited to great beneficial effect. The authors aim to advance the field of spherical array processing by helping those new to the field to study it efficiently and from a single source, as well as by offering a way for more experienced researchers and engineers to consolidate their understanding, adding either or both of breadth and depth. The level of the presentation corresponds to graduate studies at MSc and PhD level. This book begins with a presentation of some of the essential mathematical and physical theory relevant to spherical microphone arrays, and of an acoustic impulse response simulation method, which can be used to comprehensively evaluate spherical array processing algorithms in reverberant environments. The chapter on acoustic parameter estimation describes the way in which useful descriptions of acoustic scenes can be parameterized, and the signal processing algorithms that can be used to estimate the parameter values using spherical microphone arrays. Subsequent chapters exploit these parameters including in particular measures of direction-of-arrival and of diffuseness of a sound field. The array processing algorithms are then classified into two main classes, each described in a separate chapter. These are signal-dependent and signal-independent beamforming algorithms. Although signal-dependent beamforming algorithms are in theory able to provide better performance compared to the signal-independent algorithms, they are currently rarely used in practice. The main reason for this is that the statistical information required by these algorithms is difficult to estimate. In a subsequent chapter it is shown how the estimated acoustic parameters can be used in the design of signal-dependent beamforming algorithms. This final step closes, at least in part, the gap between theory and practice.

This book treats important topics in "Acoustic Echo and Noise Control" and reports the latest developments. Methods for enhancing the quality of transmitted speech signals are gaining growing attention in universities and in industrial development laboratories. This book, written by an international team of highly qualified experts, concentrates on the modern and advanced methods.

In the past few years we have written and edited several books in the area of acousticandspeechsignalprocessing. Thereasonbehindthisendeavoristhat there were almost no books available in the literature when we ?rst started while there was (and still is) a real need to publish manuscripts summarizing the most useful ideas, concepts, results, and state-of-the-art algorithms in this important area of research. According to all the feedback we have received so far, we can say that we were right in doing this. Recently, several other researchers have followed us in this journey and have published interesting books with their own visions and perspectives. The idea of writing a book on Microphone Array Signal Processing comes from discussions we have had with many colleagues and friends. As a c- sequence of these discussions, we came up with the conclusion that, again, there is an urgent need for a monograph that carefully explains the theory and implementation of microphone arrays. While there are many manuscripts on antenna arrays from a narrowband perspective (narrowband signals and narrowband processing), the literature is quite scarce when it comes to s- sor arrays explained from a truly broadband perspective. Many algorithms for speech applications were simply borrowed from narrowband antenna - rays. However, a direct application of narrowband ideas to broadband speech processing may not be necessarily appropriate and can lead to many m- understandings.

This book provides a comprehensive introduction to the theory and practice of spherical microphone arrays. It is written for graduate students, researchers and engineers who work with spherical microphone arrays in a wide range of applications. The first two chapters provide the reader with the necessary mathematical and physical background, including an introduction to the spherical Fourier transform and the formulation of plane-wave sound fields in the spherical harmonic domain. The third chapter covers the theory of spatial sampling, employed when selecting the positions of microphones to sample sound pressure functions in space. Subsequent chapters present various spherical array configurations, including the popular rigid-sphere-based configuration. Beamforming (spatial filtering) in the spherical harmonics domain, including axis-symmetric beamforming, and the performance measures of directivity index and white noise gain are introduced, and a range of optimal beamformers for spherical arrays, including beamformers that achieve maximum directivity and maximum robustness, and the Dolph-Chebyshev beamformer are developed. The final chapter discusses more advanced beamformers, such as MVDR and LCMV, which are tailored to the measured sound field.

