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Multirate Filtering for Digital Signal Processing: MATLAB Applications Springer Nature
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.
8th International Conference, SECITC 2015, Bucharest, Romania, June 11-12, 2015. Revised Selected Papers John Wiley & Sons
Theory and Applications of Digital Speech

Processing is ideal for graduate students in digital signal processing, and undergraduate students in Electrical and Computer Engineering. With its clear, up-to-date, hands-on coverage of digital speech processing, this text is also suitable for practicing engineers in speech processing. This new text presents the basic concepts and theories of speech processing with clarity and currency, while providing hands-on computer-based laboratory experiences for students. The material is organized in a manner that builds a strong foundation of basics first, and then concentrates on a range of signal processing methods for representing and processing the speech signal.

Principles and Practice **Elsevier**

Over the last 20 years, approaches to designing speech and language processing algorithms have moved from methods based on linguistics and speech science to data-driven pattern recognition techniques. These techniques have been the focus of intense, fast-moving research and have contributed to significant advances in this field. Pattern Reco

Theory and Applications of Digital Speech Processing Springer

Theory and Applications of Digital Speech ProcessingPrentice Hall

14th International Conference, KES 2010, Cardiff, UK, September 8-10, 2010, Proceedings CRC Press

This is the first book to introduce and integrate advanced digital signal processing (DSP) and classification together, and the only volume to introduce state-of-the-art transforms including DFT, FFT, DCT, DHT, PCT, CDT, and ODT together for DSP and communication applications. You get step-by-step guidance in discrete-time domain signal processing and frequency domain signal analysis; digital filter design and adaptive filtering; multirate digital processing; and statistical signal classification. It also helps you overcome problems associated with multirate A/D and D/A converters.

Automatic Speech Analysis and Recognition Springer Science & Business Media

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.

Digital Signal Processing and Applications with the C6713 and C6416 DSK Springer Science & Business Media

This book discusses advances in smart and sustainable development of smart environments. The authors discuss the challenges faced in developing sustainable smart applications and provide potential solutions. The solutions are aimed at improving reliability and security with the goal of affordability, safety, and durability. Topics include health care applications, sustainable smart transportation systems, intelligent sustainable wearable electronics, and sustainable smart building and alert systems. Authors are from both industry and academia and present research from around the world. Addresses problems and solutions for sustainable development of smart cities; Includes applications such as healthcare, transportation, wearables, security, and more; Relevant for scientist and researchers working on real time smart city development.

Numerical Solutions of Realistic Nonlinear Phenomena Theory and Applications of Digital Speech Processing

Leading experts present the latest research results in adaptive signal processing Recent developments in signal processing have made it clear that significant performance gains can be achieved beyond those achievable using standard adaptive filtering approaches. Adaptive Signal Processing presents the next generation of algorithms that will produce these desired results, with an emphasis on important applications and theoretical advancements. This highly unique resource brings together leading authorities in the field writing on the key topics of significance, each at the cutting edge of its own area of specialty. It begins by addressing the problem of optimization in the complex domain, fully developing a framework that enables taking full advantage of the power of complex-valued processing. Then, the challenges of multichannel processing of complex-valued signals are explored. This comprehensive volume goes on to cover Turbo processing, tracking in the subspace domain, nonlinear sequential state estimation, and speech-bandwidth extension. Examines the seven most important topics in adaptive filtering that will define the next-generation adaptive filtering solutions Introduces the powerful adaptive signal processing methods developed within the last ten years to account for the characteristics of real-life data: non-Gaussianity, non-circularity, non-stationarity, and non-linearity Features self-contained chapters, numerous examples to clarify concepts, and end-of-chapter problems to reinforce understanding of the material Contains contributions from acknowledged leaders in the field Adaptive Signal Processing is an invaluable tool for graduate students, researchers, and practitioners working in the areas of signal processing, communications, controls, radar, sonar, and biomedical engineering.

The Development of the SPHINX System

Pearson Education India

One of the goals of artificial intelligence (AI) is creating autonomous agents that must make decisions based on uncertain and incomplete information. The goal is to design rational agents that must take the best action given the information available and their goals. Decision Theory Models for Applications in Artificial Intelligence: Concepts and Solutions provides an introduction to different types of decision theory techniques, including MDPs, POMDPs, Influence Diagrams, and Reinforcement Learning, and illustrates their application in artificial intelligence. This book provides insights into the advantages and challenges of using decision theory models for developing intelligent systems.

