Acoustic Signal Processing In Passive Sonar System With

The NATO Advanced Research Workshop on Issues in Acoustic Signal/Image Processing and Recognition was held August 5-9, 1982 at the Cappuccini complex in San Miniato Italy. The Workshop was primarily concerned with the underwater acoustic signal processing and seismic signal analysis and a major effort was made to link these topics with pattern recognition, i~ge processing and artificial intelligence. Major issues and new approaches in these interrelated areas were closely examined in the Workshop. In addition to paper presentations three discussion sessions were held on! (1) spectral analysis in underwater acoustics, (2) seismic wave propaga tion, seismic imaging and migration, and seismic inversion, and (3) unresolved issues and future directions. This Proceedings volume includes most presentations made at the Workshop. The publication, like the meeting itself, is unique in the sense that it provides exten sive interactions among the closely related areas stated above. Such interactions which usually result in the integration of different systems or approaches are certainly much needed to achieve some performance breakthrough while individual systems or approaches reach their performance limit. I am grateful to all participants for their active participation that makes the Workshop very productive, and to Dr. Lewis J. Lloyd and Dr. Ralph Goodman for their help to arrange an informative visit to the SACLANT ASW Research Centre for the Workshop participants. I am confident that this publication will be equally produc tive to report important current research results and near-future research activity particularly in underwater acoustic signal processing.

The comprehensive research activity around the World in the fields of Underwater Acoustics and Signal Processing being strongly supported by new experimental technique and equipment and by the parallel fast developments in computer technology and solid state devices, which has led to a rapidly reducing cost of digital processing thus enabling more complex processing to be carried out economically, emphasize how necessary it is at intervals of a few years through a NATO Advanced Study Institute (NATO ASI) and guided by leading experts to study the conquests in the fields of Underwater Acoustics and Signal Processing. This need of study is moreover stressed by the interdisciplina rity of Underwater Acoustics and Signal Processing, where a strong impact from other branches of science, - Geophysics, Radioastronomy, Bioengineering, Telecommunication, Seismology, Space Research etc. - is taking place, which makes it an extre mely difficult task for scientists to follow-up the development in all its phases and to preserve the general view of its rapid ly increasing number of possibilities. The present Proceedings of the NATO ASI held in Copenhagen during August 1980 join the series of proceedings of NATO summer schools on Underwater Acoustics and Signal Processing held during the past 20 years. The equality and the fusion of the individual research fields of Underwater Acoustics and Signal Processing and the separate introduction of advanced research results from other scientific areas related to underwater acoustics such as transducers characterize the subject matter of this NATO ASI.

The demand to explore the largest and also one of the richest parts of our planet, the advances in signal processing promoted by an exponential growth in computation power and a thorough study of sound propagation in the underwater realm, have lead to remarkable advances in sonar technology in the last years.The work on hand is a sum of knowledge of several authors who contributed in various aspects of sonar technology. This book intends to give a broad overview of the advances in sonar technology of the last years that resulted from the research effort of the authors in both sonar systems and their applications. It is intended for scientist and engineers from a variety of backgrounds and even those that never had contact with sonar technology before will find an easy introduction with the topics and principles exposed here.

This graduate-level text lays out the foundation of DSP for audio and the fundamentals of auditory perception, then goes on to discuss immersive audio rendering and synthesis, the digital equalization of room acoustics, and various DSP implementations. It covers a variety of topics and up-to-date results in immersive audio processing research: immersive audio synthesis and rendering, multichannel room equalization, audio selective signal cancellation, multirate signal processing for audio applications, surround sound processing, psychoacoustics and its incorporation in audio signal processing algorithms for solving various problems, and DSP implementations of audio processing algorithms on semiconductor devices.

This discussion of sonar signal processing bridges a number of related fields, including acoustic propagation in the medium, detection and estimation theory, filter theory, digital filtering, sensor array processing, spectral analysis, fast transforms and digital signal processing. The book begins with a discussion of the topics of analogue signalling conditioning, digital filtering, and the calculation of the discrete Fourier transform. Other topics discussed include analogue filters and analogue-to-digital conversion, finite impulse and infinite impulse response digital filters, and multirate processing techniques.

