Subband Adaptive Filtering Theory And Implementation

Thank you for reading **Subband Adaptive Filtering Theory And Implementation**. As you may know, people have look numerous times for their favorite novels like this Subband Adaptive Filtering Theory And Implementation, but end up in infectious downloads.
Rather than enjoying a good book with a cup of coffee in the afternoon, instead they juggled with some harmful virus inside their desktop computer.

Subband Adaptive Filtering Theory And Implementation is available in our book collection an online access to it is set as public so you can download it instantly.
Our digital library hosts in multiple countries, allowing you to get the most less latency time to download any of our books like this one.
Kindly say, the Subband Adaptive Filtering Theory And Implementation is universally compatible with any devices to read

Adaptive Signal Models - Michael M. Goodwin 2012-09-10
Adaptive Signal Models: Theory, Algorithms and Audio Applications presents methods for deriving mathematical models of natural signals. The introduction covers the fundamentals of analysis-synthesis systems and signal representations. Some of the topics in the introduction include perfect and near-perfect reconstruction, the distinction between parametric and nonparametric methods, the role of compaction in signal modeling, basic and overcomplete signal expansions, and time-frequency resolution issues. These topics arise throughout the book as do a number of other topics such as filter banks and multiresolution. The second chapter gives a detailed development of the sinusoidal model as a parametric extension of the short-time Fourier transform. This leads to multiresolution sinusoidal modeling techniques in Chapter Three, where wavelet-like approaches are merged with the sinusoidal model to yield improved models. In Chapter Four, the analysis-synthesis residual is considered; for realistic synthesis, the residual must be separately modeled after coherent components (such as sinusoids) are removed. The residual modeling approach is based on psychoacoustically motivated nonuniform filter banks. Chapter Five deals with pitch-synchronous versions of both the wavelet and the Fourier transform; these allow for compact models of pseudo-periodic signals. Chapter Six discusses recent algorithms for deriving signal representations based on time-frequency atoms; primarily, the matching pursuit algorithm is reviewed and extended. The signal models discussed in the book are compact, adaptive, parametric, time-frequency representations that are useful for analysis, coding, modification, and synthesis of natural signals such as audio. The models are all interpreted as methods for decomposing a signal in terms of fundamental time-frequency atoms; these interpretations, as well as the adaptive and parametric natures of the models, serve to link the various methods dealt with in the text. Adaptive Signal Models: Theory, Algorithms and Audio Applications serves as an excellent reference for researchers of signal processing and may be used as a text for advanced courses on the topic.
Proceedings of 14th International Conference on Electromechanics and Robotics “Zavalishin's Readings” - Andrey Ronzhin 2019-08-29
This book features selected papers presented at the 14th International Conference on Electromechanics and Robotics ‘Zavalishin’s Readings’ – ER(ZR) 2019, held in Kursk, Russia, on April 17–20, 2019. The contributions, written by professionals, researchers and students, cover topics in the field of automatic control systems, electromechanics, electric power engineering and electrical engineering, mechatronics, robotics, automation and vibration technologies. The Zavalishin's Readings conference was established as a tribute to the memory of Dmitry Aleksandrovich Zavalishin (1900–1968) – a Russian scientist, corresponding member of the USSR Academy of Sciences, and founder of the school of valve energy converters based on electric machines and valve converters energy. The first conference was organized by the Institute of Innovative Technologies in Electromechanics and Robotics at the Saint Petersburg State University of Aerospace Instrumentation in 2006. The 2019 conference was held with the XIII International Scientific and Technical Conference “Vibration 2019”, and was organized by Saint Petersburg State University of Aerospace Instrumentation (SUAI), Saint Petersburg Institute for Informatics and Automation of the Russian Academy of Sciences (SPIIRAS) and the Southwest State University (SWSU) in with cooperation Russian Foundation for Basic Research (project No. 19-08-20021).
