Fundamentals Of Spherical Array Processing Springer Topics In Signal Processing

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Phased Array Antennas - An Introduction | Lecture #8 | Alan Fenn *Focused Near-Field Techniques for Evaluating Adaptive Phased Arrays | Lecture #3 | Alan Fenn Planewaves, Uniform Linear Arrays and the Array Manifold Vector* 161 Array Processing Introduction to Radar Systems – Lecture 1 – Introduction; Part 1 fMRI Short Course #9: ROI Analysis Building a Radar Data Cube with MATLAB and Phased Array System Toolbox *Bloodless Surgery Using Dual-Mode Ultrasound Arrays: The Future is Now* ~~Phased Array Receivers for Radio Astronomy, Remote Sensing, and Satellite Comm - Karl Warnick~~ Fundamentals of Radar ~~New Structures and Architectures for Communication Systems~~ *9.1: What is an Array? - Processing Tutorial Phased array of speakers* ~~5G Phased Array Antenna Design and Beamforming using CST~~

Ep 8. Analog versus Digital Beamforming (with Bengt Lindoff) [Wireless Future Podcast]

TSP #181 - Starlink Dish Phased Array Design, Architecture \u0026 RF In-depth Analysis

ArrayProcessing. What is ArrayProcessing?~~2.8 - MIMO TECHNIQUES - CAPACITY \u0026 COVERAGE ENHANCEMENT IN 4G LTE~~ ~~Phased Array Antennas~~ *Starlink Teardown: DISHY DESTROYED!* **Finite Antenna Array design using HFSS** *Software Defined Radio Introduction to Three-Dimensional (3D) Arrays*

Blind Deconvolution Using Unconventional Beamforming*How Acoustics are Used to Detect and Localize an Active Shooter or Sniper Basics of Antennas and Beamforming - Massive MIMO Networks*

6.8: Combining Steering Behaviors: Flocking - The Nature of Code

Adaptive Antennas and Degrees of Freedom | Lecture #1 | Alan Fenn An Introduction to 3D Beamforming What the HECK is a Tensor?!? *Fundamentals Of Spherical Array Processing*
Including numerous examples with complete, worked-out solutions, this book is designed to present the fundamental concepts of electromagnetic field theory as they relate to modern engineering ...

1.5: SPHERICAL COORDINATE SYSTEM
Snieder, Roel and Sens-Schönfelder, Christoph 2015. Seismic interferometry and stationary phase at caustics. Journal of Geophysical Research: Solid Earth, Vol. 120 ...

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Sushil K. Misra, PhD
Swaminathan, Vikhram V. Gibson, Larry R. Pinti, Marie Prakash, Shaurya Bohn, Paul W. and Shannon, Mark A. 2012. Nanotechnology for Sustainable Development. p. 17.

Essentials of Micro- and Nanofluidics
that is, the phase of the arriving energy is constant over any plane perpendicular to the direction of arrival, as shown in Figure 2.1. Because most sources of electromagnetic energy are small, these ...

Chapter 2: Antennas
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exceptional foundation in the STEM fields. Applications of technology and data ...

Business Administration Master of business administration (MBA) degree
Sushil K. Misra received his Ph. D. degree in Physics from St. Louis University, St. Louis, Missouri, USA in the area of condensed matter physics. He did his post-doctoral work at the University of ...

Sushil K. Misra, PhD
Statistical distributions useful in general insurance. Inferences from general insurance data. Experience rating. Credibility theory: full credibility, partial credibility, Bayesian credibility.

This book provides a comprehensive introduction to the theory and practice of spherical microphone arrays, and was written for graduate students, researchers and engineers who work with spherical microphone arrays in a wide range of applications. The new edition includes additions and modifications, and references supplementary Matlab code to provide the reader with a straightforward start for own implementations. The book is also accompanied by a Matlab manual, which explains how to implement the examples and simulations presented in the book. 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. In turn, 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 highlight 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 those that achieve maximum directivity and maximum robustness are developed, along with the Dolph – Chebyshev beamformer. The final chapter discusses more advanced beamformers, such as MVDR (minimum variance distortionless response) and LCMV (linearly constrained minimum variance) types, which are tailored to the measured sound field. Mathworks kindly distributes the Matlab sources for this book on https://www.mathworks.com/matlabcentral/fileexchange/68655-fundamentals-of-spherical-array-processing.