The book is an edited collection of research articles covering the current state of sonar systems, the signal processing methods and their applications prepared by experts in the field. The first section is dedicated to the theory and applications of innovative synthetic aperture, interferometric, multistatic sonars and modeling and simulation. Special section in the book is dedicated to sonar signal processing methods covering: passive sonar array beamforming, direction of arrival estimation, signal detection and classification using DEMON and LOFAR principles, adaptive matched field signal processing. The image processing techniques include: image denoising, detection and classification of artificial mine like objects and application of hidden Markov model and artificial neural networks for signal classification. The biology applications include the analysis of biosonar capabilities and underwater sound influence on human hearing. The marine science applications include fish species target strength modeling, identification and discrimination from bottom scattering and pelagic biomass neural network estimation methods. Marine geology has place in the book with geomorphological parameters estimation from side scan sonar images. The book will be interesting not only for specialists in the area but also for readers as a guide in sonar systems principles of operation, signal processing methods and marine applications.

This book contains the papers that were accepted for presentation at the 1988 NATO Advanced Study Institute on Underwater Acoustic Data Processing, held at the Royal Military College of Canada from 18 to 29 July, 1988. Approximately 110 participants from various NATO countries were in attendance during this two week period. Their research interests range from underwater acoustics to signal processing and computer science; some are renowned scientists and some are recent Ph.D. graduates. The purpose of the ASI was to provide an authoritative summing up of the various research activities related to sonar technology. The exposition on each subject began with one or two tutorials prepared by invited lecturers, followed by research papers which provided indications of the state of development in that specific area. I have broadly classified the papers into three sections under the titles of I. Propagation and Noise, II. Signal Processing and III. Post Processing. The reader will find in Section I papers on low frequency acoustic sources and effects of the medium on underwater acoustic propagation. Problems such as coherence loss due to boundary interaction, wavefront distortion and multipath transmission were addressed. Besides the medium, corrupting noise sources also have a strong influence on the performance of a sonar system and several researchers described methods of modeling these sources.

Passive acoustics, or the recording of pressure signals from uncontrolled sound sources, is a powerful tool for monitoring man-made and natural sounds in the ocean. Passive acoustics can be used to detect changes in physical processes within the environment, study behavior and movement of marine animals, or observe presence and motion of ocean vessels and vehicles. Advances in ocean instrumentation and data storage have improved the availability and quality of ambient noise recordings, but there is an ongoing effort to improve signal processing algorithms for extracting useful information from the ambient noise. This dissertation uses machine learning as a framework to address problems in underwater passive acoustic signal processing. Statistical learning has been used for decades, but machine learning has recently gained popularity due to the exponential growth of data and its ability to capitalize on these data with efficient GPU computation. The chapters within this dissertation cover two types of problems: characterization and classification of ambient noise, and localization of passive acoustic sources. First, ambient noise in the eastern Arctic was studied from April to September 2013 using a vertical hydrophone array as it drifted from near the North Pole to north of Fram Strait. Median power spectral estimates and empirical probability density functions (PDFs) along the array transit show a change in the ambient noise levels corresponding to seismic survey airgun occurrence and received level at low frequencies and transient ice noises at high frequencies. Noise contributors were manually identified and included broadband and tonal ice noises, bowhead whale calling, seismic airgun surveys, and earthquake T phases. The bowhead whale or whales detected were believed to belong to the endangered Spitsbergen population and were recorded when the array was as far north as 86°24'N. Then, ambient noise recorded in a Hawaiian coral reef was analyzed for classification of whale song and fish calls. Using automatically detected acoustic events, two clustering processes were proposed: clustering handpicked acoustic metrics using unsupervised methods, and deep embedded clustering (DEC) to learn latent features and clusters from fixed-length power spectrograms. When compared on simulated signals of fish calls and whale song, the unsupervised clustering methods were confounded by overlap in the handpicked features while DEC identified clusters with fish calls, whale song, and events with simultaneous fish calls and whale song. Both clustering approaches were applied to recordings from directional autonomous seafloor acoustic recorder (DASAR) sensors on a Hawaiian coral reef in February 2020. Next, source localization in ocean acoustics was posed as a machine learning problem in which data-driven methods learned source ranges or direction-of-arrival directly from observed acoustic data. The pressure received by a vertical linear array was preprocessed by constructing a normalized sample covariance matrix (SCM) and used as the input for three machine learning methods: feed-forward neural networks (FNN), support vector machines (SVM) and random forests (RF). The FNN, SVM, RF and conventional matched-field processing were applied to recordings from ships in the Noise09 experiment to demonstrate the potential of machine learning for underwater source localization. The source localization problem was extended by examining the relationship between conventional beamforming and linear supervised learning. Then, a nonlinear deep feedforward neural network (FNN) was developed for direction-of-arrival (DOA) estimation for two-source DOA and for K-source DOA, where K is unknown. With multiple snapshots, K-source FNN achieved resolution and accuracy similar to Multiple Signal Classification (MUSIC) and SBL for an unknown number of sources. The practicality of the deep FNN model was demonstrated on ships in the Swellex96 experimental data.

