Acoustic Signal Processing For Telecommunication The Springer International Series In Engineering And Computer Science

Fundamentals of Signal Processing for Sound and Vibration Engineers

This work examines the applications of algorithms to audio-video and telecommunications, with special emphasis on wavelets, digital filters, neural nets, image processing and acoustic processing.

Audio Signal Processing and Coding

Speech Dereverberation gathers together an overview, a mathematical formulation of the problem and the state-of-the-art solutions for dereverberation. Speech Dereverberation presents current approaches to the problem of reverberation. It provides a review of topics in room acoustics and also describes performance measures for dereverberation. The algorithms are then explained with mathematical analysis and examples that enable the reader to see the strengths and weaknesses of the various techniques, as well as giving an understanding of the questions still to be addressed. Techniques rooted in speech enhancement are included, in addition to a treatment of multichannel blind acoustic system identification and inversion. The TRINICON framework is shown in the context of dereverberation to be a generalization of the signal processing for a range of analysis and enhancement techniques. Speech Dereverberation is suitable for students at masters and doctoral level, as well as established researchers.

Springer Handbook of Speech Processing

Adaptive filters with a large number of coefficients are usually involved in both network and acoustic echo cancellation. Consequently, it is important to improve the convergence rate and tracking of the conventional algorithms used for these applications. This can be achieved by exploiting the sparseness character of the echo paths. Identification of sparse impulse responses was addressed mainly in the last decade with the development of the so-called ``proportionate''-type algorithms. The goal of this book is to present the most important sparse adaptive filters developed for echo cancellation. Besides a comprehensive review of the basic proportionate-type algorithms, we also present some of the latest developments in the field and propose some new solutions for further performance improvement, e.g., variable step-size versions and novel proportionate-type affine projection algorithms. An experimental study is also provided in order to compare many sparse adaptive filters in different echo cancellation scenarios. Table of Contents: Introduction / Sparseness Measures / Performance Measures / Wiener and Basic Adaptive Filters / Basic Proportionate-Type NLMS Adaptive Filters / The Exponentiated Gradient Algorithms / The Mu-Law PNLMS and Other PNLMS-Type Algorithms / Variable Step-Size PNLMS Algorithms / Proportionate Affine Projection Algorithms / Experimental Study

Digital Underwater Acoustic Communications

This work examines the applications of algorithms to audio-video and telecommunications, with special emphasis on wavelets, digital filters, neural nets, image processing and acoustic processing.

Wireless Communications

Now available in a three-volume set, this updated and expanded edition of the bestselling The Digital Signal Processing Handbook continues to provide the engineering community with authoritative coverage of the fundamental and specialized aspects of information-bearing signals in digital form. Encompassing essential background material, technical details, standards, and software, the second edition reflects cutting-edge information on signal processing algorithms and protocols related to speech, audio, multimedia, and video processing technology associated with standards ranging from WiMax to MP3 audio, low-power/high-performance DSPs, color image processing, and chips on video. Drawing on the experience of leading engineers, researchers, and scholars, the three-volume set contains 29 new chapters that address multimedia and Internet technologies, tomography, radar systems, architecture, standards, and future applications in speech, acoustics, video, radar, and telecommunications. Emphasizing theoretical concepts, Digital Signal Processing Fundamentals provides comprehensive coverage of the basic foundations of DSP and includes the following parts: Signals and Systems; Signal Representation and Quantization;

Fourier Transforms; Digital Filtering; Statistical Signal Processing; Adaptive Filtering; Inverse Problems and Signal Reconstruction; and Time–Frequency and Multirate Signal Processing.

Acoustic Signal Processing for Telecommunication

Digital Signal Processing Fundamentals

An authoritative text covering the key topics, concepts and analytical tools needed to understand modern communication and radar systems. With numerous examples, exercises and computational results, it is an invaluable resource for graduate students in electrical and computer engineering, and practitioners in communications and radar engineering.

