Adaptive Filters Prentice Hall Signal Processing Series

This book is based on a graduate level course offered by the author at UCLA and has been classed tested there and at other universities over a number of years. This will be the most comprehensive book on the market today providing instructors a wide choice in designing their courses. * Offers computer problems to illustrate real life applications for students and professionals alike * An Instructor's Manual presenting detailed solutions to all the problems in the book is available from the Wiley editorial department. An Instructor's Manual presenting detailed solutions to all the problems in the book is available from the Wiley editorial department.

The aim of this book is to provide an overview of recent developments in Kalman filter theory and their applications in engineering and scientific fields. The book is divided into 24 chapters and organized in five blocks corresponding to recent advances in Kalman filtering theory, applications in medical and biological sciences, tracking and positioning systems, electrical engineering and, finally, industrial processes and communication networks.

This encyclopaedia covers Characterization Hierarchy Containing Augmented Characterizations to Video Compression.

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.

A comprehensive and practical treatment of adaptive signal processing featuring frequent use of examples.
This book is an accessible guide to adaptive signal processing methods that equips the reader with advanced theoretical and practical tools for the study and development of circuit structures and provides robust algorithms relevant to a wide variety of application scenarios. Examples include multimodal and multimedia communications, the biological and biomedical fields, economic models, environmental sciences, acoustics, telecommunications, remote sensing, monitoring and in general, the modeling and prediction of complex physical phenomena. The reader will learn not only how to design and implement the algorithms but also how to evaluate their performance for specific applications utilizing the tools provided. While using a simple mathematical language, the employed approach is very rigorous. The text will be of value both for research purposes and for courses of study.
This book is Volume IV of the series DSP for MATLABâ„¢ and LabVIEWâ„¢. Volume IV is an introductory treatment of LMS Adaptive Filtering and applications, and covers cost functions, performance surfaces, coefficient perturbation to estimate the gradient, the LMS algorithm, response of the LMS algorithm to narrow-band signals, and various topologies such as ANC (Active Noise Cancelling) or system modeling, Noise Cancellation, Interference Cancellation, Echo Cancellation (with single- and dual-H topologies), and Inverse Filtering/Deconvolution. The entire series consists of four volumes that collectively cover basic digital signal processing in a practical and accessible manner, but which nonetheless include all essential foundation mathematics. As the series title implies, the scripts (of which there are more than 200) described in the text and supplied in code form (available via the internet at www.morganclaypool.com/page/isen) will run on both MATLABâ„¢ and LabVIEWâ„¢. The text for all volumes contains many examples, and many useful computational scripts, augmented by demonstration scripts and LabVIEWâ„¢ Virtual Instruments (VIs) that can be run to illustrate various signal processing concepts graphically on the user's computer screen. Volume I consists of four chapters that collectively set forth a brief overview of the field of digital signal processing, useful signals and concepts (including convolution, recursion, difference equations, LTI systems, etc), conversion from the continuous to discrete domain and back (i.e., analog-to-digital and digital-to-analog conversion), aliasing, the Nyquist rate, normalized frequency, sample rate conversion and Mu-law compression, and signal processing principles including correlation, the correlation sequence, the Real DFT, correlation by convolution, matched filtering, simple FIR filters, and simple IIR filters. Chapter 4 of Volume I, in particular, provides an intuitive or "first principle" understanding of how digital filtering and frequency transforms work. Volume II provides detailed coverage of discrete frequency transforms, including a brief overview of common frequency transforms, both discrete and continuous, followed by detailed treatments of the Discrete Time Fourier Transform (DTFT), the z-Transform (including definition and

properties, the inverse z-transform, frequency response via z-transform, and alternate filter realization topologies including Direct Form, Direct Form Transposed, Cascade Form, Parallel Form, and Lattice Form), and the Discrete Fourier Transform (DFT) (including Discrete Fourier Series, the DFT-IDFT pair, DFT of common signals, bin width, sampling duration, and sample rate, the FFT, the Goertzel Algorithm, Linear, Periodic, and Circular convolution, DFT Leakage, and computation of the Inverse DFT). Volume III covers digital filter design, including the specific topics of FIR design via windowed-ideal-lowpass filter, FIR highpass, bandpass, and bandstop filter design from windowed-ideal lowpass filters, FIR design using the transition-band-optimized Frequency Sampling technique (implemented by Inverse-DFT or Cosine/Sine Summation Formulas), design of equiripple FIRs of all standard types including Hilbert Transformers and Differentiators via the Remez Exchange Algorithm, design of Butterworth, Chebyshev (Types I and II), and Elliptic analog prototype lowpass filters, conversion of analog lowpass prototype filters to highpass, bandpass, and bandstop filters, and conversion of analog filters to digital filters using the Impulse Invariance and Bilinear Transform techniques. Certain filter topologies specific to FIRs are also discussed, as are two simple FIR types, the Comb and Moving Average filters.