Adaptive Filtering - Paulo S.R. Diniz 2013-03-14
Adaptive Filtering: Algorithms and Practical Implementation, Second Edition, presents a concise overview of adaptive filtering, covering as many algorithms as possible in a unified form that avoids repetition and simplifies notation. It is suitable as a textbook for senior undergraduate or first-year graduate courses in adaptive signal processing and adaptive filters. The philosophy of the presentation is to expose the material with a solid theoretical foundation, to concentrate on algorithms that really work in a finite-precision implementation, and to provide easy access to working algorithms. Hence, practicing engineers and scientists will also find the book to be an excellent reference. This second edition contains a substantial amount of new material: -Two new chapters on nonlinear and subband adaptive filtering; -Linearly constrained Weiner filters and LMS algorithms; -LMS algorithm behavior in fast adaptation; -Affine projection algorithms; -Derivation smoothing; -MATLAB codes for algorithms.
Blind Source Separation - Ganesh R. Naik 2014-05-21
Blind Source Separation intends to report the new results of the efforts on the study of Blind Source Separation (BSS). The book collects novel research ideas and some training in BSS, independent component analysis (ICA), artificial intelligence and signal processing applications. Furthermore, the research results previously scattered in many journals and conferences worldwide are methodically edited and presented in a unified form. The book is likely to be of interest to university researchers, R&D engineers and graduate students in computer science and electronics who wish to learn the core principles, methods, algorithms and applications of BSS. Dr. Ganesh R. Naik works at University of Technology, Sydney, Australia; Dr. Wenwu Wang works at University of Surrey, UK.
Starting Digital Signal Processing in Telecommunication Engineering - Tomasz P. Zieliński 2021-01-29
This hands-on, laboratory driven textbook helps readers understand principles of digital signal processing (DSP) and basics of software-based digital communication, particularly software-defined networks (SDN) and software-defined radio (SDR). In the book only the most important concepts are presented. Each book chapter is an introduction to computer laboratory and is accompanied by complete laboratory exercises and ready-to-go Matlab programs with figures and comments (available at the book webpage and running also in GNU Octave 5.2 with free software packages), showing all or most details of relevant algorithms. Students are tasked to understand programs, modify them, and apply presented concepts to recorded real RF signal or simulated received signals, with modelled transmission condition and hardware imperfections. Teaching is done by showing examples and their modifications to different real-world telecommunication-like applications. The book consists of three parts: introduction to DSP (spectral analysis and digital filtering), introduction to DSP advanced topics (multi-rate, adaptive, model-based and multimedia - speech, audio, video - signal analysis and processing) and introduction to software-defined modern telecommunication systems (SDR technology, analog and digital modulations, single- and multi-carrier systems, channel estimation and correction as well as synchronization issues). Many real signals are processed in the book, in the first part – mainly speech and audio, while in the second part – mainly RF recordings taken from RTL-SDR USB stick and ADALM-PLUTO module, for example captured IQ data of VOR avionics signal, classical FM radio with RDS, digital DAB/DAB+ radio and 4G-LTE digital telephony. Additionally, modelling and simulation of some transmission scenarios are tested in software in the book, in particular TETRA, ADSL and 5G signals. Provides an introduction to digital signal processing and software-based digital communication; Presents a transition from digital signal processing to software-defined telecommunication; Features a suite of pedagogical materials including a laboratory test-bed and computer exercises/experiments.
Adaptive Signal Processing - Jacob Benesty 2013-03-09
For the first time, a reference on the most relevant applications of adaptive filtering techniques. Top researchers in the field contributed