Communication Acoustics

Digital Underwater Acoustic Communications focuses on describing the differences between underwater acoustic communication channels and radio channels, discusses loss of transmitted sound in underwater acoustic channels, describes digital underwater acoustic communication signal processing, and provides a comprehensive reference to digital underwater acoustic communication equipment. This book is designed to serve as a reference for postgraduate students and practicing engineers involved in the design and analysis of underwater acoustic communications systems as well as for engineers involved in underwater acoustic engineering. Introduces the basics of underwater acoustics, along with the advanced functionalities needed to achieve reliable communications in underwater environment Identifies challenges in underwater acoustic channels relative to radio channels, underwater acoustic propagation, and solutions Shows how multi-path structures can be thought of as time diversity signals Presents a new, robust signal processing system, and an advanced FH-SS system for multimedia underwater acoustic communications with moderate communication ranges (above 20km) and rates (above 600bps) Describes the APNFM system for underwater acoustic communication equipment (including both civil and military applications), to be employed in active sonar to improve its performance

Active Noise Control Systems

Acoustic MIMO Signal Processing

Multimedia Signal Processing is a comprehensive and accessible text to the theory and applications of digital signal processing (DSP). The applications of DSP are pervasive and include multimedia systems, cellular communication, adaptive network management, radar, pattern recognition, medical signal processing, financial data forecasting, artificial intelligence, decision making, control systems and search engines. This book is organised in to three major parts making it a coherent and structured presentation of the theory and applications of digital signal processing. A range of important topics are covered in basic signal processing, model-based statistical signal processing and their applications. Part 1: Basic Digital Signal Processing gives an introduction to the topic, discussing sampling and quantization, Fourier analysis and synthesis, Z-transform, and digital filters. Part 2: Model-based Signal Processing covers probability and information models, Bayesian inference, Wiener filter, adaptive filters, linear prediction hidden Markov models and independent component analysis. Part 3: Applications of Signal Processing in Speech, Music and Telecommunications explains the topics of speech and music processing, echo cancellation, deconvolution and channel equalization, and mobile communication signal processing. Covers music signal processing, explains the anatomy and psychoacoustics of hearing and the design of MP3 music coder Examines speech processing technology including speech models, speech coding for mobile phones and speech recognition Covers single-input and multiple-inputs denoising methods, bandwidth extension and the recovery of lost speech packets in applications such as voice over IP (VoIP) Illustrated throughout, including numerous solved problems, Matlab experiments and demonstrations Companion website features Matlab and C++ programs with electronic copies of all figures. This book is ideal for researchers, postgraduates and senior undergraduates in the fields of digital signal processing, telecommunications and statistical data analysis. It will also be a valuable text to professional engineers in telecommunications and audio and signal processing industries.

Multimedia Signal Processing

This book focuses on a class of single-channel noise reduction methods that are performed in the frequency domain via the short-time Fourier transform (STFT). The simplicity and relative effectiveness of this class of approaches make them the dominant choice in practical systems. Even though many popular algorithms have been proposed through more than four decades of continuous research, there are a number of critical areas where our understanding and capabilities still remain quite rudimentary, especially with respect to the relationship between noise reduction and speech distortion. All existing frequency-domain algorithms, no matter how they are developed, have one feature in common: the solution is eventually expressed as a gain function applied to the STFT of the noisy signal only in the current frame. As a result, the narrowband signal-to-noise ratio (SNR) cannot be improved, and any gains achieved in noise reduction on the fullband basis come with a price to pay, which is speech distortion. In this book, we present a new perspective on the problem by exploiting the difference between speech and typical noise in circularity and interframe self-correlation, which were ignored in the past. By gathering the STFT of the microphone signal of the current frame, its complex conjugate, and the STFTs in the previous frames, we construct several new, multiple-observation signal models similar to a microphone array system: there are multiple noisy speech observations, and their speech components are correlated but not completely coherent while their noise components are presumably uncorrelated. Therefore, the multichannel Wiener filter and the minimum variance distortionless response (MVDR) filter that were usually associated with microphone arrays will be developed for single-channel noise reduction in this book. This might instigate a paradigm shift geared toward speech distortionless noise reduction techniques. Table of Contents: Introduction / Problem Formulation / Performance Measures / Linear and Widely Linear Models / Optimal Filters with Model 1 / Optimal Filters with Model 2 / Optimal Filters with Model 3 / Optimal Filters with Model 4 / Experimental Study