"Monitoring of endangered species is time intensive and expensive. Often, only a few individuals can be followed at any one time due to the expense of transmitters and the time taken for telemetry or direct observation. In addition, some individuals may not be monitored due to their remote location. However, the use of microphone arrays allows the position of a bird to be determined by comparing the time of arrival of its calls at each microphone in the array. The use of microphone arrays allows to automate the process of remote monitoring, thus reducing the cost of management. Although microphone arrays have been developed in the past, little emphasis has been placed on back-end signal processing systems necessary to locate calls within the recording stream, or to identify the species and individuals that produced the calls. The aim of this research is to: 1) develop a signal recognition system, based on Hidden Markov Models, to automatically detect the calls of kokako (Callaeas cinerea) as they are recorded by an experimental microphone array, 2) develop a recognition system, based on human speech recognition algorithms, to identify the dialect and sex of calling kokako as well as the individual identity of calling birds, and 3) test the detection and identification systems, in combination with the array, by mapping the territories of male and female kokako in the Waitakere Ranges. Two microphone arrays were used to record recently released kokako in the Waitakere Ranges. Later, the Hidden Markov Model detector was used to find and extract kokako calls from the background forest noise. A neural network classifier was developed to classify kokako calls to individual, sex and dialect. The Hidden Markov Model based detector was able to detect and isolate from the forest noise 85% of calls produced by the kokako and only 15 non-kokako calls detected. Using Mel-scale cepstral coefficients and artificial neural networks we were able to classify kokako calls to individuals with up to 97% accuracy. Kokako dialects were classified with 96% accuracy and the sex of birds was classified with 92% accuracy. The position of birds, estimated by the microphone array was a good match compared with the recent results of a kokako survey. The work presented here represents a useful tool for the monitoring of birds. Using the system developed here it is possible not only passively monitor kokako in their natural habitat but also track the distribution of individuals and interactions between birds of different sexes and dialects. Moreover, this system could later be retrained and used to monitor and track almost any other sound producing animal"--Abstract.

It gives me immense pleasure to introduce this timely handbook to the research/- velopment communities in the ?eld of signal processing systems (SPS). This is the ?rst of its kind and represents state-of-the-arts coverage of research in this ?eld. The driving force behind information technologies (IT) hinges critically upon the major advances in both component integration and system integration. The major breakthrough for the former is undoubtedly the invention of IC in the 50's by Jack S. Kilby, the Nobel Prize Laureate in Physics 2000. In an integrated circuit, all components were made of the same semiconductor material. Beginning with the pocket calculator in 1964, there have been many increasingly complex applications followed. In fact, processing gates and memory storage on a chip have since then grown at an exponential rate, following Moore's Law. (Moore himself admitted that Moore's Law had turned out to be more accurate, longer lasting and deeper in impact than he ever imagined. ) With greater device integration, various signal processing systems have been realized for many killer IT applications. Further breakthroughs in computer sciences and Internet technologies have also catalyzed large-scale system integration. All these have led to today's IT revolution which has profound impacts on our lifestyle and overall prospect of humanity. (It is hard to imagine life today without mobiles or Internets!) The success of SPS requires a well-concerted integrated approach from mul- ple disciplines, such as device, design, and application.

The Handbook of Signal Processing in Acoustics brings together a wide range of perspectives from over 100 authors to reveal the interdisciplinary nature of the subject. It brings the key issues from both acoustics and signal processing into perspective and is a unique resource for experts and practitioners alike to find new ideas and techniques within the diversity of signal processing in acoustics.

This is the first book to provide a single complete reference on microphone arrays. Top researchers in this field contributed articles documenting the current state of the art in microphone array research, development and technological application.

This handbook plays a fundamental role in sustainable progress in speech research and development. With an accessible format and with accompanying DVD-Rom, it targets three categories of readers: graduate students, professors and active researchers in academia, and engineers in industry who need to understand or implement some specific algorithms for their speech-related products. It is a superb source of application-oriented, authoritative and comprehensive information about these technologies, this work combines the established knowledge derived from research in such fast evolving disciplines as Signal Processing and Communications, Acoustics, Computer Science and Linguistics.

This third volume, edited and authored by world leading experts, gives a review of the principles, methods and techniques of important and emerging research topics and technologies in array and statistical signal processing. With this reference source you will: Quickly grasp a new area of research Understand the underlying principles of a topic and its application Ascertain how a topic relates to other areas and learn of the research issues yet to be resolved Quick tutorial reviews of important and emerging topics of research in array and statistical signal processing Presents core principles and shows their application Reference content on core principles, technologies, algorithms and applications Comprehensive references to journal articles and other literature on which to build further, more specific and detailed knowledge Edited by leading people in the field who, through their reputation, have been able to commission experts to write on a particular topic

Users of signal processing systems are never satis?ed with the system they currently use. They are constantly asking for higher quality, faster perf- mance, more comfort and lower prices. Researchers and developers should be appreciative for this attitude. It justi?es their constant e?ort for improved systems. Better knowledge about biological and physical interrelations c- ing along with more powerful technologies are their engines on the endless road to perfect systems. This book is an impressive image of this process. After "Acoustic Echo 1 and Noise Control" published in 2004 many new results lead to "Topics in 2 Acoustic Echo and Noise Control" edited in 2006 . Today