New Issues and Trends Frontiers Media SA

The creation of the text really began in 1976 with the author being involved with a group of researchers at Stanford University and the Naval Ocean Systems Center, San Diego. At that time, adaptive techniques were more laboratory (and mental) curiosities than the accepted and pervasive categories of signal processing that they have become. Over the lasl 10 years, adaptive filters have become standard components in telephony, data communications, and signal detection and tracking systems. Their use and consumer acceptance will undoubtedly only increase in the future. The mathematical principles underlying adaptive signal processing were initially fascinating and were my first experience in seeing applied mathematics work for a paycheck. Since that time, the application of even more advanced mathematical techniques have kept the area of adaptive signal processing as exciting as those initial days. The text seeks to be a bridge between the open literature in the professional journals, which is usually quite concentrated, concise, and advanced, and the graduate classroom and research environment where underlying principles are often more important.

The Digital Signal Processing Handbook Springer Science & Business Media

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

Essential principles, practical examples, current applications, and leading-edge research. In this book, Thomas F. Quatieri presents the field's most intensive, up-to-date tutorial and reference on discrete-time speech signal processing. Building on his MIT graduate course, he introduces key principles, essential applications, and state-of-the-art research, and he identifies limitations that point the way to new research opportunities. Quatieri provides an excellent balance of theory and application, beginning with a complete framework for understanding discrete-time speech signal processing. Along the way, he presents important advances never before covered in a speech signal processing text book, including sinusoidal speech processing, advanced time-frequency analysis, and nonlinear aeroacoustic speech production modeling. Coverage includes: Speech production and speech perception: a dual view Crucial distinctions between stochastic and deterministic problems Pole-zero speech models Homomorphic signal processing Short-time Fourier transform analysis/synthesis Filter-bank and wavelet analysis/synthesis Nonlinear measurement and modeling techniques The book's in-depth applications coverage includes speech coding, enhancement, and modification; speaker recognition; noise reduction; signal restoration; dynamic range compression, and more. Principles of Discrete-Time Speech Processing also contains an exceptionally complete series of examples and Matlab exercises, all carefully integrated into the book's coverage of theory and applications.

Mobile Speech and Advanced Natural Language Solutions CRC Press

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.

Concepts and Solutions CRC Press

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.

Pattern Recognition in Speech and Language Processing Prentice Hall

This book is the result of the second NATO Advanced Study Institute on speech processing held at the Chateau de Bonas, France, from June 29th to July 10th, 1981. This Institute provided a high-level coverage of the fields of speech transmission, recognition and understanding, which constitute important areas where research activity has re cently been associated with actual industrial developments. This book will therefore include both fundamental and applied topics. Ten survey papers by some of the best specialists in the field are included. They give an up-to-date presentation of several important problems in automatic speech processing. As a consequence the book can be considered as a reference manual on some important areas of automatic speech processing. The surveys are indicated by 'a * in the table of contents. This book also contains research papers corresponding to original works, which were presented during the panel sessions of the Institute. For the sake of clarity the book has been divided into five sections : 1. Speech Analysis and Transmission: An emphasis has been laid on the techniques of linear prediction (LPC), and the problems involved in the transmission of speech at various bit rates are addressed in details. 2. Acoustics and Phonetics : One'of the major bottleneck in the development of speech recogni tion systems remains the transcription of the continuous speech wave into some discrete strings or lattices of phonetic symbols. Two survey papers discuss this problem from different points of view and several practical systems are also described.

Electronic Synthesis of Speech Springer Science & Business Media

"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.

Theory and Practice Springer Science & Business Media

The Handbook of Neural Computation is a practical, hands-on guide to the design and implementation of neural networks used by scientists and engineers to tackle difficult and/or time-consuming problems. The handbook bridges an information pathway between scientists and engineers in different disciplines who apply neural networks to similar probl

Adaptive Signal Processing Artech House

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Single Channel Phase-Aware Signal Processing in Speech Communication Springer Science & Business Media

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.

Biometric Solutions Springer

Guest Editor: JOSEF A. NOSSEK This is a special issue of the Journal of VLSI Signal Processing comprising eight contributions invited for publica tion on the basis of novel work presented in a special session on "Parallel Processing on VLSI Arrays" at the International Symposium on Circuits and Systems (ISCAS) held in New Orleans in May 1990. Massive parallelism to cope with high-speed requirements stemming from real-time applications and the restrictions in architectural and circuit design, such as regularity and local connectedness, brought about by the VLSI technology are the key questions addressed in these eight papers. They can be grouped into three subsections elaborating on: • Simulation of continuous physical systems, i. e. , numerically solving partial differential equations. • Neural architectures for image processing and pattern recognition. • Systolic architectures for implementing regular and irregular algorithms in VLSI technology. The paper by A. Fettweis and O. Nitsche advocates a signal processing approach for the numerical integration of partial differential equations (PD Es). It is based on the principles of multidimensional wave digital filters (MDWDFs) thereby preserving the passivity of energy dissipating physical systems. It is particularly suited for systems ofPDEs involving time and finite propagation speed. The basic ideas are explained using Maxwell's equa tions as a vehicle for the derivation of a multidimensional equivalent circuit representing the spatially infinitely extended arrangement with only very few circuit elements.