chapters addressing applications in acoustics, speech, wireless and networking, where research is still very active and open.

Embedded Signal Processing with the Micro Signal Architecture - Woon-Seng Gan 2007-02-26

This is a real-time digital signal processing textbook using the latest embedded Blackfin processor Analog Devices, Inc (ADI). 20% of the text is dedicated to general real-time signal processing principles. The remaining text provides an overview of the Blackfin processor, its programming, applications, and hands-on exercises for users. With all the practical examples given to expedite the learning development of Blackfin processors, the textbook doubles as a ready-to-use user's guide. The book is based on a step-by-step approach in which readers are first introduced to the DSP systems and concepts. Although, basic DSP concepts are introduced to allow easy referencing, readers are recommended to complete a basic course on "Signals and Systems" before attempting to use this book. This is also the first textbook that illustrates graphical programming for embedded processor using the latest LabVIEW Embedded Module for the ADI Blackfin Processors. A solutions manual is available for adopters of the book from the Wiley editorial department.

Adaptive Filtering - Paulo S. R. Diniz 2019-11-28

In the fifth edition of this textbook, author Paulo S.R. Diniz presents updated text on the basic concepts of adaptive signal processing and adaptive filtering. He first introduces the main classes of adaptive filtering algorithms in a unified framework, using clear notations that facilitate actual implementation. Algorithms are described in tables, which are detailed enough to allow the reader to verify the covered concepts. Examples address up-to-date problems drawn from actual applications. Several chapters are expanded and a new chapter 'Kalman Filtering' is included. The book provides a concise background on adaptive filtering, including the family of LMS, affine projection, RLS, set-membership algorithms and Kalman filters, as well as nonlinear, sub-band, blind, IIR adaptive filtering, and more. Problems are included at the end of chapters. A MATLAB package is provided so the reader can solve new problems and test algorithms. The book also offers easy access to working algorithms for practicing engineers.

Adaptive Filters - Ali H. Sayed 2008-04-14

Adaptive filtering is a topic of immense practical and theoretical value, having applications in areas ranging from digital and wireless communications to biomedical systems. This book enables readers to gain a gradual and solid introduction to the subject, its applications to a variety of topical problems, existing limitations, and extensions of current theories. The book consists of eleven parts?each part containing a series of focused lectures and ending with bibliographic comments, problems, and computer projects with MATLAB solutions.

Acoustic Signal Processing for Telecommunication - Steven L. Gay 2012-12-06

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.

Audio Signal Processing for Next-Generation Multimedia Communication Systems - Yiteng (Arden) Huang 2007-05-08

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.

Software-Defined Radio for Engineers - Alexander M. Wyglinski 2018-04-30

Based on the popular Artech House classic, Digital Communication Systems Engineering with Software-Defined Radio, this book provides a practical approach to quickly learning the software-defined radio (SDR) concepts needed for work in the field. This up-to-date volume guides readers on how to quickly prototype wireless designs using SDR for real-world testing and experimentation. This book explores advanced wireless communication techniques such as OFDM, LTE, WLA, and hardware targeting. Readers will gain an understanding of the core concepts behind wireless hardware, such as the radio frequency front-end, analog-to-digital and digital-to-analog converters, as well as various processing technologies. Moreover, this volume includes chapters on timing estimation, matched filtering, frame synchronization message decoding, and source coding. The orthogonal frequency division multiplexing is explained and details about HDL code generation and deployment are provided. The book concludes with coverage of the WLAN toolbox with OFDM beacon reception and the LTE toolbox with downlink reception. Multiple case studies are provided throughout the book. Both MATLAB and Simulink source code are included to assist readers with their projects in the field.

Advanced Signal Processing and Digital Noise Reduction - Saeed V. Vaseghi 1996-07-25

Noise cancellation is particularly important in the new mobile communications field, with respect to background noise and acoustic interference in moving vehicles. This comprehensive text develops a coherent and structured presentation of a broad range of the theory and application of statistical signal processing, with emphasis on digital noise reduction algorithms. Other applications covered are spectral estimation, channel equalisation, speech coding over noisy channels, speech recognition in adverse environments, active noise control, echo cancellation, restoration of lost filters, and adaptive notch filters.

Window Functions and Their Applications in Signal Processing - K. M. M. Prabhu 2018-09-03

Window functions—otherwise known as weighting functions, tapering functions, or apodization functions—are mathematical functions that are zero-valued outside the chosen interval. They are well established as a vital part of digital signal processing. Window Functions and their Applications in Signal Processing presents an exhaustive and detailed account of window functions and their applications in signal processing, focusing on the areas of digital spectral analysis, design of FIR filters, pulse compression radar, and speech signal processing. Comprehensively reviewing previous research and recent developments, this book: Provides suggestions on how to choose a window function for particular applications Discusses Fourier analysis techniques and pitfalls in the computation of the DFT Introduces window functions in the continuous-time and discrete-time domains Considers two implementation strategies of window functions in the time- and frequency domain Explores well-known applications of window functions in the fields of radar, sonar, biomedical signal analysis, audio processing, and synthetic aperture radar

FPGA-based Implementation of Signal Processing Systems - Roger Woods 2017-05

Revised edition of: FPGA-based implementation of signal processing systems / Roger Woods ... [et al.]. 2008.