– in 2008 – even morenew?ndingsandsystemscouldbecollectedinthisbook.Comparingthe contributions in both edited volumes progress in knowledge and technology becomesclearlyvisible:Blindmethodsandmultiinputsystemsreplace“h- ble” low complexity systems. The functionality of new systems is less and less limited by the processing power available under economic constraints. The editors have to thank all the authors for their contributions. They cooperated readily in our e?ort to unify the layout of the chapters, the ter- nology, and the symbols used. It was a pleasure to work with all of them. Furthermore, it is the editors concern to thank Christoph Baumann and the Springer Publishing Company for the encouragement and help in publi- ing this book.

This book addresses the problem of separating spontaneous multi-party speech by way of microphone arrays (beamformers) and adaptive signal processing techniques. It is written is a concise manner and an effort has been made such that all presented algorithms can be straightforwardly implemented by the reader. All experimental results have been obtained with real in-car microphone recordings involving simultaneous speech of the driver and the co-driver.

Recently, we proposed a completely novel and efficient way to design differential beamforming algorithms for linear microphone arrays. Thanks to this very flexible approach, any order of differential arrays can be designed. Moreover, they can be made robust against white noise amplification, which is the main inconvenience in these types of arrays. The other well-known problem with linear arrays is that electronic steering is not really feasible. In this book, we extend all these fundamental ideas to circular microphone arrays and show that we can design small and compact differential arrays of any order that can be electronically steered in many different directions and offer a good degree of control of the white noise amplification problem, high directional gain, and frequency-independent response. We also present a number of practical examples, demonstrating that differential beamforming with circular microphone arrays is likely one of the best candidates for applications involving speech enhancement (i.e., noise reduction and dereverberation). Nearly all of the material presented is new and will be of great interest to engineers, students, and researchers working with microphone arrays and their applications in all types of telecommunications, security and surveillance contexts.

Provides state-of-the-art algorithms for sound capture, processing and enhancement Sound Capture and Processing: Practical Approaches covers the digital signal processing algorithms and devices for capturing sounds, mostly human speech. It explores the devices and technologies used to capture, enhance and process sound for the needs of communication and speech recognition in modern computers and communication devices. This book gives a comprehensive introduction to basic acoustics and microphones, with coverage of algorithms for noise reduction, acoustic echo cancellation, dereverberation and microphone arrays; charting the progress of such technologies from their evolution to present day standard. Sound Capture and Processing: Practical Approaches Brings together the state-of-the-art algorithms for sound capture, processing and enhancement in one easily accessible volume Provides invaluable implementation techniques required to process algorithms for real life applications and devices Covers a number of advanced sound processing techniques, such as multichannel acoustic echo cancellation, dereverberation and source separation Generously illustrated with figures and charts to demonstrate how sound capture and audio processing systems work An accompanying website containing Matlab code to illustrate the algorithms This invaluable guide will provide audio, R&D and software engineers in the industry of building systems or computer peripherals for speech enhancement with a comprehensive overview of the technologies, devices and algorithms required for modern computers and communication devices. Graduate students studying electrical engineering and computer science, and researchers in multimedia, cell-phones, interactive systems and acousticians will also benefit from this book.

The focus of this book is on array processing and beamforming with Kronecker products. It considers a large family of sensor arrays that allow the steering vector to be decomposed as a Kronecker product of two steering vectors of smaller virtual arrays. Instead of directly designing a global beamformer for the original array, once the steering vector has been decomposed, smaller virtual beamformers are designed and separately optimized for each virtual array. This means the matrices that need to be inverted are smaller, which increases the robustness of the beamformers, and reduces the size of the observations. The book explains how to perform beamforming with Kronecker product filters using an unconventional approach. It shows how the Kronecker product formulation can be used to derive fixed, adaptive, and differential beamformers with remarkable flexibility. Furthermore, it demonstrates how fixed and adaptive beamformers can be intelligently combined, optimally exploiting the advantages of both. The problem of spatiotemporal signal enhancement is also addressed, and readers will learn how to perform Kronecker product filtering in this context.

A strong reference on the problem of signal and speech enhancement, describing the newest developments in this exciting field. The general emphasis is on noise reduction, because of the large number of applications that can benefit from this technology.