Speech Processing in Modern Communication

A blend of introductory material and advanced signal processing and communication techniques, of critical importance to underwater system and network development This book, which is the first to describe the processing techniques central to underwater OFDM, is arranged into four distinct sections: First, it describes the characteristics of underwater acoustic channels, and stresses the difference from wireless radio channels. Then it goes over the basics of OFDM and channel coding. The second part starts with an overview of the OFDM receiver, and develops various modules for the receiver design in systems with single or multiple transmitters. This is the main body of the book. Extensive experimental data sets are used to verify the receiver performance. In the third part, the authors discuss applications of the OFDM receiver in i) deep water channels, which may contain very long separated multipath clusters, ii) interference-rich environments, where an unintentional interference such as Sonar will be present, and iii) a network with multiple users where both non-cooperative and cooperative underwater communications are developed. Lastly, it describes the development of a positioning system with OFDM waveforms, and the progress on the OFDM modem development. Closely related industries include the development and manufacturing of autonomous underwater vehicles (AUVs) and scientific sensory equipment. AUVs and sensors in the future could integrate modems, based on the OFDM technology described in this book. Contents includes: Underwater acoustic channel characteristics/OFDM basics/Peak-to-average-ratio control/Detection and Doppler estimation (Doppler scale and CFO)/Channel estimation and noise estimation/A block-by-block progressive receiver and performance results/Extensions to multi-input multi-output OFDM/Receiver designs for multiple users/Cooperative underwater OFDM (Physical layer network coding and dynamic coded cooperation)/Localization with OFDM waveforms/Modem developments A valuable resource for Graduate and postgraduate students on electrical engineering or physics courses; electrical engineers, underwater acousticians, communications engineers

Audio Signal Processing for Next-Generation Multimedia Communication Systems

Digital Signal Processing in Audio and Acoustical Engineering

An in-depth treatment of algorithms and standards for perceptual coding of high-fidelity audio, this self-contained reference surveys and addresses all aspects of the field. Coverage includes signal processing and perceptual (psychoacoustic) fundamentals, details on relevant research and signal models, details on standardization and applications, and details on performance measures and perceptual measurement systems. It includes a comprehensive bibliography with over 600 references, computer exercises, and MATLAB-based projects for use in EE multimedia, computer science, and DSP courses. An ftp site containing supplementary material such as wave files, MATLAB programs and workspaces for the students to solve some of the numerical problems and computer exercises in the book can be found at ftp://ftp.wiley.com/public/sci_tech_med/audio_signal

Handbook of Signal Processing in Acoustics

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.

Acoustic MIMO Signal Processing

This book brings together many advanced topics in network and acoustic echo cancellation aimed towards enhancing the echo cancellation performance of next-generation telecommunication systems. The resulting compendium provides a coherent treatment of such topics not found otherwise in journals or other books.

Blind Speech Separation

Surface acoustic wave (SAW) devices are widely used in mobile communications, a rapidly evolving market. This book gives an overview on the latest SAW technologies with an emphasis on the design and simulation of devices, such as resonator-based devices employing the SH-type leaky SAW.

OFDM for Underwater Acoustic Communications

This book collects a wealth of information about spatial audio coding into one comprehensible volume. It is a thorough reference to the 3GPP and MPEG Parametric Stereo standards and the MPEG Surround multi-channel audio coding standard. It describes key developments in coding techniques, which is an important factor in the optimization of advanced entertainment, communications and signal processing applications. Until recently, technologies for coding audio signals, such as redundancy reduction and sophisticated source and receiver models did not incorporate spatial characteristics of source and receiving ends. Spatial audio coding achieves much higher compression ratios than conventional coders. It does this by representing multi-channel audio signals as a downmix signal plus side information that describes the perceptually-relevant spatial information. Written by experts in spatial audio coding, Spatial Audio Processing: reviews psychoacoustics (the relationship between physical measures of sound and the corresponding percepts) and spatial audio sound formats and reproduction systems; brings together the processing, acquisition, mixing, playback, and perception of spatial audio, with the latest coding techniques; analyses algorithms for the efficient manipulation of multiple, discrete and combined spatial audio channels, including both MP3 and MPEG Surround; shows how the same insights on source and receiver models can also be applied for manipulation of audio signals, such as the synthesis of virtual auditory scenes employing head-related transfer function (HRTF) processing and stereo to N-channel audio upmix. Audio processing research engineers and audio coding research and implementation engineers will find this an insightful guide. Academic audio and psychoacoustic researchers, including post-graduate and third/fourth year students taking courses in signal processing, audio and speech processing, and telecommunications, will also benefit from the information inside.