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. An instructor's manual, a set of master transparencies, and the MATLAB codes for all of the algorithms described in the text are also available. Useful to both professional researchers and students, the text includes 185 problems; over 38 examples, and over 130 illustrations. It is of primary interest to those working in signal processing, communications, and circuits and systems. It will also be of interest to those working in power systems, networks, learning systems, and intelligent systems.

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

different echo cancellation scenarios. --Book Jacket.

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.

Interest in filter theory and design has been growing with the telecommunications industry since the late nineteenth century. Now that telecommunications has become so critical to industry, filter research has assumed even greater importance at companies and academic institutions around the world. The CRC Handbook of Electrical Filters fills in the gaps for engineers and scientists who need a basic introduction to the subject. Unlike the currently available textbooks, which are filled with detailed, highly technical analysis geared to the specialist, this practical guide provides useful information for the non-specialist about the various types of filters, their design, and applications. The handbook covers approximation theory and methods and introduces CAD packages that perform approximation and synthesis for both analog and digital filters. Also included are design methods for LCR, active-RC, digital, mechanical, and switched capacitor (SC) filters. A thorough survey of current design trends rounds out this complete assessment of a key field of study.

Nonlinear signal and image processing methods are fast emerging as an alternative to established linear methods for meeting the challenges of increasingly sophisticated applications. Advances in computing performance and nonlinear theory are making nonlinear techniques not only viable, but practical. This book details recent advances in nonl

I feel very honoured to have been asked to write a brief foreword for this book on QRD-RLS Adaptive Filtering–asubjectwhichhas been close to my heart for many years. The book is well written and very timely – I look forward personally to seeing it in print. The editor is to be congratulated on assembling such a highly esteemed team of contributing authors able to span the broad range of topics and concepts which underpin this subject. In many respects, and for reasons well expounded by the authors, the LMS al- rithm has reigned supreme since its inception, as the algorithm of choice for prac- cal applications of adaptive ltering. However, as a result of the relentless advances in electronic technology, the demand for stable and ef cient RLS algorithms is growing rapidly – not just because the higher computational load is no longer such a serious barrier, but also because the technological pull has grown much stronger in the modern commercial world of 3G mobile communications, cognitive radio, high speed imagery, and so on.

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

This book presents a mechatronic approach to Active Noise Control (ANC). It describes the required elements of system theory, engineering acoustics, electroacoustics and adaptive signal processing in a comprehensive, consistent and systematic manner using a unified notation. Furthermore, it includes a design methodology for ANC-systems, explains its application and describes tools to be used for ANC-system design. From the research point of view, the book presents new approaches to sound source localization in weakly damped interiors. One is based on the inverse finite element method, the other is based on a sound intensity probe with an active free field. Furthermore, a prototype of an ANC-system able to reach the physical limits of local (feed-forward) ANC is described. This is one example for applied research in ANC-system design. Other examples are given for (i) local ANC in a semi-enclosed subspace of an aircraft cargo hold and (ii) for the combination of audio entertainment with ANC.

This unified survey focuses on linear discrete-time systems and explores natural extensions to nonlinear systems. It emphasizes discrete-time systems, summarizing theoretical and practical aspects of a large class of adaptive algorithms. 1984 edition.

Because of the wide use of adaptive filtering in digital signal processing and, because most of the modern electronic devices include some type of an adaptive filter, a text that brings forth the fundamentals of this field was necessary. The material and the principles presented in this book are easily accessible to engineers, scientists, and students who would like to learn the fundamentals of this field and have a background at the bachelor level. Adaptive Filtering Primer with MATLAB® clearly explains the fundamentals of adaptive filtering supported by numerous examples and computer simulations. The authors introduce discrete-time signal processing, random variables and stochastic processes, the Wiener filter, properties of the error surface, the steepest descent method, and the least mean square (LMS) algorithm. They also supply many MATLAB® functions and m-files along with computer experiments to illustrate how to apply the concepts to real-world problems. The book includes problems along with hints, suggestions, and solutions for solving them. An appendix on matrix computations completes the self-contained coverage. With applications across a wide range of areas, including radar, communications, control, medical instrumentation, and seismology, Adaptive Filtering Primer with MATLAB® is an ideal companion for quick reference and a perfect, concise introduction to the field.

