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For Authentication in an E-World
Handbook of Signal Processing in Acoustics
The Development of the SPHINX System
New Advances and Trends
Theory and Application of Digital Signal Processing
MATLAB Applications
14th International Conference, KES 2010, Cardiff, UK, September 8-10, 2010, Proceedings
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Digital Signal Processing and Statistical Classification
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Digital Signal Processing and Applications with the TMS320C6713 and TMS320C6416 DSK
The Digital Signal Processing Handbook
Parallel Processing on VLSI Arrays
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Adaptive Signal Processing
Fundamentals of Speech Recognition
Adaptive Signal Processing
8th International Conference, SECITC 2015, Bucharest, Romania, June 11-12, 2015. Revised Selected Papers
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Rising Threats in Expert Applications and Solutions
Single Channel Phase-Aware Signal Processing in Speech Communication
Speech Recognition and Coding
Proceedings of the NATO Advanced Study Institute held at Bonas, France, June 29–July 10, 1981
Knowledge-Based and Intelligent Information and Engineering Systems
Deep Learning for NLP and Speech Recognition
Principles and Practice

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For Authentication in an E-World Springer
"This book covers basic and the advanced approaches in the design and implementation of multirate filtering"--Provided by publisher.
Handbook of Signal Processing in Acoustics IGI Global
This textbook explains Deep Learning Architecture, with applications to various NLP Tasks, including Document Classification, Machine Translation, Language Modeling, and Speech Recognition. With the widespread adoption of deep learning, natural language processing (NLP),and speech applications in many areas (including Finance, Healthcare, and Government) there is a growing need for one comprehensive resource that maps deep learning techniques to NLP and speech and provides insights into using the tools and libraries for real-world applications. Deep Learning for NLP and Speech Recognition explains recent deep learning methods applicable to NLP and speech, provides state-of-the-art approaches, and offers real-world case studies with code to provide hands-on experience. Many books focus on deep learning theory or deep learning for NLP-specific tasks while others are cookbooks for tools and libraries, but the constant flux of new algorithms, tools, frameworks, and libraries in a rapidly evolving landscape means that there are few available texts that offer the material in this book. The book is organized into three parts, aligning to different groups of readers and their expertise. The three parts are: Machine Learning, NLP, and Speech Introduction The first part has three chapters that introduce readers to the fields of NLP, speech recognition, deep learning and machine learning with basic theory and hands-on case studies using Python-based tools and libraries. Deep Learning Basics The five chapters in the second part introduce deep learning and various topics that are crucial for speech and text processing, including word embeddings, convolutional neural networks, recurrent neural networks and speech recognition basics. Theory, practical tips, state-of-the-art methods, experimentations and analysis in using the methods discussed in theory on real-world tasks. Advanced Deep Learning Techniques for Text and Speech The third part has five chapters that discuss the latest and cutting-edge research in the areas of deep learning that intersect with NLP and speech. Topics including attention mechanisms, memory augmented networks, transfer learning, multi-task learning, domain adaptation, reinforcement learning, and end-to-end deep learning for speech recognition are covered using case studies.
The Development of the SPHINX System IGI Global
This book constitutes the thoroughly refereed post-conference proceedings of the 8th International Conference on Security for Information Technology and Communications, SECITC 2015, held in Bucharest, Romania, in June 2015. The 17 revised full papers were carefully reviewed and selected from 36 submissions. In addition with 5 invited talks the papers cover topics such as Cryptographic Algorithms and Protocols, Security Technologies for IT&C, Information Security Management, Cyber Defense, and Digital Forensics.
New Advances and Trends Springer Science & Business Media
Although speech is the primary behavioral medium by which humans communicate, its auditory basis is poorly understood, having profound implications on efforts to ameliorate the behavioral consequences of hearing impairment and on the development of robust algorithms for computer speech recognition. In this volume, the authors provide an up-to-date synthesis of recent research in the area of speech processing in the auditory system, bringing together a diverse range of scientists to present the subject from an interdisciplinary perspective. Of particular concern is the ability to understand speech in uncertain, potentially adverse acoustic environments, currently the bane of both hearing aid and speech recognition technology. There is increasing evidence that the perceptual stability characteristic of speech understanding is due, at least in part, to elegant transformations of the acoustic signal performed by auditory mechanisms. As a comprehensive review of speech's auditory basis, this book will interest physiologists, anatomists, psychologists, phoneticians, computer scientists, biomedical and electrical engineers, and clinicians.
Theory and Application of Digital Signal Processing Artech House
158 2. Wiener Filtering 159 3. Speech Enhancement by Short-Time Spectral Modification 3. 1 Short-Time Fourier Analysis and Synthesis 159 160 3. 2 Short-Time Wiener Filter 161 3. 3 Power Subtraction 3. 4 Magnitude Subtraction 162 3. 5 Parametric Wiener Filtering 163 164 3. 6 Review and Discussion Averaging Techniques for Envelope Estimation 169 4. 169 4. 1 Moving Average 170 4. 2 Single-Pole Recursion 170 4. 3 Two-Sided Single-Pole Recursion 4. 4 Nonlinear Data Processing 171 5. Example Implementation 172 5. 1 Subband Filter Bank Architecture 172 173 5. 2 A-Posteriori-SNR Voice Activity Detector 5. 3 Example 175 6. Conclusion 175 Part IV Microphone Arrays 10 Superdirectional Microphone Arrays 181 Gary W. Elko 1. Introduction 181 2. Differential Microphone Arrays 182 3. Array Directional Gain 192 4. Optimal Arrays for Spherically Isotropic Fields 193 4. 1 Maximum Gain for Omnidirectional Microphones 193 4. 2 Maximum Directivity Index for Differential Microphones 195 4. 3 Maximimum Front-to-Back Ratio 197 4. 4 Minimum Peak Directional Response 200 4. 5 Beamwidth 201 5. Design Examples 201 5. 1 First-Order Designs 202 5. 2 Second-Order Designs 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.
MATLAB Applications CRC Press
"Mobile Speech and Advanced Natural Language Solutions" presents the discussion of the most recent advances in intelligent human-computer interaction, including fascinating new study findings on talk-in-interaction, which is the province of conversation analysis, a subfield in sociology/sociolinguistics, a new and emerging area in natural language understanding. Editors Amy Neustein and Judith A. Markowitz have recruited a talented group of contributors to introduce the next generation natural language technologies for practical speech processing applications that serve the consumer’s need for well-functioning natural language-driven personal assistants and other mobile devices, while also addressing business’ need for better