Passive acoustic monitoring is increasingly used by the scientific community to study, survey and census marine mammals, especially cetaceans, many of which are easier to hear than to see. PAM is also used to support efforts to mitigate potential negative effects of human activities such as ship traffic, military and civilian sonar and offshore exploration. Walter Zimmer provides an integrated approach to PAM, combining physical principles, discussion of technical tools and application-oriented concepts of operations. Additionally, relevant information and tools necessary to assess existing and future PAM systems are presented, with Matlab code used to generate figures and results so readers can reproduce data and modify code to analyse the impact of changes. This allows the principles to be studied whilst discovering potential difficulties and side effects. Aimed at graduate students and researchers, the book provides all information and tools necessary to gain a comprehensive understanding of this interdisciplinary subject.

Underwater Acoustic Signal ProcessingModeling, Detection, and EstimationSpringer

Applied Underwater Acoustics meets the needs of scientists and engineers working in underwater acoustics and graduate students solving problems in, and preparing theses on, topics in underwater acoustics. The book is structured to provide the basis for rapidly assimilating the essential underwater acoustic knowledge base for practical application to daily research and analysis. Each chapter of the book is self-supporting and focuses on a single topic and its relation to underwater acoustics. The chapters start with a brief description of the topic’s physical background, necessary definitions, and a short description of the applications, along with a roadmap to the chapter. The subtopics covered within individual subchapters include most frequently used equations that describe the topic. Equations are not derived, rather, assumptions behind equations and limitations on the applications of each equation are emphasized. Figures, tables, and illustrations related to the sub-topic are presented in an easy-to-use manner, and examples on the use of the equations, including appropriate figures and tables are also included. Provides a complete and up-to-date treatment of all major subjects of underwater acoustics Presents chapters written by recognized experts in their individual field Covers the fundamental knowledge scientists and engineers need to solve problems in underwater acoustics Illuminates, in shorter sub-chapters, the modern applications of underwater acoustics that are described in worked examples Demands no prior knowledge of underwater acoustics, and the physical principles and mathematics are designed to be readily understood by scientists, engineers, and graduate students of underwater acoustics Includes a comprehensive list of literature references for each chapter

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"This book provides a comprehensive approach of signal processing tools regarding the enhancement, recognition, and protection of speech and audio signals. It offers researchers and practitioners the information they need to develop and implement efficient signal processing algorithms in the enhancement field"--Provided by publisher.

Advances in digital signal processing algorithms and computer technology have combined to produce real-time systems with capabilities far beyond those of just few years ago. Nonlinear, adaptive methods for signal processing have emerged to provide better array gain performance, however, they lack the robustness of conventional algorithms. The challenge remains to develop a concept that exploits the advantages of both-a scheme that integrates these methods in practical, real-time systems. The Advanced Signal Processing Handbook helps you meet that challenge. Beyond offering an outstanding introduction to the principles and applications of advanced signal processing, it develops a generic processing structure that takes advantage of the similarities that exist among radar, sonar, and medical imaging systems and integrates conventional and nonlinear processing schemes.

FROM THE PREFACE: Many new useful ideas are presented in this handbook, including new finite impulse response (FIR) filter design techniques, half-band and multiplierless FIR filters, interpolated FIR (IFIR) structures, and error spectrum shaping.