Partial-Update Adaptive Signal Processing - Kutluyil Dogancay 2008-09-17

Partial-update adaptive signal processing algorithms not only permit significant complexity reduction in adaptive filter implementations, but can also improve adaptive filter performance in telecommunications applications. This book gives state-of-the-art methods for the design and development of partial-update adaptive signal processing algorithms for use in systems development. Partial-Update Adaptive Signal Processing provides a comprehensive coverage of key partial updating schemes, giving detailed information on the theory and applications of acoustic and network echo cancellation, channel equalization and multiuser detection. It also examines convergence and stability issues for partial update algorithms, providing detailed complexity analysis and a unifying

treatment of partial-update techniques. Features: • Advanced analysis and design tools • Application examples illustrating the use of partial-update adaptive signal processing • MATLAB codes for developed algorithms This unique reference will be of interest to signal processing and communications engineers, researchers, R&D engineers and graduate students. "This is a very systematic and methodical treatment of an adaptive signal processing topic, of particular significance in power limited applications such as in wireless communication systems and smart ad hoc sensor networks. I am very happy to have this book on my shelf, not to gather dust, but to be consulted and used in my own research and teaching activities" – Professor A. G. Constantinides, Imperial College, London About the author: Kutluyil Dogançay is an associate professor of Electrical Engineering at the University of South Australia. His research interests span statistical and adaptive signal processing and he serves as a consultant to defence and private industry. He was the Signal Processing and Communications Program Chair of IDC Conference 2007, and is currently chair of the IEEE South Australia Communications and Signal Processing Chapter. Advanced analysis and design tools Algorithm summaries in tabular format Case studies illustrate the application of partial update adaptive signal processing
Multimedia Signal Processing - Saeed V. Vaseghi 2007-10-22
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.
On Adaptive Filtering in Oversampled Subbands - Stephan Weiss 1998
For a number of applications like acoustic echo cancellation, adaptive filters are required to identify very long impulse responses. To reduce the computational cost in implementations, adaptive filtering in subband is known to be beneficial. Based on a review of popular fullband adaptive filtering algorithms and various subband approaches, this thesis investigates the implementation, design, and limitations of oversampled subband adaptive filter systems based on modulated complex and real valued filter banks. The main aim is to achieve a computationally efficient implementation for adaptive filter systems, for which fast methods of performing both the subband decomposition and the subband processing are researched. Therefore, a highly efficient polyphase implementation of a complex valued modulated generalized DFT (GDFT) filter bank with a judicious selection of properties for non-integer oversampling ratios is introduced. By modification, a real valued single sideband modulated filter bank is derived. Non-integer oversampling ratios are particularly important when addressing the efficiency of the subband processing. Analysis is presented to decide in which cases it is more advantageous to perform real or complex valued subband processing. Additionally, methods to adaptively adjust the filter lengths in subband adaptive filter (SAF) systems are discussed. Convergence limits for SAFs and the accuracy of the achievable equivalent fullband model based on aliasing and other distortions introduced by the employed filter banks are explicitly derived. Both an approximation of the minimum mean square error and the model accuracy can be directly linked to criteria in the design of the prototype filter for the filter bank. Together with an iterative least-squares design algorithm, it is therefore possible to construct filter banks for SAF applications with pre-defined performance limits. Simulation results are presented which demonstrate the validity and properties of the discussed SAF methods and their advantage over fullband and critically sampled SAF systems.
Digital Signal Processing - Lizhe Tan 2013-01-21
Digital Signal Processing, Second Edition enables electrical engineers and technicians in the fields of biomedical, computer, and electronics engineering to master the essential fundamentals of DSP principles and practice. Many instructive worked examples are used to illustrate the material, and the use of mathematics is minimized for easier grasp of concepts. As such, this title is also useful to undergraduates in electrical engineering, and as a reference for science students and practicing engineers. The book goes beyond DSP theory, to show implementation of algorithms in hardware and software. Additional topics covered include adaptive filtering with noise reduction and echo cancellations, speech compression, signal sampling, digital filter realizations, filter design, multimedia applications, over-sampling, etc. More advanced topics are also covered, such as adaptive filters, speech compression such as PCM, u-law, ADPCM, and multi-rate DSP and over-sampling ADC. New to this edition: MATLAB projects dealing with practical applications added throughout the book New chapter (chapter 13) covering sub-band coding and wavelet transforms, methods that have become popular in the DSP field New applications included in many chapters, including applications of DFT to seismic signals, electrocardiography data, and vibration signals All real-time C programs revised for the TMS320C6713 DSK Covers DSP principles with emphasis on communications and control applications Chapter objectives, worked examples, and end-of-chapter exercises aid the reader in grasping key concepts and solving related problems Website with MATLAB programs for simulation and C programs for real-time DSP
Adaptive Filters - Behrouz Farhang-Boroujeny 2013-04-02
This second edition of Adaptive Filters: Theory andApplications has been updated throughout to reflect the latestdevelopments in this field; notably an increased coverage given tothe practical applications of the theory to illustrate the muchbroader range of adaptive filters applications developed in recentyears. The book offers an easy to understand approach to the theoryand application of adaptive filters by clearly illustrating how thetheory explained in the early chapters of the book is modified forthe various applications discussed in detail in later chapters.This integrated approach makes the book a valuable resource forgraduate students; and the inclusion of more advanced applicationsincluding antenna arrays and wireless communications makes it asuitable technical reference for engineers, practitioners andresearchers. Key features: • Offers a thorough treatment of the theory of adaptivesignal processing; incorporating new material on transform domain,frequency domain, subband adaptive filters, acoustic echocancellation and active noise control. • Provides an in-depth study of applications which nowincludes extensive coverage of OFDM, MIMO and smart antennas. • Contains exercises and computer simulation problems atthe end of each chapter. • Includes a new companion website hosting MATLAB®simulation programs which complement the theoretical analyses,enabling the reader to gain an in-depth understanding of thebehaviours and properties of the various adaptive algorithms.
Audio Signal Processing and Coding - Andreas Spanias 2006-09-11
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
Adaptive Blind Signal and Image Processing - Andrzej Cichocki 2002-06-14

