Blind Equalization And System Identification Batch Processing Algorithms Performance And Applicatio

The book constitutes the refereed proceedings of the 13th EAI International Conference on Communications and Networking, held in October 2018 in Chengdu, China. The 71 papers presented were carefully selected from 114 submissions. The papers are organized in topical sections on wireless communications and networking, next generation WLAN, big data networks, cloud communications and networking, ad hoc and sensor networks, satellite and space communications and networking, optical communications and networking, information and coding theory, multimedia communications and smart networking, green communications and computing, signal processing for communications, network and information security, machine-to-machine and IoT, communication QoS, reliability and modeling, cognitive radio and networks, smart internet of things modeling, pattern recognition and image signal processing, digital audio and video signal processing, antenna and microwave communications, radar imaging and target recognition, and video coding and image signal processing.

Abstract: "Adaptive linear predictors have been used extensively in practice in a wide variety of forms. In the main, their theoretical development is based upon the assumption of stationarity of the signals involved, particularly with respect to the second order statistics. On this basis, the well-known normal equations can be formulated. If high-order statistical stationarity is assumed, then the equivalent normal equations involve high-order signal moments. In either case, the cross moments (second or higher) are needed. This renders the adaptive prediction procedure non-blind. A novel procedure for blind adaptive prediction has been proposed and considerable implementation has been made in our contributions in the past year. The approach is based upon a suitable interpretation of blind equalization methods that satisfy the constant modulus property and offers significant deviations from the standard prediction methods. These blind adaptive algorithms are derived by formulating Lagrange equivalents from mechanisms of constrained optimization. In this report, other new update algorithms are derived from the fundamental concepts of advanced system identification to carry out the proposed blind adaptive prediction. The results of the work can be extended to a number of control-related problems, such as disturbance identification. The basic principles are outlined in this report and differences from other existing methods are discussed. The applications implemented are speech processing, such as coding and synthesis. Simulations are included to verify the novel modelling method."

Signal Processing for Wireless Communication Systems brings together in one place important contributions and up-to-date research results in this fast moving area. The Contributors to this work were selected from leading researchers and practitioners in this field. The book's 18 chapters are divided into three areas: systems, Networks, and Implementation Issues; Channel Estimation and Equalization; and Multiuser Detection. The Work, originally published as Volume 30, Numbers 1-3 of the Journal of VLSI Signal Processing Systems for Signal, Image, and Video Technology, will be valuable to anyone working or researching in the field of wireless communication systems. It serves as an excellent reference, providing insight into some of the most challenging issues being examined today.

A novel coding technique is presented for signal prediction with applications including speech coding, system identification, and estimation of input excitation. The approach is based on the blind equalization method for speech signal processing in conjunction with the geometric subspace projection theory to formulate the basic prediction equation. The speech-coding problem is often divided into two parts, a linear prediction model and excitation input. The parameter coefficients of the linear predictor and the input excitation are solved simultaneously and recursively by a conventional recursive least-squares algorithm. The excitation input is computed by coding all possible outcomes into a binary codebook. The coefficients of the linear predictor and excitation, and the index of the codebook can then be used to represent the signal. In addition, a variable-frame concept is proposed to block the same excitation signal in sequence in order to reduce the storage size and increase the transmission rate. The results of this work can be easily extended to the problem of disturbance identification. The basic principles are outlined in this report and differences from other existing methods are discussed. Simulations are included to demonstrate the proposed method.Juang, Jer-Nan and Chen, Ya-ChinLangley Research CenterSYSTEM IDENTIFICATION; SIGNAL PROCESSING; MATHEMATICAL MODELS; LINEAR PREDICTION; CODING; ALGORITHMS; VOICE DATA PROCESSING; SIMULATION

Blind Identification of Structured Dynamic Systems

IEEE Transactions on Circuits and Systems

Signal Processing Advances in Wireless and Mobile Communications: Trends in channel estimation and equalization