Proceedings of International Conference on Information, Communications, and Signal Processing

Multimedia Signal Processing is a comprehensive and accessible text to the theory and applications of digital signal processing (DSP). The applications of DSP are pervasive and include multimedia systems, cellular communication, adaptive network management, radar, pattern recognition, medical signal processing, financial data forecasting, artificial intelligence, decision making, control systems and search engines. This book is organised in to three major parts making it a coherent and structured presentation of the theory and applications of digital signal processing. A range of important topics are covered in basic signal processing, model-based statistical signal processing and their applications. Part 1: Basic Digital Signal Processing gives an introduction to the topic, discussing sampling and quantization, Fourier analysis and synthesis, Z-transform, and digital filters. Part 2: Model-based Signal Processing covers probability and information models, Bayesian inference, Wiener filter, adaptive filters, linear prediction hidden Markov models and independent component analysis. Part 3: Applications of Signal Processing in Speech, Music and Telecommunications explains the topics of speech and music processing, echo cancellation, deconvolution and channel equalization, and mobile communication signal processing. Covers music signal processing, explains the anatomy and psychoacoustics of hearing and the design of MP3 music coder Examines speech processing technology including speech models, speech coding for mobile phones and speech recognition Covers single-input and multiple-inputs denoising methods, bandwidth extension and the recovery of lost speech packets in applications such as voice over IP (VoIP) Illustrated throughout, including numerous solved problems, Matlab experiments and demonstrations Companion website features Matlab and C++ programs with electronic copies of all figures. This book is ideal for researchers, postgraduates and senior undergraduates in the fields of digital signal processing, telecommunications and statistical data analysis. It will also be a valuable text to professional engineers in telecommunications and audio and signal processing industries.

IEEE ASSP Workshop on Applications of Signal Processing to Audio and Acoustics

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

Surface Acoustic Wave Devices in Telecommunications

Signals and Images: Advances and Results in Speech, Estimation, Compression, Recognition, Filtering, and Processing cohesively combines contributions from field experts to deliver a comprehensive account of the latest developments in signal processing. These experts detail the results of their research related to audio and speech enhancement, acoustic image estimation, video compression, biometric recognition, hyperspectral image analysis, tensor decomposition with applications in communications, adaptive sparse-interpolated filtering, signal processing for power line communications, bio-inspired signal processing, seismic data processing, arithmetic transforms for spectrum computation, particle filtering in cooperative networks, three-dimensional television, and more. This book not only shows how signal processing theory is applied in current and emerging technologies, but also demonstrates how to tackle key problems such as how to enhance speech in the time domain, improve audio quality, and meet the desired electrical consumption target for controlling carbon emissions. Signals and Images: Advances and Results in Speech, Estimation, Compression, Recognition, Filtering, and Processing serves as a guide to the next generation of signal processing solutions for speech and video coding, hearing aid devices, big data processing, smartphones, smart digital communications, acoustic sensors, and beyond.

A Perspective on Single-channel Frequency-domain Speech Enhancement

This is the world's first edited book on independent component analysis (ICA)-based blind source separation (BSS) of convolutive mixtures of speech. This book brings together a small number of leading researchers to provide tutorial-like and in-depth treatment on major ICA-based BSS topics, with the objective of becoming the definitive source for current, comprehensive, authoritative, and yet accessible treatment.

Advances in Network and Acoustic Echo Cancellation

With a novel, less classical approach to the subject, the authors have written a book with the conviction that signal processing should be taught to be fun. The treatment is therefore less focused on the mathematics and more on the conceptual aspects, the idea being to allow the readers to think about the subject at a higher conceptual level, thus building the foundations for more advanced topics. The book remains an engineering text, with the goal of helping students solve real-world problems. In this vein, the last chapter pulls together the individual topics as discussed throughout the book into an in-depth look at the development of an end-to-end communication system, namely, a modem for communicating digital information over an analog channel.

Multimedia Signal Processing

Signal Processing Algorithms for Communication and Radar Systems

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.

Signals and Images

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

Speech Enhancement in the Karhunen-Loeve Expansion Domain

Starting with essential maths, fundamentals of signals and systems, and classical concepts of DSP, this book presents, from an application-oriented perspective, modern concepts and methods of DSP including machine learning for audio acoustics and engineering. Content highlights include but are not limited to room acoustic parameter measurements, filter design, codecs, machine learning for audio pattern recognition and machine audition, spatial audio, array technologies and hearing aids. Some research outcomes are fed into book as worked examples. As a research informed text, the book attempts to present DSP and machine learning from a new and more relevant angle to acousticians and audio engineers. Some MATLAB® codes or frameworks of algorithms are given as downloads available on the CRC Press website. Suggested exploration and mini project ideas are given for "proof of concept" type of exercises and directions for further study and investigation. The book is intended for researchers, professionals, and senior year students in the field of audio acoustics.