With solid theoretical foundations and numerous potential applications, Blind Signal Processing (BSP) is one of the hottest emerging areas in Signal Processing. This volume unifies and extends the theories of adaptive blind signal and image processing and provides practical and efficient algorithms for blind source separation: Independent, Principal, Minor Component Analysis, and Multichannel Blind Deconvolution (MBD) and Equalization. Containing over 1400 references and mathematical expressions Adaptive Blind Signal and Image Processing delivers an unprecedented collection of useful techniques for adaptive blind signal/image separation, extraction, decomposition and filtering of multi-variable signals and data. Offers a broad coverage of blind signal processing techniques and algorithms both from a theoretical and practical point of view Presents more than 50 simple algorithms that can be easily modified to suit the reader's specific real world problems Provides a guide to fundamental mathematics of multi-input, multi-output and multi-sensory systems Includes illustrative worked examples, computer simulations, tables, detailed graphs and conceptual models within self contained chapters to assist self study Accompanying CD-ROM features an electronic, interactive version of the book with fully coloured figures and text. C and MATLAB user-friendly software packages are also provided MATLAB is a registered trademark of The MathWorks, Inc. By providing a detailed introduction to BSP, as well as presenting new results and recent developments, this informative and inspiring work will appeal to researchers, postgraduate students, engineers and scientists working in biomedical engineering, communications, electronics, computer science, optimisations, finance, geophysics and neural networks.

Multirate Systems: Design and Applications - Jovanovic-Dolecek, Gordana 2001-07-01

Digital signal processing is an area of science and engineering that has been developed rapidly over the past years. This rapid development is the result of the significant advances in digital computer technology and integrated circuits fabrication. Many of the signal processing tasks conventionally performed by analog means are realized today by less expensive and often more reliable digital hardware. Multirate Systems: Design and Applications addresses the rapid development of multirate digital signal processing and how it is complemented by the emergence of new applications.

Real-Time Digital Signal Processing - Sen M. Kuo 2006-05-01

Real-time Digital Signal Processing: Implementations and Applications has been completely updated and revised for the 2nd edition and remains the only book on DSP to provide an overview of DSP theory and programming with hands-on experiments using MATLAB, C and the newest fixed-point processors from Texas Instruments (TI).