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 describes the background and technology of array signal modeling. It presents the concept and formulation of beamformers and discusses several commonly used array performance measures. It also introduces two traditional types of beamformers: delay-and-sum and optimum beamformers. Chapter 1 includes background information on array processing, while Chapters 2 and 3 discuss the DFT-based frequency-domain implementation of a broadband beamformer and the design of subband beamformers for frequency-domain broadband beamformers. Chapter 4 presents the FIR-based, time-domain implementation of the broadband beamformer, where the FIR beamformer is designed by separately designing the subband beamformers and the corresponding FIR filters. The techniques for optimal design of the FIR beamformer are developed in Chapter 5, and Chapters 6 and 7 focus on the modal beamforming problem for circular arrays for the frequency-domain modal beamformer and the time-domain modal beamformer. Lastly, the final chapters present frequency-domain and time-domain modal beamformers for spherical arrays.

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.

This open access book provides a concise explanation of the fundamentals and background of the surround sound recording and playback technology Ambisonics. It equips readers with the psychoacoustical, signal processing, acoustical, and mathematical knowledge needed to understand the inner workings of modern processing utilities, special equipment for recording, manipulation, and reproduction in the higher-order Ambisonic format. The book comes with various practical examples based on free software tools and open scientific data for reproducible research. The book ' s introductory section offers a perspective on Ambisonics spanning from the origins of coincident recordings in the 1930s to the Ambisonic concepts of the 1970s, as well as classical ways of applying Ambisonics in first-order coincident sound scene recording and reproduction that have been practiced since the 1980s. As, from time to time, the underlying mathematics become quite involved, but should be comprehensive without sacrificing readability, the book includes an extensive mathematical appendix. The book offers readers a deeper understanding of Ambisonic technologies, and will especially benefit scientists, audio-system and audio-recording engineers. In the advanced sections of the book, fundamentals and modern techniques as higher-order Ambisonic decoding, 3D audio effects, and higher-order recording are explained. Those techniques are shown to be suitable to supply audience areas ranging from studio-sized to hundreds of listeners, or headphone-based playback, regardless whether it is live, interactive, or studio-produced 3D audio material.

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 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.

This book constitutes the proceedings of the 14th International Conference on Latent Variable Analysis and Signal Separation, LVA/ICA 2018, held in Guildford, UK, in July 2018.The 52 full papers were carefully reviewed and selected from 62 initial submissions. As research topics the papers encompass a wide range of general mixtures of latent variables models but also theories and tools drawn from a great variety of disciplines such as structured tensor decompositions and applications; matrix and tensor factorizations; ICA methods; nonlinear mixtures; audio data and methods; signal separation evaluation campaign; deep learning and data-driven methods; advances in phase retrieval and applications; sparsity-related methods; and biomedical data and methods.

Providing a wealth of information on fundamental topics in the areas of linear air and underwater acoustics, as well as space-time signal processing, this book provides real-world design and analysis equations. As a consequence of the interdisciplinary nature of air and underwater acoustics, the book is divided into two parts: Acoustic Field Theory and Space-Time Signal Processing. It covers the fundamentals of acoustic wave propagation as well as the fundamentals of aperture theory, array theory, and signal processing. Starting with principles and using a consistent, mainly standard notation, this book develops, in detail, basic results that are useful in a variety of air and underwater acoustic applications. Numerous figures, examples, and problems are included.

Automatic speech recognition (ASR) systems are finding increasing use in everyday life. Many of the commonplace environments where the systems are used are noisy, for example users calling up a voice search system from a busy cafeteria or a street. This can result in degraded speech recordings and adversely affect the performance of speech recognition systems. As the use of ASR systems increases, knowledge of the state-of-the-art in techniques to deal with such problems becomes critical to system and application engineers and researchers who work with or on ASR technologies. This book presents a comprehensive survey of the state-of-the-art in techniques used to improve the robustness of speech recognition systems to these degrading external influences. Key features: Reviews all the main noise robust ASR approaches, including signal separation, voice activity detection, robust feature extraction, model compensation and adaptation, missing data techniques and recognition of reverberant speech. Acts as a timely exposition of the topic in light of more widespread use in the future of ASR technology in challenging environments. Addresses robustness issues and signal degradation which are both key requirements for practitioners of ASR. Includes contributions from top ASR researchers from leading research units in the field