This book provides an excellent reference for all professionals working in the area of array signal processing and its applications in wireless communications. Wideband beamforming has advanced with the increasing bandwidth in wireless communications and the development of ultra wideband (UWB) technology. In this book, the authors address the fundamentals and most recent developments in the field of wideband beamforming. The book provides a thorough coverage of the subject including major sub-areas such as sub-band adaptive beamforming, frequency invariant beamforming, blind wideband beamforming, beamforming without temporal processing, and beamforming for multi-path signals. Key Features: Unique book focusing on wideband beamforming Discusses a hot topic coinciding with the increasing bandwidth in wireless communications and the development of UWB technology Addresses the general concept of beamforming including fixed beamformers and adaptive beamformers Covers advanced topics including sub-band adaptive beamforming, frequency invariant beamforming, blind wideband beamforming, beamforming without temporal processing, and beamforming for multi-path signals Includes various design examples and corresponding complexity analyses This book provides a reference for engineers and researchers in wireless communications and signal processing fields. Postgraduate students studying signal processing will also find this book of interest.

The work presented in this text relates to research work in the general area of adaptive filter theory and practice which has been carried out at the Department of Electrical Engineering, University of Edinburgh since 1977. Much of the earlier work in the department was devoted to looking at the problems associated with the physical implementation of these structures. This text relates to research which has been undertaken since 1984 which is more involved with the theoretical development of adaptive algorithms. The text sets out to provide a coherent framework within which general adaptive algorithms for finite impulse response adaptive filters may be evaluated. It further presents one approach to the problem of finding a stable solution to the infinite impulse response adaptive filter problem. This latter objective being restricted to the communications equaliser application area. The authors are indebted to a great number of people for their help, guidance and encouragement during the course of preparing this text. We should first express our appreciation for the support given by two successive heads of department at Edinburgh, Professor J. H. Collins and Professor J. Mavor. The work reported here could not have taken place without their support and also that of many colleagues, principally Professor P. M. Grant who must share much of the responsibility for instigating this line of research at Edinburgh.

Signal processing plays an increasingly central role in the development of modern telecommunication and information processing systems, with a wide range of applications in areas such as multimedia technology, audio-visual signal processing, cellular mobile communication, radar systems and financial data forecasting. The theory and application of signal processing deals with the identification, modelling and utilisation of patterns and structures in a signal process. The observation signals are often distorted, incomplete and noisy and hence, noise reduction and the removal of channel distortion is an important part of a signal processing system. Advanced Digital Signal Processing and Noise Reduction, Third Edition, provides a fully updated and structured presentation of the theory and applications of statistical signal processing and noise reduction methods. Noise is the eternal bane of communications engineers, who are always striving to find new ways to improve the signal-to-noise ratio in communications systems and this resource will help them with this task. * Features two new chapters on Noise, Distortion and Diversity in Mobile Environments and Noise Reduction Methods for Speech Enhancement over Noisy Mobile Devices. * Topics discussed include: probability theory, Bayesian estimation and classification, hidden Markov models, adaptive filters, multi-band linear prediction, spectral estimation, and impulsive and transient noise removal. * Explores practical solutions to interpolation of missing signals, echo cancellation, impulsive and transient noise removal, channel equalisation, HMM-based signal and noise decomposition. This is an invaluable text for senior undergraduates, postgraduates and researchers in the fields of digital signal processing, telecommunications and statistical data analysis. It will also appeal to engineers in telecommunications and audio and signal processing industries.

A practical and fascinating book on a topic at the forefront of communications technology. Field-Programmable Gate Arrays (FPGAs) are on the verge of revolutionizing digital signal processing. Novel FPGA families are replacing ASICs and PDSPs for front-end digital signal processing algorithms at an accelerating rate. The efficient implementation of these algorithms is the main goal of this book. It starts with an overview of today's FPGA technology, devices, and tools for designing state-of-the-art DSP systems. Each of the book’s chapter contains exercises. The VERILOG source code and a glossary are given in the appendices.

This book presents the basic concepts of adaptive signal processing and adaptive filtering in a concise and straightforward manner, using clear notations that facilitate actual implementation. Important algorithms are described in detailed tables which allow the reader to verify learned concepts. The book covers the family of LMS and algorithms as well as set-membership, sub-band, blind, IIR adaptive filtering, and more. The book is also supported by a web page maintained by the author.