Presents a unified framework of far-field and near-field array techniques for noise source identification and sound field visualization, from theory to application. Acoustic Array Systems: Theory, Implementation, and Application provides an overview of microphone array technology with applications in noise source identification and sound field visualization. In the comprehensive treatment of microphone arrays, the topics covered include an introduction to the theory, far-field and near-field array signal processing algorithms, practical implementations, and common applications: vehicles, computing and communications equipment, compressors, fans, and household appliances, and hands-free speech. The author concludes with other emerging techniques and innovative algorithms. Encompasses theoretical background, implementation considerations and application know-how Shows how to tackle broader problems in signal processing, control, and transudcers Covers both farfield and nearfield techniques in a balanced way Introduces innovative algorithms including equivalent source imaging (NESI) and high-resolution nearfield arrays Selected code examples available for download for readers to practice on their own Presentation slides available for instructor use A valuable resource for Postgraduates and researchers in acoustics, noise control engineering, audio engineering, and signal processing.

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Microphone arrays have attracted a lot of interest over the last few decades since they have the potential to solve many important problems such as noise reduction/speech enhancement, source separation, dereverberation, spatial sound recording, and source localization/tracking, to name a few. However, the design and implementation of microphone arrays with beamforming algorithms is not a trivial task when it comes to processing broadband signals such as speech. Indeed, in most sensor arrangements, the beamformer output tends to have a frequency-dependent response. One exception, perhaps, is the family of differential microphone arrays (DMAs) who have the promise to form frequency-independent responses. Moreover, they have the potential to attain high directional gains with small and compact apertures. As a result, this type of microphone arrays has drawn much research and development attention recently. This book is intended to provide a systematic study of DMAs from a signal processing perspective. The primary objective is to develop a rigorous but yet simple theory for the design, implementation, and performance analysis of DMAs. The theory includes some signal processing techniques for the design of commonly used first-order, second-order, third-order, and also the general Nth-order DMAs. For each order, particular examples are given on how to form standard directional patterns such as the dipole, cardioid, supercardioid, hypercardioid, subcardioid, and quadrupole. The study demonstrates the performance of the different order DMAs in terms of beampattern, directivity factor, white noise gain, and gain for point sources. The inherent relationship between differential processing and adaptive beamforming is discussed, which provides a better understanding of DMAs and why they can achieve high directional gain. Finally, we show how to design DMAs that can be robust against white noise amplification.

This book studies the link between differential beamforming and differential equations which in turn enables the study of fundamental theory and methods of beamforming from a different perspective, leading to new insights into the problem and new methods to solve the problem. The book first presents a brief overview of the problems and methods for beamforming and some performance measures popularly used either to evaluate beamformers or to derive optimal beamformers. Then, first-order, second-order, and general high-order linear difference equations are discussed, based on which the authors show how to formulate the beamforming problem and derive different beamforming methods, including fixed and adaptive ones. Furthermore, the authors show how to apply the theory of difference equations to the general problem of speech enhancement, and deduce a number of noise reduction filters, including the maximum SNR filter, the Wiener filter, the MVDR filter, etc. Also covered in the book are the difference equations and differential beamforming from the spectral graph perspective. Presents basic concepts, fundamental principles, and methods for beamforming from the perspective of linear difference equations; Provides formulation and methods of conventional beamforming, and first-order, second-order, and general high-order linear difference equations for beamforming; Includes the applications of linear difference equations to the problem of noise reduction; Explains beamforming based on difference equations with graphs.

Learn the technology behind hearing aids, Siri, and Echo Audio source separation and speech enhancement aim to extract one or more source signals of interest from an audio recording involving several sound sources. These technologies are among the most studied in audio signal processing today and bear a critical role in the success of hearing aids, hands-free phones, voice command and other noise-robust audio analysis systems, and music post-production software. Research on this topic has followed three convergent paths, starting with sensor array processing, computational auditory scene analysis, and machine learning based approaches such as independent component analysis, respectively. This book is the first one to provide a comprehensive overview by presenting the common foundations and the differences between these techniques in a unified setting. Key features: Consolidated perspective on audio source separation and speech enhancement. Both historical perspective and latest advances in the field, e.g. deep neural networks. Diverse disciplines: array processing, machine learning, and statistical signal processing. Covers the most important techniques for both single-channel and multichannel processing. This book provides both introductory and advanced material suitable for people with basic knowledge of signal processing and machine learning. Thanks to its comprehensiveness, it will help students select a promising research track, researchers leverage the acquired cross-domain knowledge to design improved techniques, and engineers and developers choose the right technology for their target application scenario. It will also be useful for practitioners from other fields (e.g., acoustics, multimedia, phonetics, and musicology) willing to exploit audio source separation or speech enhancement as pre-processing tools for their own needs.