functioning IVR-driven call centers that yield a more satisfying experience for the caller. This anthology is aimed at two distinct audiences: one consisting of speech engineers and system developers; the other comprised of linguists and cognitive scientists. The text builds on the experience and knowledge of each of these audiences by exposing them to the work of the other.

14th International Conference, KES 2010, Cardiff, UK, September 8-10, 2010, Proceedings Theory and Applications of Digital Speech Processing

This collection covers new aspects of numerical methods in applied mathematics, engineering, and health sciences. It provides recent theoretical developments and new techniques based on optimization theory, partial differential equations (PDEs), mathematical modeling and fractional calculus that can be used to model and understand complex behavior in natural phenomena. Specific topics covered in detail include new numerical methods for nonlinear partial differential equations, global optimization, unconstrained optimization, detection of HIV-Protease, modelling with new fractional operators, analysis of biological models, and stochastic modelling.

Theory and Applications of Digital Speech Processing Pearson Education

Introduction to EEG- and Speech-Based Emotion Recognition Methods examines the background, methods, and utility of using electroencephalograms (EEGs) to detect and recognize different emotions. By incorporating these methods in brain-computer interface (BCI), we can achieve more natural, efficient communication between humans and computers. This book discusses how emotional states can be recognized in EEG images, and how this is useful for BCI applications. EEG and speech processing methods are explored, as are the technological basics of how to operate and record EEGs. Finally, the authors include information on EEG-based emotion recognition, classification, and a proposed EEG/speech fusion method for how to most accurately detect emotional states in EEG recordings. Provides detailed insight on the science of emotion and the brain signals underlying this phenomenon Examines emotions as a multimodal entity, utilizing a bimodal emotion recognition system of EEG and speech data Details the implementation of techniques used for acquiring as well as analyzing EEG and speech signals for emotion recognition