Telecommunication systems and human-machine interfaces have begun using multiple microphones and loudspeakers to render interaction more lifelike, and more efficient. This raises acoustic signal processing problems under multiple-input multiple-output (MIMO) scenarios, encompassing distant speech acquisition, sound source localization and tracking, echo and noise control, source separation and speech dereverberation, and many others. The book opens with an acoustic MIMO paradigm, establishing fundamentals, and linking acoustic MIMO signal processing with classical signal processing and communication theories. The second part of the book presents a novel analysis of acoustic applications carried out in the paradigm to reinforce the fundamentals of acoustic MIMO signal processing.

This monograph presents a unified approach to model-based processing for underwater acoustic arrays. The use of physical models in passive array processing is not a new idea, but it has been used on a case-by-case basis, and as such, lacks any unifying structure. This work views all such processing methods as estimation procedures, which then can be unified by treating them all as a form of joint estimation based on a Kalman-type recursive processor, which can be recursive either in space or time, depending on the application. This is done for three reasons. First, the Kalman filter provides a natural framework for the inclusion of physical models in a processing scheme. Second, it allows poorly known model parameters to be jointly estimated along with the quantities of interest. This is important, since in certain areas of array processing already in use, such as those based on matched-field processing, the so-called mismatch problem either degrades performance or, indeed, prevents any solution at all. Thirdly, such a unification provides a formal means of quantifying the performance improvement. The term model-based will be strictly defined as the use of physics-based models as a means of introducing a priori information. This leads naturally to viewing the method as a Bayesian processor. Short expositions of estimation theory and acoustic array theory are presented, followed by a presentation of the Kalman filter in its recursive estimator form. Examples of applications to localization, bearing estimation, range estimation and model parameter estimation are provided along with experimental results verifying the method. The book is sufficiently self-contained to serve as a guide for the application of model-based array processing for the practicing engineer.

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.

A systematic and integrated account of signal and data processing with emphasis on the distinctive marks of the ocean environment is provided in this informative text. Underwater problems such as space-time processing relations vs. disjointed ones, processing of passive observations vs. active ones, time delay estimation vs. frequency estimation, channel effects vs. transparent ones, integrated study of signal, data, and channel processing vs. separate ones, are highlighted. The book provides the beginner with a concise presentation of the essential concepts, defines the basic computational steps, and gives the mature reader an advanced view of underwater systems and the relationships among their building blocks. It presents the needed topics on applied estimation theory within the underwater systems context. Included are topics in linear and nonlinear filtering, spectral analysis, generalized correlation, cepstrum and complex demodulation, Cramer-Rao Bounds, maximum likelihood, weighted least-squares, Kalman filtering, expert systems, wave propagation and their use, as well as their performance in applications to canonical ocean problems. The applications center on the definition, analysis, and solution implementations to representative underwater signal analysis problems dealing with signals estimation, their location and motion. The potential limitations and pitfalls of the implementations are delineated in homogeneous, noisy, interfering, inhomogeneous, multipath, distortions, and/or dispersive channels.

The statistical bootstrap is one of the methods that can be used to calculate estimates of a certain number of unknown parameters of a random process or a signal observed in noise, based on a random sample. Such situations are common in signal processing and the bootstrap is especially useful when only a small sample is available or an analytical analysis is too cumbersome or even impossible. This book covers the foundations of the bootstrap, its properties, its strengths and its limitations. The authors focus on bootstrap signal detection in Gaussian and non-Gaussian interference as well as bootstrap model selection. The theory developed in the book is supported by a number of useful practical examples written in MATLAB. The book is aimed at graduate students and engineers, and includes applications to real-world problems in areas such as radar and sonar, biomedical engineering and automotive engineering.

In 1993, the first edition of The Electrical Engineering Handbook set a new standard for breadth and depth of coverage in an engineering reference work. Now, this classic has been substantially revised and updated to include the latest information on all the important topics in electrical engineering today. Every electrical engineer should have an opportunity to expand his expertise with this definitive guide. In a single volume, this handbook provides a complete reference to answer the questions encountered by practicing engineers in industry, government, or academia. This well-organized book is divided into 12 major sections that encompass the entire field of electrical engineering, including circuits, signal processing, electronics, electromagnetics, electrical effects and devices, and energy, and the emerging trends in the fields of communications, digital devices, computer engineering, systems, and biomedical engineering. A compendium of physical, chemical, material, and mathematical data completes this comprehensive resource. Every major topic is thoroughly covered and every important concept is defined, described, and illustrated. Conceptually challenging but carefully explained articles are equally valuable to the practicing engineer, researchers, and students. A distinguished advisory board and contributors including many of the leading authors, professors, and researchers in the field today assist noted author and professor Richard Dorf in offering complete coverage of this rapidly expanding field. No other single volume available today offers this combination of broad coverage and depth of exploration of the topics. The Electrical Engineering Handbook will be an invaluable resource for electrical engineers for years to come.

