Adaptive Filters Structures Algorithms And Applications The Springer International Series In Engineering And Computer Science

Adaptive filtering is a branch of digital signal processing which enables the selective enhancement of desired elements of a signal and the reduction of undesired elements. Change detection is another kind of adaptive filtering for non-stationary signals, and is the basic tool in fault detection and diagnosis. This text takes the unique approach that change detection is a natural extension of adaptive filtering, and the broad coverage encompasses both the mathematical tools needed for adaptive filtering and change detection and the applications of the technology. Real engineering applications covered include aircraft, automotive, communication systems, signal processing and automatic control problems. The unique integration of both theory and practical applications makes this book a valuable resource combining information otherwise only available in separate sources Comprehensive coverage includes many examples and case studies to illustrate the ideas and show what can be achieved Uniquely integrates applications to airborne, automotive and communications systems with the essential mathematical tools Accompanying Matlab toolbox available on the web illustrating the main ideas and enabling the reader to do simulations using all the figures and numerical examples featured This text would prove to be an essential reference for postgraduates and researchers studying digital signal processing as well as practising digital signal processing engineers.

Today's embedded and real-time systems contain a mix of processor types: off-the-shelf microcontrollers, digital signal processors (DSPs), and custom processors. The decreasing cost of DSPs has made these sophisticated chips very attractive for a number of embedded and real-time applications, including automotive, telecommunications, medical imaging, and many others—including even some games and home appliances. However, developing embedded and real-time DSP applications is a complex task influenced by many parameters and issues. DSP Software Development Techniques for Embedded and Real-Time Systems is an introduction to DSP software development for embedded and real-time developers giving details on how to use digital signal processors efficiently in embedded and real-time systems. The book covers software and firmware design principles, from processor architectures and basic theory to the selection of appropriate languages and basic algorithms. The reader will find practical guidelines, diagrammed techniques, tool descriptions, and code templates for developing and optimizing DSP software and firmware. The book also covers integrating and testing DSP systems as well as managing the DSP development effort. Digital signal processors (DSPs) are the future of microchips! Includes practical guidelines, diagrammed techniques, tool descriptions, and code templates to aid in the development and optimization of DSP software and firmware

Integrates rational approximation with adaptive filtering, providing viable, numerically reliable procedures for creating adaptive infinite impulse response (IIR) filters. The choice of filter structure to adapt, algorithm design and the approximation properties for each type of algorithm are also addressed. This work recasts the theory of adaptive IIR filters by concentrating on recursive lattice filters, freeing systems from the need for direct-form filters.;A solutions manual is available for instructors only. College or university bookstores may order five or more copies at a special student price which is available upon request.

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.

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.

Genetic Algorithms (GA) are based on the principles of natural selection and natural genetics that originate in biology. The Genetic Algorithm (GA) has been used for IIR adaptive system identification to deal with its multimodal error surface. The Genetic Algorithm (GA) can be very useful in the all three structures of IIR filters while the Gradient Algorithm experiences many difficulties due to the recursive feedback. This thesis will focus on the different performances on three structures of IIR adaptive filters based on the Genetic Algorithm (GA) and Multi-Parents Genetic Algorithm (MPGA). Experimental results demonstrate that, in general, the standard Genetic Algorithm (GA) direct form will have lower Mean Square Error (MSE), while the cascade and parallel forms will have higher convergence rates. The relative performance of three structures for Multi-Parents Genetic Algorithm (MPGA) is similar to the 2-parent Genetic Algorithm, but the rate of convergence is higher than the standard GA, which means the MPGA converges faster than the standard GA. Furthermore the performances of the three structures for the IIR filter based on modified Multi-Parents Genetic Algorithm (MPGA) are very similar. Simulation results demonstrate that when compared with the GA, the MPGA operates similarly on the three different structures, increases the rate of convergence rate and reduces the computational complexity. Finally, the Genetic Algorithm and the Gradient Algorithm were combined on the direct form to take advantage of each algorithm. When the rate of convergence decreases into a steady level the Gradient Algorithm is then applied so that the MSE will decrease again to a lower value, demonstrating that the combined algorithm obtains a more precise result and improve the performance.

Adaptive Filters: Structures, Algorithms and ApplicationsSpringerAdaptive Filtering ApplicationsBoD – Books on Demand

“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.

Adaptive filtering is useful in any application where the signals or the modeled system vary over time. The configuration of the system and, in particular, the position where the adaptive processor is placed generate different areas or application fields such as prediction, system identification and modeling, equalization, cancellation of interference, etc., which are very important in many disciplines such as control systems, communications, signal processing, acoustics, voice, sound and image, etc. The book consists of noise and echo cancellation, medical applications, communications systems and others hardly joined by their heterogeneity. Each application is a case study with rigor that shows weakness/strength of the method used, assesses its suitability and suggests new forms and areas of use. The problems are becoming increasingly complex and applications must be adapted to solve them. The adaptive filters have proven to be useful in these environments of multiple input/output, variant-time behaviors, and long and complex transfer functions effectively, but fundamentally they still have to evolve. This book is a demonstration of this and a small illustration of everything that is to come.