Spatial Audio Processing

This book is devoted to the study of the problem of speech enhancement whose objective is the recovery of a signal of interest (i.e., speech) from noisy observations. Typically, the recovery process is accomplished by passing the noisy observations through a linear filter (or a linear transformation). Since both the desired speech and undesired noise are filtered at the same time, the most critical issue of speech enhancement resides in how to design a proper optimal filter that can fully take advantage of the difference between the speech and noise statistics to mitigate the noise effect as much as possible while maintaining the speech perception identical to its original form. The optimal filters can be designed either in the time domain or in a transform space. As the title indicates, this book will focus on developing and analyzing optimal filters in the Karhunen-Loève expansion (KLE) domain. We begin by describing the basic problem of speech enhancement and the fundamental principles to solve it in the time domain. We then explain how the problem can be equivalently formulated in the KLE domain. Next, we divide the general problem in the KLE domain into four groups, depending on whether interframe and interband information is accounted for, leading to four linear models for speech enhancement in the KLE domain. For each model, we introduce signal processing measures to quantify the performance of speech enhancement, discuss the formation of different cost functions, and address the optimization of these cost functions for the derivation of different optimal filters. Both theoretical analysis and experiments will be provided to study the performance of these filters and the links between the KLE-domain and time-domain optimal filters will be examined. Table of Contents: Introduction / Problem Formulation / Optimal Filters in the Time Domain / Linear Models for Signal Enhancement in the KLE Domain / Optimal Filters in the KLE Domain with Model 1 / Optimal Filters in the KLE Domain with Model 2 / Optimal Filters in the KLE Domain with Model 3 / Optimal Filters in the KLE Domain with Model 4 / Experimental Study

Speech Dereverberation

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.

Signal Processing for Communications

Now available in a three-volume set, this updated and expanded edition of the bestselling The Digital Signal Processing Handbook continues to provide the engineering community with authoritative coverage of the fundamental and specialized aspects of information-bearing signals in digital form. Encompassing essential background material, technical details, standards, and software, the second edition reflects cutting-edge information on signal processing algorithms and protocols related to speech, audio, multimedia, and video processing technology associated with standards ranging from WiMax to MP3 audio, low-power/high-performance DSPs, color image processing, and chips on video. Drawing on the experience of leading engineers, researchers, and scholars, the three-volume set contains 29 new chapters that address multimedia and Internet technologies, tomography, radar systems, architecture, standards, and future applications in speech, acoustics, video, radar, and telecommunications. This volume, Video, Speech, and Audio Signal Processing and Associated Standards, provides thorough coverage of the basic foundations of speech, audio, image, and video processing and associated applications to broadcast, storage, search and retrieval, and communications.

Video, Speech, and Audio Signal Processing and Associated

Standards

This publication deals with the application of advanced digital signal processing techniques and neural networks to various telecommunication problems. The editor presents the latest research results in areas such as arrays, mobile channels, acoustic echo cancellation, speech coding and adaptive filtering in varying environments.

Signal Processing Methods for Audio, Images and Telecommunications

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

Speech and Audio Signal Processing

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.

Academic Press Library in Signal Processing, Volume 7

Academic Press Library in Signal Processing, Volume 7: Array, Radar and Communications Engineering is aimed at university researchers, post graduate students and R&D engineers in the industry, providing a tutorial-based, comprehensive review of key topics and technologies of research in Array and Radar Processing, Communications Engineering and Machine Learning. Users will find the book to be an invaluable starting point to their research and initiatives. With this reference, readers will quickly grasp an unfamiliar area of research, understand the underlying principles of a topic, learn how a topic relates to other areas, and learn of research issues yet to be resolved. Presents a quick tutorial of reviews of important and emerging topics of research Explores core principles, technologies, algorithms and applications Edited and contributed by international leading figures in the field Includes comprehensive references to journal articles and other literature upon which to build further, more detailed knowledge

Communication Acoustics

- Speech Generation: Acoustics, Models and Applications (Arild Lacroix). - The Evolution of Digital Audio Technology (John Mourjopoulos). - Audio-Visual Interaction (Armin Kohlrausch) . - Speech and Audio Coding (Ulrich Heute) . - Binaural Technique (Dorte Hammerhoei, Henrik Moeller). - Auditory Virtual Environment (Pedro Novo). - Evolutionary Adaptions for Auditory Communication (Georg Klump). - A Functional View on the Human Hearing Organ (Herbert Hudde). - Modeling of Binaural Hearing (Jonas Braasch). - Psychoacoustics and Sound Quality (Hugo Fastl). - Semiotics for Engineers (Ute Jekosch). - Quality of Transmitted Speech for Humans and Machines (Sebastian Möller).