Signal Processing: A Mathematical Approach is designed to show how many of the mathematical tools the reader knows can be used to understand and employ signal processing techniques in an applied environment. Assuming an advanced undergraduate- or graduate-level understanding of mathematics—including familiarity with Fourier series, matrices, probability, and statistics—this Second Edition: Contains new chapters on convolution and the vector DFT, plane-wave propagation, and the BLUE and Kalman filters Expands the material on Fourier analysis to three new chapters to provide additional background information Presents real-world examples of applications that demonstrate how mathematics is used in remote sensing Featuring problems for use in the classroom or practice, Signal Processing: A Mathematical Approach, Second Edition covers topics such as Fourier series and transforms in one and several variables; applications to acoustic and electro-magnetic propagation models, transmission and emission tomography, and image reconstruction; sampling and the limited data problem; matrix methods, singular value decomposition, and data compression; optimization techniques in signal and image reconstruction from projections; autocorrelations and power spectra; high-resolution methods; detection and optimal filtering; and eigenvector-based methods for array processing and statistical filtering, time-frequency analysis, and wavelets.

The reports cover progress to develop a signal processing structure that exploits available knowledge of the environment and of signal and noise variability induced by the environment. The research is directed toward passive sonar detection and classification, continuous wave (CW) and broadband signals, shallow water operation, both platform-mounted and distributed systems, and frequencies below 1 kHz. The results of this research are expected to lead to new passive sonar detectors and classifiers that take advantage of knowledge of medium variability and uncertainty. The results are mainly applicable to passive processing. However, the active processor can be considered "a detector matched to the estimated ocean." These results could have significant impact on Navy sonar system applications.

Accurate and timely environmental information can provide a tactical advantage to U.S. naval forces during warfare. This report analyzes the current environmental information system used by the U.S. Navy and Marine Corps and recommends ways to address uncertainty and leverage network-centric operating principles to enhance the value of environmental information.

In two editions spanning more than a decade, The Electrical Engineering Handbook stands as the definitive reference to the multidisciplinary field of electrical engineering. Our knowledge continues to grow, and so does the Handbook. For the third edition, it has grown into a set of six books carefully focused on specialized areas or fields of study. Each one represents a concise yet definitive collection of key concepts, models, and equations in its respective domain, thoughtfully gathered for convenient access. Combined, they constitute the most comprehensive, authoritative resource available. Circuits, Signals, and Speech and Image Processing presents all of the basic information related to electric circuits and components, analysis of circuits, the use of the Laplace transform, as well as signal, speech, and image processing using filters and algorithms. It also examines emerging areas such as text to speech synthesis, real-time processing, and embedded signal processing. Electronics, Power Electronics, Optoelectronics, Microwaves, Electromagnetics, and Radar delves into the fields of electronics, integrated circuits, power electronics, optoelectronics, electromagnetics, light waves, and radar, supplying all of the basic information required for a deep understanding of each area. It also devotes a section to electrical effects and devices and explores the emerging fields of microlithography and power electronics. Sensors, Nanoscience, Biomedical Engineering, and Instruments provides thorough coverage of sensors, materials and nanoscience, instruments and measurements, and biomedical systems and devices, including all of the basic information required to thoroughly understand each area. It explores the emerging fields of sensors, nanotechnologies, and biological effects. Broadcasting and Optical Communication Technology explores communications, information theory, and devices, covering all of the basic information needed for a thorough understanding of these areas. It also examines the emerging areas of adaptive estimation and optical communication. Computers, Software Engineering, and Digital Devices examines digital and logical devices, displays, testing, software, and computers, presenting the fundamental concepts needed to ensure a thorough understanding of each field. It treats the emerging fields of programmable logic, hardware description languages, and parallel computing in detail. Systems, Controls, Embedded Systems, Energy, and Machines explores in detail the fields of energy devices, machines, and systems as well as control systems. It provides all of the fundamental concepts needed for thorough, in-depth understanding of each area and devotes special attention to the emerging area of embedded systems. Encompassing the work of the world's foremost experts in their respective specialties, The Electrical Engineering Handbook, Third Edition remains the most convenient, reliable source of information available. This edition features the latest developments, the broadest scope of coverage, and new material on nanotechnologies, fuel cells, embedded systems, and biometrics. The engineering community has relied on the Handbook for more than twelve years, and it will continue to be a platform to launch the next wave of advancements. The Handbook's latest incarnation features a protective slipcase, which helps you stay organized without overwhelming your bookshelf. It is an attractive addition to any collection, and will help keep each volume of the Handbook as fresh as your latest research.