Independent Component Analysis and Applications

Recent Advances in Digital Telecommunications and Signal Processing

Many processes in nature arise from the interaction of periodic phenomena with random phenomena. The results are processes that are not periodic, but whose statistical functions are periodic functions of time. These processes are called cyclostationary and are an appropriate mathematical model for signals encountered in many fields including communications, radar, sonar, telemetry, acoustics, mechanics, econometrics, astronomy, and biology. Cyclostationary Processes and Time Series: Theory, Applications, and Generalizations addresses these issues and includes the following key features. Presents the foundations and developments of the second- and higher-order theory of cyclostationary signals Performs signal analysis using both the classical stochastic process approach and the functional approach for time series Provides applications in signal detection and estimation, filtering, parameter estimation, source location, modulation format classification, and biological signal characterization Includes algorithms for cyclic spectral analysis along with Matlab/Octave code Provides generalizations of the classical cyclostationary model in order to account for relative motion between transmitter and receiver and describe irregular statistical cyclicity in the data

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.

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 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.

Signal Processing for Wireless Communication Systems

Circuits and Systems for Wireless Communications

Blind Channel Identification of QAM Signals Via Fixed Stochastic Gradient Estimation

Linear Algebra for Signal Processing

Algorithms and Practical Implementation

Contemporary Coding Techniques and Applications for Mobile Communications

A triennial summation of the state of the art in radio science This book is the fourth in the modern series of triennial reviews prepared by the International Union of Radio Science to further communication and understanding of the status and future of radio science, both for those working in the field, and for those who want to know what is of current importance in this area. The International Union of Radio Science, URSI (Union Radio-Scientifique Internationale), has divided the subject of "Radio Science" according to the ten topics of the Scientific Commissions that make up URSI. This volume consists of thirty-eight original, peer-reviewed papers. Each paper provides a critical, in-depth review of–and, in many cases, tutorial on–advances and research that have been of significant importance within the area of interest of the Commissions during the past three to four years. Among the topics covered are: Electromagnetic metrology Fields and waves Signals and systems Electronics and photonics Electromagnetic noise and interference Wave propagation and remote sensing Ionospheric radio and propagation Waves in plasmas Radio astronomy Electromagnetics in biology and medicine With an included CD-ROM of the full book text, allowing the user to do full-text searching of all the papers, the Review of Radio Science: 1999—2002 is a resource of vital importance to anyone working in, or with an interest in, radio science.

Signal processing applications have burgeoned in the past decade. During the same time, signal processing techniques have matured rapidly and now include tools from many areas of mathematics, computer science, physics, and engineering. This trend will continue as many new signal processing applications are opening up in consumer products and communications systems. In particular, signal processing has been making increasingly sophisticated use of linear algebra on both theoretical and algorithmic fronts. This volume gives particular emphasis to exposing broader contexts of the signal processing problems so that the impact of algorithms and hardware can be better understood; it brings together the writings of signal processing engineers, computer engineers, and applied linear algebraists in an exchange of problems, theories, and techniques. This volume will be of interest to both applied mathematicians and engineers.

Blind Equalization and System IdentificationBatch Processing Algorithms, Performance and ApplicationsSpringer Science & Business Media

Simulation is a widely used mechanism for validating the theoretical models of networking and communication systems. Although the claims made based on simulations are considered to be reliable, how reliable they really are is best determined with real-world implementation trials. Simulation Technologies in Networking and Communications: Selecting the Best Tool for the Test addresses the spectrum of issues regarding the different mechanisms related to simulation technologies in networking and communications fields. Focusing on the practice of simulation testing instead of the theory, it presents the work of more than 50 experts from around the world. Considers superefficient Monte Carlo simulations Describes how to simulate and evaluate multicast routing algorithms Covers simulation tools for cloud computing and broadband passive optical networks Reports on recent developments in simulation tools for WSNs Examines modeling and simulation of vehicular networks The book compiles expert perspectives about the simulation of various networking and communications technologies. These experts review and evaluate popular simulation modeling tools and recommend the best tools for your specific tests. They also explain how to determine when theoretical modeling would be preferred over simulation. This book does not provide a verdict on the best suitable tool for simulation. Instead, it supplies authoritative analyses of the different kinds of networks and systems. Presenting best practices and insights from global experts, the book provides you with an understanding of what to simulate, where to simulate, whether to simulate or not, when to simulate, and how to simulate for a wide range of issues.

