Acoustic Signal Processing In Passive Sonar System With

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.
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.
The first book exclusively on sonar and sonar technology. Written by an engineer (with over 40 years of experience in the field) for engineers. Taking an engineering approach rather than a physics/math one it provides an understanding of the basic principles of sonar and develops the formulae and "rules of thumb" for sonar design and performance analysis.
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.

"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.

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.

Surface Acoustic Wave Devices and Their Signal Processing Applications is a textbook that combines experiment and theory in assessing the signal processing applications of surface acoustic wave (SAW) devices. The operating principles of SAW devices are described from a circuit design viewpoint. This book is comprised of 18 chapters and begins with a historical background on surface acoustic waves and a discussion on the merits of SAW devices as well as their applications. The next chapter introduces the reader to the basics of acoustic waves and piezoelectricity, together with the effect of acoustic bulk waves on the performance of SAW filters. The principles of linear phase SAW filter design and equivalent circuit models for a SAW filter are then described. The remaining chapters focus on trade-offs in linear phase SAW filter design; compensation for second-order effects; harmonic SAW delay lines for gigahertz frequencies; and coding techniques using linear SAW transducers. The final chapter highlights Some other significant alternative design techniques and applications for SAW devices. This monograph will be suitable for engineering or physics students as well as engineers, scientists, and technical staff in industry who seek further information on SAW-based circuits, systems, and applications.

Modal Array Signal Processing: Principles and Applications of Acoustic Wavefield Decomposition
Underwater Signal and Data Processing
Acoustic Signal Processing for Ocean Exploration
Underwater Acoustic Signal Processing
Acoustic Green's Function Extraction in the Ocean
Acoustic MIMO Signal Processing

Signal Processing Techniques and Applications
A Mathematical Approach, Second Edition
Issues in Acoustic Signal — Image Processing and Recognition
Passive Acoustic Signal Processing Capabilities

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.

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.*

Matched field processing (MFP) provides a means of attaining the full gains available from the shallow-water acoustic channel in passive

sonar signal processing. By modeling the full field structure of acoustic signals propagating in the ocean MFP offers the potential for both detection gain (through its better signal model) and localization gain (through its additional discrimination capability in range and depth) over traditional planewave processing. However, high spatial ambiguities and mismatch present formidable challenges in practice limiting the performance gains that are realistically achievable with MFP. Prediction of MFP localization performance is a challenging problem. MFP replica (steering) vectors can be

highly ambiguous in range and depth resulting in significant non-local estimation errors at low signal-to-noise ratios (SNRs)-errors not modeled by traditional localization measures such as the Cramer-Rao bound. Recent work has demonstrated the accuracy of an interval-error-based method referred to herein as the "method of interval errors" (MIE), in predicting mean-squared error localization performance well into the threshold region where non-local errors may dominate. This work uses the MIE to predict the mean-squared error accuracy of MFP range and depth estimates for two well-known approaches:

(i) conventional beamforming (equivalent to maximum likelihood estimation for white noise) and (ii) Capon-MVDR adaptive beamforming. Simulation results will characterize localization performance as a function of SNR, for apertures and environments of interest. Particular attention will be given to the "threshold SNR" (below which localization performance degrades rapidly due to global estimation errors) and to the minimum SNR required to achieve acceptable range/depth localization. Initial work will also be presented assessing the MIE's potential to characterize localization performance in the

presence of mismatch.
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.*

207 5. 3 Third-Order Designs 216 5. 4 Higher-Order designs 221 6. Optimal Arrays for Cylindrically Isotropic Fields 222 6. 1 Maximum Gain for Omnidirectional Microphones 222 6. 2 Optimal Weights for Maximum Directional Gain 224 6. 3 Solution for Optimal Weights for Maximum Front-to-Back Ratio for Cylindrical Noise 225 7. Sensitivity to Microphone Mismatch and Noise 230 8.