These Proceedings, consisting of Parts A and B, contain the edited versions of most of the papers presented at the annual Review of Progress in Quantitative Nondestructive Evaluation held at the University of Washington, Seattle on July 30 to August 4, 1995. The Review was organized by the Center for NDE at Iowa State University, in cooperation with the Ames Laboratory of the USDOE, the American Society of Nondestructive Testing, the Department of Energy, the National Institute of Standards and Technology, the Federal Aviation Administration, the National Science Foundation IndustryiUniversity Cooperative Research Centers, and the Working Group in Quantitative NDE. This year's Review of Progress in QNDE was attended by approximately 450 participants from the US and many foreign countries who presented over 375 papers. The meeting was divided into 36 sessions with as many as four sessions running concurrently. The Review covered all phases of NDE research and development from fundamental investigations to engineering applications or inspection systems, and it included many important methods of inspection science from acoustics to x-rays. In the last several years, the Review has stabilized at about its current size. Most participants seem to agree it is large enough to permit a full-scale overview of the latest developments but still small enough to retain the collegial atmosphere which has marked the Review since its inception. The Proceedings are structured in a format to reflect the organization of the Review itself, producing a more logical organization for both the meeting and the present volume.

This book provides comprehensive coverage of the detection and processing of signals in underwater acoustics. Background material on active and passive sonar systems, underwater acoustics, and statistical signal processing makes the book a self-contained and valuable resource for graduate students, researchers, and active practitioners alike. Signal detection topics span a range of common signal types including signals of known form such as active sonar or communications signals; signals of unknown form, including passive sonar and narrowband signals; and transient signals such as marine mammal vocalizations. This text, along with its companion volume on beamforming, provides a thorough treatment of underwater acoustic signal processing that speaks to its author’s broad experience in the field.

"Digital Sonar Design in Underwater Acoustics Principles and Applications" provides comprehensive and up-to-date coverage of research on sonar design, including the basic theory and techniques of digital signal processing, basic concept of information theory, ocean acoustics, underwater acoustic signal propagation theory, and underwater signal processing theory. This book discusses the general design procedure and approaches to implementation, the design method, system simulation theory and techniques, sonar tests in the laboratory, lake and sea, and practical validation criteria and methods for digital sonar design. It is intended for researchers in the fields of underwater signal processing and sonar design, and also for navy officers and ocean explorers. Qihu Li is a professor at the Institute of Acoustics, Chinese Academy of Sciences, and an academician of the Chinese Academy of Sciences.

Acoustic Signal Processing for Ocean Explortion has two major goals: (i) to present signal processing algorithms that take into account the models of acoustic propagation in the ocean and; (ii) to give a perspective of the broad set of techniques, problems, and applications arising in ocean exploration. The book discusses related issues and problems focused in model based acoustic signal processing methods. Besides addressing the problem of the propagation of acoustics in the ocean, it presents relevant acoustic signal processing methods like matched field processing, array processing, and localization and detection techniques. These more traditional contexts are herein enlarged to include imaging and mapping, and new signal representation models like time/frequency and wavelet transforms. Several applied aspects of these topics, such as the application of acoustics to fisheries, sea floor swath mapping by swath bathymetry and side scan sonar, autonomous underwater vehicles and communications in underwater are also considered.

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 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.

This book aims to develop a framework for a fully explanatory theory of speech production and speech perception. It emphasises the difference between static models (primarily descriptive) and dynamic models that attempt to show how the basic linguistics and phonetics are related in an actual human speaker/listener.

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