Multirate Filtering for Digital Signal Processing: MATLAB Applications - Milic, Ljiljana 2009-01-31

"This book covers basic and the advanced approaches in the design and implementation of multirate filtering"--Provided by publisher.

Fundamentals of Signals and Systems Using MATLAB - Edward W. Kamen 1997

This text presents an accessible yet comprehensive analytical treatment of signals and systems, and also incorporates a strong emphasis on solving problems and exploring concepts using MATLAB

Adapted Wavelet Analysis - Mladen Victor Wickerhauser 1996-04-17

This detail-oriented text is intended for engineers and applied mathematicians who must write computer programs to perform wavelet and related analysis on real data. It contains an overview of mathematical prerequisites and proceeds to describe hands-on programming techniques to implement special programs for signal analysis and other applications. From the table of contents: - Mathematical Preliminaries - Programming Techniques - The Discrete Fourier Transform - Local Trigonometric Transforms - Quadrature Filters - The Discrete Wavelet Transform - Wavelet Packets - The Best Basis Algorithm - Multidimensional Library Trees - Time-Frequency Analysis - Some Applications - Solutions to Some of the Exercises - List of Symbols - Quadrature Filter Coefficients

Fast Fourier Transform - Algorithms and Applications - K.R. Rao 2011-02-21

This book presents an introduction to the principles of the fast Fourier transform. This book covers FFTs, frequency domain filtering, and applications to video and audio signal processing. As fields like communications, speech and image processing, and related areas are rapidly developing, the FFT as one of essential parts in digital signal processing has been widely used. Thus there is a pressing need from instructors and students for a book dealing with the latest FFT topics. This book provides thorough and detailed explanation of important or up-to-date FFTs. It also has adopted modern approaches like MATLAB examples and projects for better understanding of diverse FFTs.

Digital Signal Processing Using MATLAB for Students and Researchers - John W. Leis 2011-10-14

Quickly Engages in Applying Algorithmic Techniques to Solve Practical Signal Processing Problems With its active, hands-on learning approach, this text enables readers to master the underlying principles of digital signal processing and its many applications in industries such as digital television, mobile and broadband communications, and medical/scientific devices. Carefully developed MATLAB® examples throughout the text illustrate the mathematical concepts and use of digital signal processing algorithms. Readers will develop a deeper understanding of how to apply the algorithms by manipulating the codes in the examples to see their effect. Moreover, plenty of exercises help to put knowledge into practice solving real-world signal processing challenges. Following an introductory chapter, the text explores: Sampled signals and digital processing Random signals Representing signals and systems Temporal and spatial signal processing Frequency analysis of signals Discrete-time filters and recursive filters Each chapter begins with chapter objectives and an introduction. A summary at the end of each chapter ensures that one has mastered all the key concepts and techniques before progressing in the text. Lastly, appendices listing selected web resources, research papers, and related textbooks enable the investigation of individual topics in greater depth. Upon completion of this text, readers will understand how to apply key algorithmic techniques to address practical signal processing problems as well as develop their own signal processing algorithms. Moreover, the text provides a solid foundation for evaluating and applying new digital processing signal techniques as they are developed.

Wavelets and Subband Coding - Jelena Kovacevic 1995

A central goal of signal processing is to describe real-time signals, be it for computation, compression, or understanding. This book presents a unified view of wavelets and subband coding with a signal processing perspective. Covers the discrete-time case, or filter banks; development of wavelets; continuous wavelet and local Fourier transforms; efficient algorithms for filter banks and wavelet computations; and signal compression. *provides broad coverage of theory and applications and a different perspective based on signal processing. *gives framework for applications in speech, audio, image and video compression as used in multimedia. *includes sufficient background material so that people without signal processing knowledge will find it useful.