The book covers the design formulations for broadband beamformer targeting nearfield and farfield sources. The book content includes background information on the acoustic environment, including propagation medium, the array geometries, signal models and basic beamformer designs. Subsequently it introduces design formulation for nearfield, farfield and mixed nearfield-farfield beamformers and extends the design formulation into electronically steerable beamformers. In addition, a robust formulation is introduced for all the designs mentioned.

For the first time, a reference on the most relevant applications of adaptive filtering techniques. Top researchers in the field contributed chapters addressing applications in acoustics, speech, wireless and networking, where research is still very active and open.

Audio Signal Processing for Next-Generation Multimedia Communication Systems presents cutting-edge digital signal processing theory and implementation techniques for problems including speech acquisition and enhancement using microphone arrays, new adaptive filtering algorithms, multichannel acoustic echo cancellation, sound source tracking and separation, audio coding, and realistic sound stage reproduction. This book's focus is almost exclusively on the processing, transmission, and presentation of audio and acoustic signals in multimedia communications for telecollaboration where immersive acoustics will play a great role in the near future.

This is the first book on the market to bring together material on array signal processing in a coherent fashion, with uniform notation and convention of models. KEY TOPICS: Using extensive examples and problems, it presents not only the theories of propagating waves and conventional array processing algorithms, but also the underlying ideas of adaptive array processing and multi-array tracking algorithms. MARKET: This manual will be valuable to engineers who wish to practice and advance their careers in the array signal processing field.

This book covers the state-of-the-art in deep neural-network-based methods for noise robustness in distant speech recognition applications. It provides insights and detailed descriptions of some of the new concepts and key technologies in the field, including novel architectures for speech enhancement, microphone arrays, robust features, acoustic model adaptation, training data augmentation, and training criteria. The contributed chapters also include descriptions of real-world applications, benchmark tools and datasets widely used in the field. This book is intended for researchers and practitioners working in the field of speech processing and recognition who are interested in the latest deep learning techniques for noise robustness. It will also be of interest to graduate students in electrical engineering or computer science, who will find it a useful guide to this field of research.

This book presents research and applications on arrival estimation and localization in speech processing to ensure that the broad vision of the direction of arrival estimation (DOAE) / localization of speech sources is well-established. The book first provides a brief overview of the most classical direction of arrival estimation and localization techniques. It then introduces the concept and model of acoustics sources and then highlights the most contemporary studies on this pervasive problem. In addition, the authors explore employing the optimization algorithms to improve the DOAE techniques. The book then highlights the concept and principles of the multi-DOAE approaches. Using a microphone array, the book introduces the localization and tracking problem of multiple speech/acoustic sources. It includes several applications and real-life speech sources localization based on the DOAE approaches. The book reports the challenges facing the DOAE techniques in speech-sources localization. The book pertains to researchers, designers, and engineers in speech processing fields.

This is the first book to describe how Autonomous Virtual Humans and Social Robots can interact with real people, be aware of the environment around them, and react to various situations. Researchers from around the world present the main techniques for tracking and analysing humans and their behaviour and contemplate the potential for these virtual humans and robots to replace or stand in for their human counterparts, tackling areas such as awareness and reactions to real world stimuli and using the same modalities as humans do: verbal and body gestures, facial expressions and gaze to aid seamless human-computer interaction (HCI). The research presented in this volume is split into three sections: ·User Understanding through Multisensory Perception: deals with the analysis and recognition of a given situation or stimuli, addressing issues of facial recognition, body gestures and sound localization. ·Facial and Body Modelling Animation: presents the methods used in modelling and animating faces and bodies to generate realistic motion. ·Modelling Human Behaviours: presents the behavioural aspects of virtual humans and social robots when interacting and reacting to real humans and each other. Context Aware Human-Robot and Human-Agent Interaction would be of great use to students, academics and industry specialists in areas like Robotics, HCI, and Computer Graphics.

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