Digital Signal Processing and Statistical Classification CRC Press

The Electrical Engineer's Handbook is an invaluable reference source for all practicing electrical engineers and students. Encompassing 79 chapters, this book is intended to enlighten and refresh knowledge of the practicing engineer or to help educate engineering students. This text will most likely be the engineer’s first choice in looking for a solution; extensive, complete references to other sources are provided throughout. No other book has the breadth and depth of coverage available here. This is a must-have for all practitioners and students! The Electrical Engineer's Handbook provides the most up-to-date information in: Circuits and Networks, Electric Power Systems, Electronics, Computer-Aided Design and Optimization, VLSI Systems, Signal Processing, Digital Systems and Computer Engineering, Digital Communication and Communication Networks, Electromagnetics and Control and Systems. About the Editor-in-Chief... Wai-Kai Chen is Professor and Head Emeritus of the Department of Electrical Engineering and Computer Science at the University of Illinois at Chicago. He has extensive experience in education and industry and is very active professionally in the fields of circuits and systems. He was Editor-in-Chief of the IEEE Transactions on Circuits and Systems, Series I and II, President of the IEEE Circuits and Systems Society and is the Founding Editor and Editor-in-Chief of the Journal of Circuits, Systems and Computers. He is the recipient of the Golden Jubilee Medal, the Education Award, and the Meritorious Service Award from the IEEE Circuits and Systems Society, and the Third Millennium Medal from the IEEE. Professor Chen is a fellow of the IEEE and the American Association for the Advancement of Science. * 77 chapters encompass the entire field of electrical engineering. * THOUSANDS of valuable figures, tables, formulas, and definitions. * Extensive bibliographic references.

Innovative Security Solutions for Information Technology and Communications Springer Science & Business Media

The four-volume set LNAI 6276--6279 constitutes the refereed proceedings of the 14th International Conference on Knowledge-Based Intelligent Information and Engineering Systems, KES 2010, held in Cardiff, UK, in September 2010. The 272 revised papers presented were carefully reviewed and selected from 360 submissions. They present the results of high-quality research on a broad range of intelligent systems topics.

Concepts and Solutions Prentice Hall

“If you have built castles in the air, your work need not be lost; that is where they should be. Now put the foundations under them. ” - Henry David Thoreau, Walden Although engineering is a study entrenched firmly in belief of pr- matism, I have always believed its impact need not be limited to pr- matism. Pragmatism is not the boundaries that define engineering, just the (sometimes unforgiving) rules by which we sight our goals. This book studies two major problems of content-based video proce- ing for a media-based technology: Video Object Plane (VOP) Extr- tion and Representation, in support of the MPEG-4 and MPEG-7 video standards, respectively. After reviewing relevant image and video p- cessing techniques, we introduce the concept of Voronoi Ordered Spaces for both VOP extraction and representation to integrate shape infor- tion into low-level optimization algorithms and to derive robust shape descriptors, respectively. We implement a video object segmentation system with a novel surface optimization scheme that integrates Voronoi Ordered Spaces with existing techniques to balance visual information against predictions of models of a priori information. With these VOPs, we have explicit forms of video objects that give users the ability to - dress and manipulate video content. We outline a general methodology of robust data representation and comparison through the concept of complex partitioning mapped onto Directed Acyclic Graphs (DAGs).

Proceedings of FICR-TEAS 2020 Springer Science & Business Media

This book is a tutorial on digital techniques for waveform generation, digital filters, and digital signal processing tools and techniques The typical chapter begins with some theoretical material followed by working examples and experiments using the TMS320C6713-based DSPStarter Kit (DSK) The C6713 DSK is TI's newest signal processor based on the C6x processor (replacing the C6711 DSK)

Electronic Synthesis of Speech John Wiley & Sons

The series Advances in Industrial Control aims to report and encourage technology transfer in control engineering. The rapid development of control technology impacts all areas of the control discipline. New theory, new controllers, actuators, sensors, new industrial processes, computer methods, new applications, new philosophies, . . . , new challenges. Much of this development work resides in industrial reports, feasibility study papers and the reports of advanced collaborative projects. The series offers an opportunity for researchers to present an extended exposition of such new work in all aspects of industrial control for wider and rapid dissemination. The emerging technologies in control include fuzzy logic, intelligent control, neural networks and hardware developments like micro-electro-mechanical systems and autonomous vehicles. This volume describes the biological background, basic construction and application of the emerging technology of Genetic Algorithms. Dr Kim Man and his colleagues have written a book which is both a primer introducing the basic concepts and a research text which describes some of the more advanced applications of the genetic algorithmic method. The applications described are especially useful since they indicate the power of the GA method in solving a wide range of problems. These sections are also instructive in showing how the mechanics of the GA solutions are obtained thereby acting as a template for similar types of problems. The volume is a very welcome contribution to the Advances in Industrial Control Series. M. J. Grimble and M. A.