Real-Time Digital Signal Processing - Sen M. Kuo 2013-08-05

Combines both the DSP principles and real-timeimplementations and applications, and now updated with the neweZdsp USB Stick, which is very low cost, portable and widelyemployed at many DSP labs. Real-Time Digital Signal Processing introducesfundamental digital signal processing (DSP) principles and will beupdated to include the latest DSP applications, introduce newsoftware development tools and adjust the software design processto reflect the latest advances in the field. In the 3rdedition of the book, the key aspect of hands-on experiments will beenhanced to make the DSP principles more interesting and directlyinteract with the real-world applications. All of the programs willbe carefully updated using the most recent version of softwaredevelopment tools and the new TMS320VC5505 eZdsp USB Stick forreal-time experiments. Due to its lower cost and portability, thenew software and hardware tools are now widely used in universitylabs and in commercial industrial companies to replace the olderand more expensive generation. The new edition will have a renewedfocus on real-time applications and will offer step-by-stephands-on experiments for a complete design cycle starting fromfloating-point C language program to fixed-point C implementation,code optimization using INTRINSICS, and mixed C-and-assemblyprogramming on fixed-point DSP processors. This new methodologyenables readers to concentrate on learning DSP fundamentals andinnovative applications by relaxing the intensive programmingefforts, namely, the traditional DSP assembly coding efforts. Thebook is organized into two parts; Part One introduces thedigital signal processing principles and theories, and PartTwo focuses on practical applications. The topics for theapplications are the extensions of the theories in Part One with anemphasis placed on the hands-on experiments, systematic design andimplementation approaches. The applications provided in the bookare carefully chosen to reflect current advances of DSP that are ofmost relevance for the intended readership. Combines both the DSP principles and real-time implementationsand applications using the new eZdsp USB Stick, which is very lowcost, portable and widely employed at many DSP labs is now used inthe new edition Places renewed emphasis on C-code experiments and reduces theexercises using assembly coding; effective use of C programming,fixed-point C code and INTRINSICS will become the main focus of thenew edition. Updates to application areas to reflect latest advances such asspeech coding techniques used for next generation networks (NGN),audio coding with surrounding sound, wideband speech codec (ITUG.722.2 Standard), fingerprint for image processing, and biomedicalsignal processing examples. Contains new addition of several projects that can be used assemester projects; as well as new many new real-time experimentsusing TI's binary libraries – the experiments areprepared with flexible interface and modular for readers to adaptand modify to create other useful applications from the providedbasic programs. Consists of more MATLAB experiments, such as filter design,algorithm evaluation, proto-typing for C-code architecture, andsimulations to aid readers to learn DSP fundamentals. Includes supplementary material of program and data files forexamples, applications, and experiments hosted on a companionwebsite. A valuable resource for Postgraduate students enrolled on DSPcourses focused on DSP implementation & applications as well asSenior undergraduates studying DSP; engineers and programmers whoneed to learn and use DSP principles and development tools fortheir projects.

Advances in Noise Analysis, Mitigation and Control - Noor Ahmed 2016-10-05

The adverse impacts from excess noise on human health and daily activities have accelerated at an alarming rate over the last few decades. This has prompted significant research into noise attenuation and mitigation of these unwanted effects. This book is a collection of works from eminent researchers from around the world, who address the aforementioned issues. It provides the most up-to-date information on current work being conducted in the field of noise pollution and is of value to a wide range of students, engineers, scientists and industry consultants who wish to further understand current methodologies and emerging concepts.

Microphone Arrays - Michael Brandstein 2013-04-17

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.

Subband Adaptive Filtering - Kong-Aik Lee 2009-07-06

Subband adaptive filtering is rapidly becoming one of the most effective techniques for reducing computational complexity and improving the convergence rate of algorithms in adaptive signal processing applications. This book provides an introductory, yet extensive guide on the theory of various subband adaptive filtering techniques. For beginners, the authors discuss the basic principles that underlie the design and implementation of subband adaptive filters. For advanced readers, a comprehensive coverage of recent developments, such as multiband tap–weight adaptation, delayless architectures, and filter–bank design methods for reducing band–edge effects are included. Several analysis techniques and complexity evaluation are also introduced in this book to provide better understanding of subband adaptive filtering. This book bridges the gaps between the mixed–domain natures of subband adaptive filtering techniques and provides enough depth to the material augmented by many MATLAB® functions and examples. Key Features: Acts as a timely introduction for researchers, graduate students and engineers who want to design and deploy subband adaptive filters in their research and applications. Bridges the gaps between two distinct domains: adaptive filter theory and multirate signal processing. Uses a practical approach through MATLAB®-based source programs on the accompanying CD. Includes more than 100 M-files, allowing readers to modify the code for different algorithms and applications and to gain more insight into the theory and concepts of subband adaptive filters. Subband Adaptive Filtering is aimed primarily at practicing engineers, as well as senior undergraduate and graduate students. It will also be of interest to researchers, technical managers, and computer scientists.