Signal Processing Methods for Audio, Images and Telecommunications

This book aims to convey to engineering students and researchers alike the relevant knowledge about the nature of acoustics, sound and hearing that will enable them to develop new technologies in this area through acquiring a thorough understanding of how sound and hearing works. There is currently no technical book available covering the communication path from sound sources through medium to the formation of auditory events in the brain – this book will fill this gap in the current book literature. It discusses the multidisciplinary area of acoustics, hearing, psychoacoustics, signal processing, speech and sound quality and is suitable for use as a main course textbook for senior undergraduate and graduate courses related to audio communication systems. It covers the basics of signal processing, traditional acoustics as well as the human hearing system and how to build audio techniques based on human hearing resolution. It discusses the technologies and applications for sound synthesis and reproduction, and for speech and audio quality evaluation.

Sparse Adaptive Filters for Echo Cancellation

Fundamentals of Signal Processing for Sound and Vibration Engineers is based on Joe Hammond’s many years of teaching experience at the Institute of Sound and Vibration Research, University of Southampton. Whilst the applications presented emphasise sound and vibration, the book focusses on the basic essentials of signal processing that ensures its appeal as a reference text to students and practitioners in all areas of mechanical, automotive, aerospace and civil engineering. Offers an excellent introduction to signal processing for students and professionals in the sound and vibration engineering field. Split into two parts, covering deterministic signals then random signals, and offering a clear explanation of their theory and application together with appropriate MATLAB examples. Provides an excellent study tool for those new to the field of signal processing. Integrates topics within continuous, discrete, deterministic and random signals to facilitate better understanding of the topic as a whole. Illustrated with MATLAB examples, some using ‘real’ measured data, as well as fifty MATLAB codes on an accompanying website.

Communications, Signal Processing, and Systems

Communications, Signal Processing, and Systems is a collection of contributions coming out of the International Conference on Communications, Signal Processing, and Systems (CSPS) held August 2012. This book provides the state-of-art developments of Communications, Signal Processing, and Systems, and their interactions in multidisciplinary fields, such as audio and acoustic signal processing. The book also examines Radar Systems, Chaos Systems, Visual Signal Processing and Communications and VLSI Systems and Applications. Written by experts and students in the fields of Communications, Signal Processing, and Systems.

Digital Signal Processing in Telecommunications

Microphone Arrays

"Professor Andreas F. Molisch, renowned researcher and educator, has put together the comprehensive book, Wireless Communications. The second edition, which includes a wealth of new material on important topics, ensures the role of the text as the key resource for every student, researcher, and practitioner in the field." —Professor Moe Win, MIT, USA Wireless communications has grown rapidly over the past decade from a niche market into one of the most important, fast moving industries. Fully updated to incorporate the latest research and developments, Wireless Communications, Second Edition provides an authoritative overview of the principles and applications of mobile communication technology. The author provides an in-depth analysis of current treatment of the area, addressing both the traditional elements, such as Rayleigh fading, BER in flat fading channels, and equalisation, and more recently emerging topics such as multi-user detection in CDMA systems, MIMO systems, and cognitive radio. The dominant wireless standards; including cellular, cordless and wireless LANs; are discussed. Topics featured include: wireless propagation channels, transceivers and signal processing, multiple access and advanced transceiver schemes, and standardised wireless systems. Combines mathematical descriptions with intuitive explanations of the physical facts, enabling readers to acquire a deep understanding of the subject. Includes new chapters on cognitive radio, cooperative communications and relaying, video coding, 3GPP Long Term Evolution, and WiMax; plus significant new sections on multi-user MIMO, 802.11n, and information theory. Companion website featuring: supplementary material on 'DECT', solutions manual and presentation slides for instructors, appendices, list of abbreviations and other useful resources.