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.
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.
Audio Signal Processing for Next-Generation Multimedia Communication Systems
Sonar Systems

Passive Acoustic Monitoring of Moving Targets by Sparse Underwater Sensor Network
Digital Sonar Design in Underwater Acoustics
Signal Processing for Passive Detection and Classification of Underwater Acoustic Signals
Principles and Applications
Wideband STAP (WB-STAP) for Passive Sonar
Handbook of Signal Processing in Acoustics
Threshold Region Performance Prediction for Adaptive Matched Field Processing Localization
Underwater Acoustic Data Processing

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.
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.

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 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.
By providing all the basic knowledge needed to assess how useful active noise control will be for a given problem, this book assists in the designing, setting up, and tuning of an active noise-control system. Written for students who have no prior knowledge of acoustics, signal processing, or noise control but who do have a reasonable grasp of basic physics and mathematics, the text is short and descriptive, leaving all mathematical details and proofs concerning vibrations, signal processing and the like to more advanced texts or research monographs. The book can thus be used in independent study, in a classroom with laboratories, or in

conjunction with a kit for experiment or demonstration. Topics covered include basic acoustics, human perception and sound, sound intensity and related concepts, fundamentals of passive noise-control strategies, basics of digital systems and adaptive controllers, and active noise control systems.

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.

Applied Underwater Acoustics
Engineering Applications
Surface Acoustic Wave Devices and Their Signal Processing Applications
Signal Processing for Multicarrier Modulation in Underwater Acoustic Communication and Passive Radar
Sonar for Practising Engineers
Passive Acoustic Monitoring of Cetaceans
Machine Learning in Passive Ocean Acoustics for Localizing and Characterizing Events
The First Eight Decades, 1923-2008

[Acoustic Signal Processing for Telecommunication]
[Sonar Signal Processing]

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.

This paper addresses an innovative method for passive sonar signal processing where it is required to suppress a field of moving acoustic interferes while simultaneously enhancing the signal from a weak moving source. Motivated by the Space-Time Adaptive Processing (STAP) technology for similar radar applications, we propose a solution to the passive broadband sonar problem using Wideband Space-Time Adaptive processing (WB-STAP) in the space-frequency domain. The overall objective is to develop models of the source/receiver dynamics directly into the *problem formulation and analyze time varying nature of the data vector and the covariance matrix by operating in the spatio-frequency domain. This allows STAP-like formulation for the broadband passive sonar in the space-frequency domain. In particular by employing subaperture techniques and exploiting the underlying model using eigen-subspace methods it is possible to obtain optimum adaptive weight vectors that require significantly less number of data samples compared to conventional methods. In addition, this formulation allows simultaneous tracking of all targets present in the field of view by jointly estimating their arrival angles and Doppler parameters.*

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*

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.*

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.

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.*

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.

The Electrical Engineering Handbook - Six Volume Set
A History of the Acoustics Division of the Naval Research Laboratory
Immersive Audio Signal Processing
Signal Processing
Modeling, Detection, and Estimation
Signaling Processing for Passive Detection and Classification of

Underwater Acoustic Signals
Handbook of Digital Signal Processing
Review of Progress in Quantitative Nondestructive Evaluation

Leak Detection for Underground Storage Tanks
Signal Processing for Passive Detection and Classification of Underwater Acoustic SignalsUnderwater Acoustic Signal ProcessingModeling, Detection, and EstimationSpringer
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.

The acoustic Green's function (GF) is the key to understanding the acoustic properties of ocean environments. With knowledge of the acoustic GF, the physics of sound propagation, such as dispersion, can be analyzed; underwater communication over thousands of miles *can be understood; physical properties of the ocean, including ocean temperature, ocean current speed, as well as seafloor bathymetry, can be investigated. Experimental methods of acoustic GF extraction can be categorized as active methods and passive methods. Active methods are based on employment of man-made sound sources. These active methods require less computational complexity and time, but may cause harm to marine mammals.*