Speech Dereverberation

A Publication of the IEEE Circuits and Systems Society. Regular papers. I

Advances in Theory and Applications

A Thesis

Fourier Transform

Nonlinear System Representation, Identification, and Implementation

This text seeks to clarify various contradictory claims regarding capabilities and limitations of blind equalization. It highlights basic operating conditions and potential for malfunction. The authors also address concepts and principles of blind algorithms for single input multiple output (SIMO) systems and multi-user extensions of SIMO equalization and identification.

This volume on implementation techniques in digital signal processing systems clearly reveals the significance and power of the techniques that are available, and with further development, the essential role they will play as applied to a wide variety of areas. The authors are all to highly commended for their splendid contributors to this volume, which will provide a significant and unique international reference source for students, research workers, practicing engineers, and others for years to come.

Edited by the people who were forerunners in creating the field, together with contributions from 34 leading international experts, this handbook provides the definitive reference on Blind Source Separation, giving a broad and comprehensive description of all the core principles and methods, numerical algorithms and major applications in the fields of telecommunications, biomedical engineering and audio, acoustic and speech processing. Going beyond a machine learning perspective, the book reflects recent results in signal processing and numerical analysis, and includes topics such as optimization criteria, mathematical tools, the design of numerical algorithms, convolutive mixtures, and time frequency approaches. This Handbook is an ideal reference for university researchers, R&D engineers and graduates wishing to learn the core principles, methods, algorithms, and applications of Blind Source Separation. Covers the principles and major techniques and methods in one book Edited by the pioneers in the field with contributions from 34 of the world's experts Describes the main existing numerical algorithms and gives practical advice on their design Covers the latest cutting edge topics: second order methods; algebraic identification of under-determined mixtures, time-frequency methods, Bayesian approaches, blind identification under non negativity approaches, semi-blind methods for communications Shows the applications of the methods to key application areas such as telecommunications, biomedical engineering, speech, acoustic, audio and music processing, while also giving a general method for developing applications

The absence of training signals from many kinds of transmission necessitates the widespread use of blind equalization and system identification. There have been many algorithms developed for these purposes, working with one- or two-dimensional signals and with single-input single-output or multiple-input multiple-output, real or complex systems. It is now time for a unified treatment of this subject, pointing out the common characteristics of these algorithms as well as learning from their different perspectives. "Blind Equalization and System Identification" provides such a unified treatment presenting theory, performance analysis, simulation, implementation and applications. This is a textbook for graduate courses in discrete-time random processes, statistical signal processing, and blind equalization and system identification. It contains material which will also interest researchers and engineers working in digital communications, source separation, speech processing, and other, similar applications.

4G Technologies

Simulation Technologies in Networking and Communications

Communications and Networking

Digital Signal Processing Systems: Implementation Techniques

Selecting the Best Tool for the Test

Advanced Wireless Communications

This is volume 1 of a 2 volume set. This volume focuses on "channel estimation and equalization."

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.

The wireless community is on the verge of the standardization of fourth generation (4G) systems. Research has generated a number of solutions for significant improvement of system performance. The development of enabling technologies such as adaptive coding and modulation, iterative (turbo) decoding algorithms and space-time coding, means that industry can now implement these solutions. Advanced Wireless Communications: 4G Technologies focuses on the system elements that provide adaptability and reconfigurability and discusses how these features can improve 4G system performance. There are several different systems comprising 4G, including adaptive WCDMA (Wideband Code Division Multiple Access), ATDMA (Adaptive Time Division Multiple Access), Multicarrier (OFDMA) and Ultra Wide Band (UWB) receiver elements. This book provides a comparative study of these technologies and focuses on their future co-existence. Topics covered include: Space Time Coding, including discussions on diversity gain, the encoding and transmission sequence, the combining s cheme and ML decision rule for two-branch transmit diversity scheme with one and M receivers. Ultra Wide Band Radio, UWB multiple access in Gaussian channels, the UWB channel, UWB system with M-ary modulation, M-ary PPM UWB multiple access, coded UWB schemes, multi-user detection in UWB radio, UWB with space time processing and beam forming for UWB radio. Antenna array signal processing with focus on Space-Time receivers for CDMA communications, MUSIC and ESPRIT DOA estimation, joint array combining and MLSE receivers, joint combiner and channel response estimation and complexity reduction in the wide-band beam forming Channel modeling and measurement, adaptive MAC, adaptive routing and TCP layer are also addressed. This book will supply the reader with a comprehensive understanding of the relationship between the systems performance, its complexity/reliability and cost-effectiveness. It gives an insight into the impact of existing and new technologies on the receiver structure and provides an understanding of current approaches and evolving directions for personal and indoor communication.