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

This volume presents the logical arithmetical or computational procedures within communications systems that will ensure the solution to various problems. The authors comprehensively introduce the theoretical elements that are at the basis of the field of algorithms for communications systems. Various applications of these algorithms are then illustrated with particular attention to wired and wireless network access technologies. * Provides a complete treatment of algorithms for communications systems, rarely presented together * Introduces the theoretical background to digital communications and signal processing * Features numerous applications including advanced wireless modems and echo cancellation techniques * Includes useful reference lists at the end of each chapter Graduate students in the fields of Telecommunications and Electrical Engineering Researchers and Professionals in the area of Digital Communications, Signal Processing and Computer Engineering will find this book invaluable.

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.

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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.

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.

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. 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.

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.

In recent years, the remarkable advances in medical imaging instruments have increased their use considerably for diagnostics as well as planning and follow-up of treatment. Emerging from the fields of radiology, medical physics and engineering, medical imaging no longer simply deals with the technology and interpretation of radiographic images. The limitless possibilities presented by computer science and technology, coupled with engineering advances in signal processing, optics and nuclear medicine have created the vastly expanded field of medical imaging. The Handbook of Medical Imaging is the first comprehensive compilation of the concepts and techniques used to analyze and manipulate medical images after they have been generated or digitized. The Handbook is organized in six sections that relate to the main functions needed for processing: enhancement, segmentation, quantification, registration, visualization as well as compression storage and telemedicine. * Internationally renowned authors(Johns Hopkins, Harvard, UCLA, Yale, Columbia, UCSF) * Includes imaging and visualization * Contains over 60 pages of stunning, four-color images

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

The main thrust is to provide students with a solid understanding of a number of important and related advanced topics in digital signal processing such as Wiener filters, power spectrum estimation, signal modeling and adaptive filtering. Scores of worked examples illustrate fine points, compare techniques and algorithms and facilitate comprehension of fundamental concepts. The book also features an abundance of interesting and challenging problems at the end of every chapter.· Background· Discrete-Time Random Processes· Signal Modeling· The Levinson Recursion· Lattice Filters· Wiener Filtering· Spectrum Estimation· Adaptive Filtering

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.

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.

Optimal and Adaptive Signal Processing covers the theory of optimal and adaptive signal processing using examples and computer simulations drawn from a wide range of applications, including speech and audio, communications, reflection seismology and sonar systems. The material is presented without a heavy reliance on mathematics and focuses on one-dimensional and array processing results, as well as a wide range of adaptive filter algorithms and implementations. Topics discussed include random signals and optimal processing, adaptive signal processing with the LMS algorithm, applications of adaptive filtering, algorithms and structures for adaptive filtering, spectral analysis, and array signal processing. Optimal and Adaptive Signal Processing is a valuable guide for scientists and engineers, as well as an excellent text for senior undergraduate/graduate level students in electrical engineering.

Harmonic analysis is a diverse field including such branches as signal processing, medical imaging, power electrical systems, wireless telecommunications, etc. This book is primarily written with the objective of providing recent developments and new techniques in harmonic analysis. In the recent years, a number of methods of quality control of signals under different perturbations, and especially the harmonics, have emerged. Some of these techniques are described in this book. This book is the result of contributions from many researchers and is a collection of eight research works, which are focused around the harmonic analysis theme but with different applications. The topics mainly concern the areas of medical imaging, biopotential systems, renewable energy conversion systems, wireless telecommunications, power converters, as well as the different techniques for estimating, analyzing, reducing, and eliminating harmonics.

The focus of this work is to explore structures and algorithms for two-dimensional adaptive signal processing. Applications in image and multichannel signal processing include 2-D adaptive differential pulse code modulation, interference cancellation, predictive coding, and noise suppression. Emphasis is placed both on FIR and IIR structures with primary benchmark issues being speed of convergence, computational complexity, and structural flexibility. The behavior of the 2-D, FIR, direct form adaptive filter is analogous to that of its 1-D counterpart. Eigenvalue disparity of the input autocorrelation matrix hinders the performance of the steepest descent adaptive algorithm. By implementing a Gauss-Newton sequential adaptive algorithm, the adaptive "modes" are effectively orthogonalized and normalized, thereby increasing the speed of convergence. An efficient block Levinson algorithm is utilized to implement the required matrix operations giving a fast quasi-Newton algorithm (FQN) with O($Nsp3$) complexity. The method exploits the Toeplitz-block Toeplitz structure of the resulting autocorrelation matrix estimate and realizes further computational savings by assuming that the autocorrelation matrix is constant over blocks of $Nsp2$ iterations. The FQN filter is compared to the 2-D transform domain filter, the McClellan transformation filter, and the 2-D recursive least squares filter. Two-dimensional infinite impulse response adaptive filters are also examined. It is found that 2-D IIR adaptive filters are plausible and useful. They exhibit convergence behavior which is dependent upon the 2-D indexing scheme. Several useful indexing methods are examined. A quasi-Newton acceleration algorithm is developed for this structure using the same method as above, except that some additional constraints must be imposed on the 2-D IIR autocorrelation matrix. The 2-D IIR error surface is not quadratic, and must be examined for the possible existence of local minima. Some preliminary results are presented. However, error surfaces can be graphically examined in the three-dimensional coefficient space for IIR filters with first-order denominators. Finally some applications are presented which utilize 2-D IIR adaptive filters. These include 2-D ADPCM and interference cancellation.