Passive methods cost much less and do not harm marine mammals, but require more theoretical and computational work. Both methods have advantages and disadvantages that should be carefully tailored to fit the need of each specific environment and application. In this dissertation, we study one passive method, the noise interferometry method, and one active method, the inverse filter processing method, to achieve acoustic GF *extraction in the ocean. The passive method of noise interferometry makes use of ambient noise to extract an approximation to the acoustic GF. In an environment with a diffusive distribution of sound sources, sound waves that pass through two hydrophones at two locations carry the information of the acoustic GF between these two locations; by listening to the long-term ambient noise signals and cross-correlating the noise data recorded at* *two locations, the acoustic GF emerges from the noise cross-correlation function (NCF); a coherent stack of many realizations of NCFs yields a good approximation to the acoustic GF between these two locations, with all the deterministic structures clearly exhibited in the waveform. To test the performance of noise interferometry in different types of ocean environments, two field experiments were performed and ambient noise data were collected* *in a 100-meter deep coastal ocean environment and a 600-meter deep ocean environment. In the coastal ocean environment, the collected noise data were processed by coherently stacking five days of cross-correlation functions between pairs of hydrophones separated by 5 km, 10 km and 15 km, respectively. NCF waveforms were modeled using the KRAKEN normal mode model, with the difference between the NCFs and the acoustic GFs quantified by* *a weighting function. Through waveform inversion of NCFs, an optimal geoacoustic model was obtained by minimizing the two-norm misfit between the simulation and the measurement. Using a simulated time-reversal mirror, the extracted GF was back propagated from the receiver location to the virtual source, and a strong focus was found in the vicinity of the source, which provides additional support for the optimality of the aforementioned* *geoacoustic model. With the extracted GF, dispersion in experimental shallow water environment was visualized in the time-frequency representation. Normal modes of GFs were separated using the time-warping transformation. By separating the modes in the frequency domain of the time-warped signal, we isolated modal arrivals and reconstructed the NCF by summing up the isolated modes, thereby significantly improving the signal-to-noise ratio of*

NCFs. Finally, these reconstructed NCFs were employed to estimate the depth-averaged current speed in the Florida Straits, based on an effective sound speed approximation. In the mid-deep ocean environment, the noise data were processed using the same noise interferometry method, but the obtained NCFs were not as good as those in the coastal ocean environment. Several highly possible reasons of the difference in the noise interferometry *performance were investigated and discussed. The first one is the noise source composition, which is different in the spectrograms of noise records in two environments. The second is strong ocean current variability that can result in coherence loss and undermine the utility of coherent stacking. The third one is the downward refracting sound speed profile, which impedes strong coupling between near surface noise sources and the near-bottom* *instruments. The active method of inverse filter processing was tested in a long-range deep-ocean environment. The high-power sound source, which was located near the sound channel axis, transmitted a pre-designed signal that was composed of a precursor signal and a communication signal. After traveling 1428.5 km distance in the north Pacific Ocean, the transmitted signal was detected by the receiver and was processed using the inverse filter. The* *probe signal, which was composed of M sequences and was known at the receiver, was utilized for the GF extraction in the inverse filter; the communication signal was then interpreted with the extracted GF. With a glitch in the length of communication signal, the inverse filter processing method was shown to be effective for long-range low-frequency deep ocean acoustic communication. In summary, this dissertation explored two creative* *methods to extract the acoustic GFs in the ocean. The extracted acoustic GFs were utilized both for studying the physical properties of the ocean and for underwater communication. The study combined experimental data analysis and numerical simulation, using various signal processing techniques. This work is valuable in both passive acoustic remote sensing and active acoustic communication.*

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.*

Passive Acoustic Source Detection and Tracking Using Personal Computer (PC) and Digital Signal Processing (DSP) Hardware

Active Noise Control Primer

Initial Survey of Existing Algorithms and Deficiencies

Leif Bjørnø

Microphone Arrays

Environmental Information for Naval Warfare

Model-Based Processing for Underwater Acoustic Arrays