Equalizers are present in all forms of communication systems. Neuro-Fuzzy Equalizers for Mobile Cellular Channels details the modeling of a mobile broadband communication channel and designing of a neuro-fuzzy adaptive equalizer for it. This book focuses on the concept of the simulation of wireless channel equalizers using the adaptive-network-based fuzzy inference system (ANFIS). The book highlights a study of currently existing equalizers for wireless channels. It discusses several techniques for channel equalization, including the type-2 fuzzy adaptive filter (type-2 FAF), compensatory neuro-fuzzy filter (CNFF), and radial basis function (RBF) neural network. Neuro-Fuzzy Equalizers for Mobile Cellular Channels starts with a brief introduction to channel equalizers, and the nature of mobile cellular channels with regard to the frequency reuse and the resulting CCI. It considers the many channel models available for mobile cellular channels, establishes the mobile indoor channel as a Rayleigh fading channel, presents the channel equalization problem, and focuses on various equalizers for mobile cellular channels. The book discusses conventional equalizers like LE and DFE using a simple LMS algorithm and transversal equalizers. It also covers channel equalization with neural networks and fuzzy logic, and classifies various equalizers. This being a fairly new branch of study, the book considers in detail the concept of fuzzy logic controllers in noise cancellation problems and provides the fundamental concepts of neuro-fuzzy. The final chapter offers a recap and explores venues for further research. This book also establishes a common mathematical framework of the equalizers using the RBF model and develops a mathematical model for ultra-wide band (UWB) channels using the channel co-variance matrix (CCM). Introduces the novel concept of the application of adaptive-network-based fuzzy inference system (ANFIS) in the design of wireless channel equalizers Provides model ultra-wide band (UWB) channels using channel co-variance matrix Offers a formulation of a unified radial basis function (RBF) framework for ANFIS-based and fuzzy adaptive filter (FAF) Type II, as well as compensatory neuro-fuzzy equalizers Includes extensive use of MATLAB® as the simulation tool in all the above cases

Review of Radio Science

13th EAI International Conference, ChinaCom 2018, Chengdu, China, October 23-25, 2018, Proceedings

Nonlinear Signal and Image Processing

System Identification (SYSID '03)

A Novel Approach for Adaptive Signal Processing

Blind Equalization and Identification

`The field of signal processing has seen explosive growth during the past decades; almost all textbooks on signal processing have a section devoted to the Fourier transform theory. For this reason, this book focuses on the Fourier transform applications in signal processing techniques. The book chapters are related to DFT, FFT, OFDM, estimation techniques and the image processing techqniques. It is hoped that this book will provide the background, references and the incentive to encourage further research and results in this area as well as provide tools for practical applications. It provides an applications-oriented to signal processing written primarily for electrical engineers, communication engineers, signal processing engineers, mathematicians and graduate students will also find it useful as a reference for their research activities.`

`Speech Dereverberation gathers together an overview, a mathematical formulation of the problem and the state-of-the-art solutions for dereverberation. Speech Dereverberation presents current approaches to the problem of reverberation. It provides a review of topics in room acoustics and also describes performance measures for dereverberation. The algorithms are then explained with mathematical analysis and examples that enable the reader to see the strengths and weaknesses of the various techniques, as well as giving an understanding of the questions still to be addressed. Techniques rooted in speech enhancement are included, in addition to a treatment of multichannel blind acoustic system identification and inversion. The TRINICON framework is shown in the context of dereverberation to be a generalization of the signal processing for a range of analysis and enhancement techniques. Speech Dereverberation is suitable for students at masters and doctoral level, as well as established researchers.`