Wavelets and Subbands - Agostino Abbate 2012-12-06

This book presents connections between the different aspects of wavelet and subband theory.

Bayesian Adaptive Methods for Clinical Trials - Scott M. Berry 2010-07-19

Already popular in the analysis of medical device trials, adaptive Bayesian designs are increasingly being used in drug development for a wide variety of diseases and conditions, from Alzheimer's disease and multiple sclerosis to obesity, diabetes, hepatitis C, and HIV. Written by leading pioneers of Bayesian clinical trial designs, Bayesian Adaptive Methods for Clinical Trials explores the growing role of Bayesian thinking in the rapidly changing world of clinical trial analysis. The book first summarizes the current state of clinical trial design and analysis and introduces the main ideas and potential benefits of a Bayesian alternative. It then gives an overview of basic Bayesian methodological and computational tools needed for Bayesian clinical trials. With a focus on Bayesian designs that achieve good power and Type I error, the next chapters present Bayesian tools useful in early (Phase I) and middle (Phase II) clinical trials as well as two recent Bayesian adaptive Phase II studies: the BATTLE and ISPY-2 trials. In the following chapter on late (Phase III) studies, the authors emphasize modern adaptive methods and seamless Phase II–III trials for maximizing information usage and minimizing trial duration. They also describe a case study of a recently approved medical device to treat atrial fibrillation. The concluding chapter covers key special topics, such as the proper use of historical data, equivalence studies, and subgroup analysis. For readers involved in clinical trials research, this book significantly updates and expands their statistical toolkits. The authors provide many detailed examples drawing on real data sets. The R and WinBUGS codes used throughout are available on supporting websites. Scott Berry talks about the book on the CRC Press YouTube Channel.

Adaptive Filters - Ali H. Sayed 2011-10-11

Adaptive filtering is a topic of immense practical and theoretical value, having applications in areas ranging from digital and wireless communications to biomedical systems. This book enables readers to gain a gradual and solid introduction to the subject, its applications to a variety of topical problems, existing limitations, and extensions of current theories. The book consists of eleven parts?each part containing a series of focused lectures and ending with bibliographic comments, problems, and computer projects with MATLAB solutions.

Signal Processing and Information Technology - Vinu V Das 2012-07-25

This book constitutes the thoroughly refereed post-conference proceedings of the First International Joint Conference on Advances in Signal Processing and Information Technology (SPIT 2011) and Recent Trends in Information Processing and Computing (IPC 2011) held in Amsterdam, The Netherlands, in December 2011. The 50 revised full papers presented were carefully selected from 298 submissions. Conference papers promote research and development activities in computer science, information technology, computational engineering, image and signal processing, and communication.

Adaptive Filter Theory - Simon S. Haykin 1996

Haykin examines both the mathematical theory behind various linear adaptive filters with finite-duration impulse response (FIR) and the elements of supervised neural networks. This edition has been updated and refined to keep current with the field and develop concepts in as unified and accessible a manner as possible. It: introduces a completely new chapter on Frequency-Domain Adaptive Filters; adds a chapter on Tracking Time-Varying Systems; adds two chapters on Neural Networks; enhances material on RLS algorithms; strengthens linkages to Kalman filter theory to gain a more unified treatment of the standard, square-root and order-recursive forms; and includes new computer experiments using MATLAB software that illustrate the underlying theory and applications of the LMS and RLS algorithms.

Digital Signal and Image Processing - Tamal Bose 2004

Introducing the first text to integrate the topics of digital signal processing (DSP), digital image processing (DIP), and adaptive signal processing (ASP)! Digital Signal and Image Processing helps students develop a well-rounded understanding of these key areas by focusing on fundamental concepts, mathematical foundations, and advanced algorithms. The presentation is mathematically thorough with clear explanations, numerous examples, illustrations, and applications. In addition to problems, MATLAB-based computer projects are assigned at the end of each chapter, making this book ideal for laboratory-based courses.