This three-volume proceedings contains revised selected papers from the Second International Conference on Artificial Intelligence and Computational Intelligence, AICI 2011, held in Taiyuan, China, in September 2011. The total of 265 high-quality papers presented were carefully reviewed and selected from 1073 submissions. The topics of Part I covered are: applications of artificial intelligence; applications of computational intelligence; automated problem solving; biomedical inforamtics and computation; brain models/cognitive science; data mining and knowledge discovering; distributed AI and agents; evolutionary programming; expert and decision support systems; fuzzy computation; fuzzy logic and soft computing; and genetic algorithms.

In this book, the authors provide insights into the basics of adaptive filtering, which are particularly useful for students taking their first steps into this field. They start by studying the problem of minimum mean-square-error filtering, i.e., Wiener filtering. Then, they analyze iterative methods for solving the optimization problem, e.g., the Method of Steepest Descent. By proposing stochastic approximations, several basic adaptive algorithms are derived, including Least Mean Squares (LMS), Normalized Least Mean Squares (NLMS) and Sign-error algorithms. The authors provide a general framework to study the stability and steady-state performance of these algorithms. The affine Projection Algorithm (APA) which provides faster convergence at the expense of computational complexity (although fast implementations can be used) is also presented. In addition, the Least Squares (LS) method and its recursive version (RLS), including fast implementations are discussed. The book closes with the discussion of several topics of interest in the adaptive filtering field.

Rather than superficially examining an extensive list of possible applications benefiting from adaptive filter use, the authors examine four such problems in detail and review the common attributes that are shared with many other applications of adaptive filtering. The authors develop the basic rules and algorithms for filter performance and provide tools for design, along with an appreciation of the complexity of behavioral analysis. Derivations and convergence discussions are kept to a basic level. The presentation focuses on a few principles and applies them to a series of motivating examples, that include in-depth discussion of implementation aspects for filter design not found in other books. Serves as a valuable reference for practicing engineers.

The Springer Handbook of Bio-/Neuro-Informatics is the first published book in one volume that explains together the basics and the state-of-the-art of two major science disciplines in their interaction and mutual relationship, namely: information sciences, bioinformatics and neuroinformatics. Bioinformatics is the area of science which is concerned with the information processes in biology and the development and applications of methods, tools and systems for storing and processing of biological information thus facilitating new knowledge discovery. Neuroinformatics is the area of science which is concerned with the information processes in biology and the development and applications of methods, tools and systems for storing and processing of biological information thus facilitating new knowledge discovery. The text contains 62 chapters organized in 12 parts, 6 of them covering topics from information science and bioinformatics, and 6 cover topics from information science and neuroinformatics. Each chapter consists of three main sections: introduction to the subject area, presentation of methods and advanced and future developments. The Springer Handbook of Bio-/Neuroinformatics can be used as both a textbook and as a reference for postgraduate study and advanced research in these areas. The target audience includes students, scientists, and practitioners from the areas of information, biological and neurosciences. With Forewords by Shun-ichi Amari of the Brain Science Institute, RIKEN, Saitama and Karlheinz Meier of the University of Heidelberg, Kirchhoff-Institute of Physics and Co-Director of the Human Brain Project.

This text emphasizes the intricate relationship between adaptive filtering and signal analysis - highlighting stochastic processes, signal representations and properties, analytical tools, and implementation methods. This second edition includes new chapters on adaptive techniques in communications and rotation-based algorithms. It provides practical applications in information, estimation, and circuit theories. Engineers in all fields will appreciate a practical guide that combines several new effective MATLAB® problem-solving approaches and the very latest in discrete random signal processing and filtering. Numerous Useful Examples, Problems, and Solutions – An Extensive and Powerful Review Written for practicing engineers seeking to strengthen their practical grasp of random signal processing, Discrete Random Signal Processing and Filtering Primer with MATLAB provides the opportunity to doubly enhance their skills. The author, a leading expert in the field of electrical and computer engineering, offers a solid review of recent developments in discrete signal processing. The book also details the latest progress in the revolutionary MATLAB language. A Practical Self-Tutorial That Transcends Theory The author introduces an incremental discussion of signal processing and filtering, and presents several new methods that can be used for a more dynamic analysis of random digital signals with both linear and non-linear filtering. Ideal as a self-tutorial, this book includes numerous examples and functions, which can be used to select parameters, perform simulations, and analyze results. This concise guide encourages readers to use MATLAB functions – and those new ones introduced as Book MATLAB Functions – to substitute many different combinations of parameters, giving them a firm grasp of how much each parameter affects results. Much more than a simple review of theory, this book emphasizes problem solving and result analysis, enabling readers to take a hands-on approach to advance their own understanding of MATLAB and the way it is used within signal processing and filtering.