`Part I: RF System Integration. 1. RF System Integration; C. Toumazou. 2. RF System Board Level Integration for Mobile Phones; G.J. Aspin. 3. Integration of RF Systems on a Chip; P.J. Mole. 4. Towards the Full Integration of Wireless Front-End Circuits; M. Steyaert. 5. GSM Transceiver Front-End Circuits in 0.25 mum CMOS; Q. Huang, et al. Part II: RF Front-End Circuits. 6. RF Front-End Circuits; Q. Huang. 7. Phase-Noise-to-Carrier Ratio in LC Oscillators; Q. Huang. 8. Design Study of a 900 MHz/1.8 GHz CMOS Transceiver for Dual-Band Applications; B. Razavi. 9. Integrated Wireless Transc.`

`An authoritative guide to up-to-date research results on chaotic signal processing aimed at researchers and graduate students in chaos, applied nonlinear dynamics, signal processing and radar communications. This book examines the applications of chaotic signal processing to radar, communications, system identification and computing.`

`Signal Processing`

`Theory, Applications, and Generalizations`

`Blind System Identification and Blind Equalization Using Higher-order Cumulants`

`Handbook of Blind Source Separation`

`The HOS for Blind Identification and Equalization of Systems`

`Advanced Signal Processing`

Chaotic Signals in Digital Communications combines fundamental background knowledge with state-of-the-art methods for using chaotic signals and systems in digital communications. The book builds a bridge between theoretical works and practical implementation to help researchers attain consistent performance in realistic environments. It shows the possible shortcomings of the chaos-based communication systems proposed in the literature, particularly when they are subjected to non-ideal conditions. It also presents a toolbox of techniques for researchers working to actually implement such systems. A Combination of Tutorials and In-Depth, Cutting-Edge Research Featuring contributions by active leading researchers, the book begins with an introduction to communication theory, dynamical systems, and chaotic communications suitable for those new to the field. This lays a solid foundation for the more applied chapters that follow. A Toolbox of Techniques—Including New Ways to Tackle Channel Imperfections The book covers typical chaos communication methods, namely chaotic masking, chaotic modulation, chaotic shift key, and symbolic message bearing, as well as bidirectional communication and secure communication. It also presents novel methodologies to deal with communication channel imperfections. These tackle band-limited channel chaos communication, radio channels with fading, and the resistance of a special chaotic signal to multipath propagations. In addition, the book addresses topics related to engineering applications, such as optical communications, chaotic matched filters and circuit implementations, and microwave frequency-modulated differential chaos shift keying (FM-DCSK) systems. Insights for Both Theoretical and Experimental Researchers Combining theory and practice, this book offers a unique perspective on chaotic communication in the context of non-ideal conditions. Written for theoretical and experimental researchers, it tackles the practical issues faced in implementing chaos-based signals and systems in digital communications applications.

Discover the Applicability, Benefits, and Potential of New Technologies As advances in algorithms and computer technology have bolstered the digital signal processing capabilities of real-time sonar, radar, and non-invasive medical diagnostics systems, cutting-edge military and defense research has established conceptual similarities in these areas. Now civilian enterprises can use government innovations to facilitate optimal functionality of complex real-time systems. Advanced Signal Processing details a cost-efficient generic processing structure that exploits these commonalities to benefit commercial applications. Learn from a Renowned Defense Scientist, Researcher, and Innovator The author preserves the mathematical focus and key information from the first edition that provided invaluable coverage of topics including adaptive systems, advanced beamformers, and volume visualization methods in medicine. Integrating the best features of non-linear and conventional algorithms and explaining their application in PC-based architectures, this text contains new data on: Advances in biometrics, image segmentation, registration, and fusion techniques for 3D/4D ultrasound, CT, and MRI Fully digital 3D/ (4D: 3D+time) ultrasound system technology, computing architecture requirements, and relevant implementation issues State-of-the-art non-invasive medical procedures, non-destructive 3D tomography imaging and biometrics, and monitoring of vital signs Cardiac motion correction in multi-slice X-ray CT imaging Space-time adaptive processing and detection of targets interference-intense backgrounds comprised of clutter and jamming With its detailed explanation of adaptive, synthetic-aperture, and fusion-processing schemes with near-instantaneous convergence in 2-D and 3-D sensors (including planar, circular, cylindrical, and spherical arrays), the quality and illustration of this text ' s concepts and techniques will make it a favored reference.