Digital Signal Processing Fundamentals Springer Science & Business Media

This book presents high-quality, peer-reviewed papers from the FICR International Conference on Rising Threats in Expert Applications and Solutions 2020, held at IIS University Jaipur, Rajasthan, India, on January 17–19, 2020. Featuring innovative ideas from researchers, academics, industry professionals and students, the book covers a variety of topics, including expert applications and artificial intelligence/machine learning; advanced web technologies, like IoT, big data, and cloud computing in expert applications; information and cybersecurity threats and solutions; multimedia applications in forensics, security and intelligence; advances in app development; management practices for expert applications; and social and ethical aspects of expert applications in applied sciences.

Genetic Algorithms for Control and Signal Processing IGI Global

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 expanded into a set of six books carefully focused on a specialized area or field of study. Each book represents a concise yet definitive collection of key concepts, models, and equations in its respective domain, thoughtfully gathered for convenient access. 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. Each article includes defining terms, references, and sources of further information. Encompassing the work of the world's foremost experts in their respective specialties, Circuits, Signals, and Speech and Image Processing features the latest developments, the broadest scope of coverage, and new material on biometrics.

Numerical Solutions of Realistic Nonlinear Phenomena Springer Science & Business Media

Digital Signal Processing and Applications with the TMS320C6713 and TMS320C6416 DSK Now in a new edition—the most comprehensive, hands-on introduction to digital signal processing The first edition of Digital Signal Processing and Applications with the TMS320C6713 and TMS320C6416 DSK is widely accepted as the most extensive text available on the hands-on teaching of Digital Signal Processing (DSP). Now, it has been fully updated in this valuable Second Edition to be compatible with the latest version (3.1) of Texas Instruments Code Composer Studio (CCS) development environment. Maintaining the original’s comprehensive, hands-on approach that has made it an instructor’s favorite, this new edition also features: Added program examples that illustrate DSP concepts in real-time and in the laboratory Expanded coverage of analog input and output New material on frame-based processing A revised chapter on IIR, which includes a number of floating-point example programs that explore IIR filters more comprehensively More extensive coverage of DSP/BIOS All programs listed in the text—plus additional applications—which are available on a companion website No other book provides such an extensive or comprehensive set of program examples to aid instructors in teaching DSP in a laboratory using audio frequency signals—making this an ideal text for DSP courses at the senior undergraduate and postgraduate levels. It also serves as a valuable resource for researchers, DSP developers, business managers, and technology solution providers who are looking for an overview and examples of DSP algorithms implemented using the TMS320C6713 and TMS320C6416 DSK.

Multirate Filtering for Digital Signal Processing: MATLAB Applications Springer Nature

Theory and Applications of Digital Speech ProcessingPrentice Hall

Digital Signal Processing and Applications with the TMS320C6713 and TMS320C6416 DSK John Wiley & Sons

An overview on the challenging new topic of phase-aware signal processing Speech communication technology is a key factor in human-machine interaction, digital hearing aids, mobile telephony, and automatic speech/speaker recognition. With the proliferation of these applications, there is a growing requirement for advanced methodologies that can push the limits of the conventional solutions relying on processing the signal magnitude spectrum. Single-Channel Phase-Aware Signal Processing in Speech Communication provides a comprehensive guide to phase signal processing and reviews the history of phase importance in the literature, basic problems in phase processing, fundamentals of phase estimation together with several applications to demonstrate the usefulness of phase processing. Key features: Analysis of recent advances demonstrating the positive impact of phase-based processing in pushing the limits of conventional methods. Offers unique coverage of the historical context, fundamentals of phase processing and provides several examples in speech communication. Provides a detailed review of many references and discusses the existing signal processing techniques required to deal with phase information in different applications involved with speech. The book supplies various examples and MATLAB® implementations delivered within the PhaseLab toolbox. Single-Channel Phase-Aware Signal Processing in Speech Communication is a valuable single-source for students, non-expert DSP engineers, academics and graduate students.

The Digital Signal Processing Handbook CRC Press

Based on a NATO Advanced Study Institute held in 1993, this book addresses recent advances in automatic speech recognition and speech coding. The book contains contributions by many of the most outstanding researchers from the best laboratories worldwide in the field. The contributions have been grouped into five parts: on acoustic modeling; language modeling; speech processing, analysis and synthesis; speech coding; and vector quantization and neural nets. For each of these topics, some of the best-known researchers were invited to give a lecture. In addition to these lectures, the topics were complemented with discussions and presentations of the work of those attending. Altogether, the reader is given a wide perspective on recent advances in the field and will be able to see the trends for future work.

Parallel Processing on VLSI Arrays Elsevier

"This book provides a general overview about research on ubiquitous and pervasive computing and its applications, discussing the recent progress in this area and pointing out to scholars what they should do (best practices) and should not do (bad practices)"--Provided by publisher.

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