The TMS320C6x is Texas Instrument's next generation DSP found in over 60 percent of wireless devices from leading manufacturers such as Ericsson, Nokia, Sony, and Handspring Author has many years experience working with the TI line of TMS DSPs and his books are based on courses and seminars given at TI sponsored meetings All programs listed in the text will be available on the Wiley FTP site In addition to its wireless applications, the TMS DSP is tailored to enable a new generation of Internet media entertainment appliances

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.

A best-seller in its print version, this comprehensive CD-ROM reference contains unique, fully searchable coverage of all major topics in digital signal processing (DSP), establishing an invaluable, time-saving resource for the engineering community. Its unique and broad scope includes contributions from all DSP specialties, including: telecommunications, computer engineering, acoustics, seismic data analysis, DSP software and hardware, image and video processing, remote sensing, multimedia applications, medical technology, radar and sonar applications

ICA3PP 2000 was an important conference that brought together researchers and practitioners from academia, industry and governments to advance the knowledge of parallel and distributed computing. The proceedings constitute a well-defined set of innovative research papers in two broad areas of parallel and distributed computing: (1) architectures, algorithms and networks; (2) systems and applications. Contents:Cluster ComputingInterconnection Networks and RoutingParallel Architecture & Parallel I/O SystemsParallel and Distributed DatabasesParallel Algorithms ITools and Environments for Parallel and Distributed Software DevelopmentParallel Algorithms IIParallel Processing on Web-Based Systems and ApplicationsDistributed and Parallel Operating Systems and MiddlewareHigh-Performance Scientific ComputingParallel and Distributed ProcessingFault-Tolerant ComputingHigh-Performance Data Management Readership: Researchers, graduate students, academics and practitioners in computing. Keywords:

“Adaptive Digital Filters” presents an important discipline applied to the domain of speech processing. The book first makes the reader acquainted with the basic terms of filtering and adaptive filtering, before introducing the field of advanced modern algorithms, some of which are contributed by the authors themselves. Working in the field of adaptive signal processing requires the use of complex mathematical tools. The book offers a detailed presentation of the mathematical models that is clear and consistent, an approach that allows everyone with a college level of mathematics knowledge to successfully follow the mathematical derivations and descriptions of algorithms. The algorithms are presented in flow charts, which facilitates their practical implementation. The book presents many experimental results and treats the aspects of practical application of adaptive filtering in real systems, making it a valuable resource for both undergraduate and graduate students, and for all others interested in mastering this important field.

Teaches students about classical and nonclassical adaptive systems within one pair of covers Helps tutors with time-saving course plans, ready-made practical assignments and examination guidance The recently developed "practical sub-space adaptive filter" allows the reader to combine any set of classical and/or non-classical adaptive systems to form a powerful technology for solving complex nonlinear problems

This book was written in response to the growing demand for a text that provides a unified treatment of linear and nonlinear complex valued adaptive filters, and methods for the processing of general complex signals (circular and noncircular). It brings together adaptive filtering algorithms for feedforward (transversal) and feedback architectures and the recent developments in the statistics of complex variable, under the powerful frameworks of CR (Wirtinger) calculus and augmented complex statistics. This offers a number of theoretical performance gains, which is illustrated on both stochastic gradient algorithms, such as the augmented complex least mean square (ACLMS), and those based on Kalman filters. This work is supported by a number of simulations using synthetic and real world data, including the noncircular and intermittent radar and wind signals.

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.

Adaptive FiltersPrentice HallAdaptive FilteringAlgorithms and Practical ImplementationSpringer Science & Business Media

Adaptive filtering can be used to characterize unknown systems in time-variant environments. The main objective of this approach is to meet a difficult comprise: maximum convergence speed with maximum accuracy. Each application requires a certain approach which determines the filter structure, the cost function to minimize the estimation error, the adaptive algorithm, and other parameters; and each selection involves certain cost in computational terms, that in any case should consume less time than the time required by the application working in real-time. Theory and application are not, therefore, isolated entities but an imbricated whole that requires a holistic vision. This book collects some theoretical approaches and practical applications in different areas that support expanding of adaptive systems.