The scope of the symposium covers all major aspects of system identification, experimental modelling, signal processing and adaptive control, ranging from theoretical, methodological and scientific developments to a large variety of (engineering) application areas. It is the intention of the organizers to promote SYSID 2003 as a meeting place where scientists and engineers from several research communities can meet to discuss issues related to these areas. Relevant topics for the symposium program include: Identification of linear and multivariable systems, identification of nonlinear systems, including neural networks, identification of hybrid and distributed systems, Identification for control, experimental modelling in process control, vibration and modal analysis, model validation, monitoring and fault detection, signal processing and communication, parameter estimation and inverse modelling, statistical analysis and uncertainty bounding, adaptive control and data-based controller tuning, learning, data mining and Bayesian approaches, sequential Monte Carlo methods, including particle filtering, applications in process control systems, motion control systems, robotics, aerospace systems, bioengineering and medical systems, physical measurement systems, automotive systems, econometrics, transportation and communication systems *Provides the latest research on System Identification *Contains contributions written by experts in the field *Part of the IFAC Proceedings Series which provides a comprehensive overview of the major topics in control engineering.

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

Adaptive Blind Signal and Image Processing

Conference Record

Cyclostationary Processes and Time Series

1999-2002 URSI

Signal Prediction with Input Identification

A Deterministic Perspective

Modern error control coding methods based on turbo coding have essentially solved the problem of reliable data communications over noisy channels. Contemporary Coding Techniques and Applications for Mobile Communications provides a clear, comprehensive, and practical grounding on the subject matter, examining the fundamentals, theory, and application of contemporary coding techniques and the applications for mobile communications. Written from the perspective that error control coding techniques will facilitate future digital data links, the book provides in-depth coverage on topics such as modulation techniques, multiplexing, channel models, MIMO systems, fundamental coding techniques, trellis coding modulation, turbo codes, and multilevel turbo codes. The first part of the text presents fundamental information on modulation, multiplexing, channel models, and traditional coding methods. The second part explains advanced coding techniques, provides simulation results, and compares them with related methods. It also provides new coding algorithms and new research areas such as image transmission with step-by-step guidelines.

Convex Optimization for Signal Processing and Communications: From Fundamentals to Applications provides fundamental background knowledge of convex optimization, while striking a balance between mathematical theory and applications in signal processing and communications. In addition to comprehensive proofs and perspective interpretations for core convex optimization theory, this book also provides many insightful figures, remarks, illustrative examples, and guided journeys from theory to cutting-edge research explorations, for efficient and in-depth learning, especially for engineering students and professionals. With the powerful convex optimization theory and tools, this book provides you with a new degree of freedom and the capability of solving challenging real-world scientific and engineering problems.

A novel coding technique is presented for signal prediction with applications including speech coding, system identification, and estimation of input excitation. The approach is based on the blind equalization method for speech signal processing in conjunction with the geometric subspace projection theory to formulate the basic prediction equation. The speech-coding problem is often divided into two parts, a linear prediction model and excitation input. The parameter coefficients of the linear predictor and the input excitation are solved simultaneously and recursively by a conventional recursive least-squares algorithm. The excitation input is computed by coding all possible outcomes into a binary notebook. The coefficients of the linear predictor and excitation, and the index of the codebook can then be used to represent the signal. In addition, a variable-frame concept is proposed to block the same excitation signal in sequence in order to reduce the storage size and increase the transmission rate. The results of this work can be easily extended to the problem of disturbance identification. The basic principles are outlined in this report and differences from other existing methods are discussed. Simulations are included to demonstrate the proposed method.

Adaptive Blind Equalization with Applications in Communication Sysytems [sic]

Theory and Implementation for Sonar, Radar, and Non-Invasive Medical Diagnostic Systems, Second Edition

Chaotic Signals in Digital Communications

Digital Signal Processing Fundamentals

A Proceedings Volume from the 13th IFAC Symposium on System Identification, Rotterdam, the Netherlands, 27-29 August 2003

Chaotic Signal Processing