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 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.

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.

The field of Digital Signal Processing has developed so fast in the last two decades that it can be found in the graduate and undergraduate programs of most universities. This development is related to the growing available techno logies for implementing digital signal processing algorithms. The tremendous growth of development in the digital signal processing area has turned some of its specialized areas into fields themselves. If accurate information of the signals to be processed is available, the designer can easily choose the most appropriate algorithm to process the signal. When dealing with signals whose statistical properties are unknown, fixed algorithms do not process these signals efficiently. The solution is to use an adaptive filter that automatically changes its characteristics by optimizing the internal parameters. The adaptive filtering algorithms are essential in many statistical signal processing applications. Although the field of adaptive signal processing has been subject of research for over three decades, it was in the eighties that a major growth occurred in research and applications. Two main reasons can be credited to this growth, the availability of implementation tools and the appearance of early textbooks exposing the subject in an organized form. Presently, there is still a lot of activities going on in the area of adaptive filtering. In spite of that, the theor etical development in the linear-adaptive-filtering area reached a maturity that justifies a text treating the various methods in a unified way, emphasizing the algorithms that work well in practical implementation.

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.

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.

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.

Although adaptive filtering and adaptive array processing began with research and development efforts in the late 1950's and early 1960's, it was not until the publication of the pioneering books by Honig and Messerschmitt in 1984 and Widrow and Stearns in 1985 that the field of adaptive signal processing began to emerge as a distinct discipline in its own right. Since 1984 many new books have been published on adaptive signal processing, which serve to define what we will refer to throughout this book as conventional adaptive signal processing. These books deal primarily with basic architectures and algorithms for adaptive filtering and adaptive array processing, with many of them emphasizing practical applications. Most of the existing textbooks on adaptive signal processing focus on finite impulse response (FIR) filter structures that are trained with strategies based on steepest descent optimization, or more precisely, the least mean square (LMS) approximation to steepest descent. While literally hundreds of archival research papers have been published that deal with more advanced adaptive filtering concepts, none of the current books attempt to treat these advanced concepts in a unified framework. The goal of this new book is to present a number of important, but not so well known, topics that currently exist scattered in the research literature. The book also documents some new results that have been conceived and developed through research conducted at the University of Illinois during the past five years.

The two-volume set CCIS 827 and 828 constitutes the thoroughly refereed proceedings of the Third International Conference on Next Generation Computing Technologies, NGCT 2017, held in Dehradun, India, in October 2017. The 135 full papers presented were carefully reviewed and selected from 948 submissions. There were organized in topical sections named: Smart and Innovative Trends in Communication Protocols and Standards; Smart and Innovative Trends in Computational Intelligence and Data Science; Smart and Innovative Trends in Image Processing and Machine Vision; Smart Innovative Trends in Natural Language Processing for Indian Languages; Smart Innovative Trends in Security and Privacy.

The book presents high-quality research papers presented at the first international conference, ICICCD 2016, organised by the Department of Electronics, Instrumentation and Control Engineering of University of Petroleum and Energy Studies, Dehradun on 2nd and 3rd April, 2016. The book is broadly divided into three sections: Intelligent Communication, Intelligent Control and Intelligent Devices. The areas covered under these sections are wireless communication and radio technologies, optical communication, communication hardware evolution, machine-to-machine communication networks, routing techniques, network analytics, network applications and services, satellite and space communications, technologies for e-communication, wireless Ad-Hoc and sensor networks, communications and information security, signal processing for communications, communication software, microwave informatics, robotics and automation, optimization techniques and algorithms, intelligent transport, mechatronics system, guidance and navigation, algorithms, linear/non-linear control, home automation, sensors, smart cities, control systems, high performance computing, cognition control, adaptive control, distributed control, prediction models, hybrid control system, control applications, power system, manufacturing, agriculture cyber physical system, network control system, genetic control based, wearable devices, nano devices, MEMS, bio-inspired computing, embedded and real-time software, VLSI and embedded systems, FPGA, digital system and logic design, image and video processing, machine vision, medical imaging, and reconfigurable computing systems.