Signal Processing First James Mcclellan

This book covers the fundamental concepts in signal processing illustrated with Python code and made available via IPython Notebooks, which are live, interactive, browser-based documents that allow one to change parameters, redraw plots, and tinker with the ideas presented in the text. Everything in the text is computable in this format and thereby invites readers to “experiment and learn” as they read. The book focuses on the core, fundamental principles of signal processing. The code corresponding to this book uses the core functionality of the scientific Python toolchain that should remain unchanged into the foreseeable future. For those looking to migrate their signal processing codes to Python, this book illustrates the key signal and plotting modules that can ease this transition. For those already comfortable with the scientific Python toolchain, this book illustrates the fundamental concepts in signal processing and provides a gateway to further signal processing concepts.

This textbook provides an introduction to the study of digital signal processing, employing a top-to-bottom structure to motivate the reader, a graphical approach to the solution of the signal processing mathematics, and extensive use of MATLAB. In contrast to the conventional teaching approach, the book offers a top-down approach which first introduces students to digital filter design, provoking questions about the mathematical tools required. The following chapters provide answers to these questions, introducing signals in the discrete domain, Fourier analysis, filters in the time domain and the Z-transform. The author introduces the mathematics in a conceptual manner with figures to illustrate the physical meaning of the equations involved. Chapter six builds on these concepts and discusses advanced filter design, and chapter seven discusses matters of practical implementation. This book introduces the corresponding MATLAB functions and programs in every chapter with examples, and the final chapter introduces the actual real-time filter from MATLAB. Aimed primarily at undergraduate students in electrical and electronic engineering, this book enables the reader to implement a digital filter using MATLAB.

Digital signal processing lies at the heart of the communications revolution and is an essential element of key technologies such as mobile phones and the Internet. This book covers all the major topics in digital signal processing (DSP) design and analysis, supported by MatLab examples and other modelling techniques. The authors explain clearly and concisely why and how to use digital signal processing systems; how to approximate a desired transfer function characteristic using polynomials and ratio of polynomials; why an appropriate mapping of a transfer function on to a suitable structure is important for practical applications; and how to analyse, represent and explore the trade-off between time and frequency representation of signals. An ideal textbook for students, it will also be a useful reference for engineers working on the development of signal processing systems.

For courses in Signals and Systems offered in departments of Electrical Engineering. This book focuses on the mathematical analysis and design of analog signal processing using a just in time approach - new ideas and topics relevant to the narrative are introduced only when needed, and no chapters are stand alone. Topics are developed throughout the narrative, and individual ideas appear frequently as needed.

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

DSP First presents basic DSP concepts in a clear and intuitive style, with a hands-on practical approach.

Methods of signal analysis represent a broad research topic with applications in many disciplines, including engineering, technology, biomedicine, seismography, eco nometrics, and many others based upon the processing of observed variables. Even though these applications are widely different, the mathematical background be hind them is similar and includes the use of the discrete Fourier transform and z-transform for signal analysis, and both linear and non-linear methods for signal identification, modelling, prediction, segmentation, and classification. These meth ods are in many cases closely related to optimization problems, statistical methods, and artificial neural networks. This book incorporates a collection of research papers based upon selected contri butions presented at the First European Conference on Signal Analysis and Predic tion (ECSAP-97) in Prague, Czech Republic, held June 24-27, 1997 at the Strahov Monastery. Even though the Conference was intended as a European Conference, at first initiated by the European Association for Signal Processing (EURASIP), it was very gratifying that it also drew significant support from other important scientific societies, including the IEE, Signal Processing Society of IEEE, and the Acoustical Society of America. The organizing committee was pleased that the re sponse from the academic community to participate at this Conference was very large; 128 summaries written by 242 authors from 36 countries were received. In addition, the Conference qualified under the Continuing Professional Development Scheme to provide PD units for participants and contributors.

Beamforming: Sensor Signal Processing for Defence Applications presents a range of important research contributions concerned with sensor array signal processing and, in particular, with the superresolution beamformers fundamental to many civilian and defence applications. Both space and space-time (STAP) beamforming algorithms and their application to radar systems are considered with emphasis given to "look-down" airborne radars, synthetic aperture radar (SAR), arrayed MIMO radar and a number of common wake-wave detection algorithms for two-dimensional SAR imagery. Furthermore, ocean towed arrays, which find applications in a variety of areas such as defence, oil and gas exploration, and geological and marine life studies, are also considered paying particular attention to receiver positional uncertainties resulting from the array's flexible structure. Array geometrical and electrical uncertainties, design of auto-calibration algorithms, beamforming "pointing" error uncertainties and robustification issues are also presented. This book is self-contained and unified in its presentation, and comprehensively covers some of the classic and fundamental models of beamforming for sensor signal processing. It is suitable as an advanced textbook for graduate students and researchers in the area of signal processing, as well as a reference book for engineers in the defence industry. Contents:Space-Time Adaptive Beamforming Algorithms for Airborne Radar Systems (Rodrigo de Lamare)Transmit Beamforming for Forward-Looking Space-Time Radars (Mathini Sellathurai and David Wilcox)Digital Beamforming for Synthetic Aperture Radar (Karen Mak and Athanassios Manikas)Arrayed MIMO Radar: Multi-target Parameter Estimation for Beamforming (Harry Commin, Kai Luo and Athanassios Manikas)Beamforming for Wake Wave Detection and Estimation — An Overview (Karen Mak and

Athanassios Manikas)Towed Arrays: Channel Estimation, Tracking and Beamforming (Vidhya Sridhar, Marc Willerton and Athanassios Manikas)Array Uncertainties and Auto-calibration (Marc Willerton, Evangelos Venieris and Athanassios Manikas)Robust Beamforming to Pointing Errors (Jie Zhuang and Athanassios Manikas) Readership: Postgraduate students and researchers working in the area of signal processing as well researchers working in the defence industry. The UDRC runs a series of short courses in signal processing for PhD students and industrial researchers and this book is recommended reading. Key Features:Unique treatment of beamformingUnique modelling techniques using array processing of modern radar systems suchs as MIMO, SAR, etc.New material related to the research carried out at UDRC. This book is considered as one of the academic outcomes of the UDRC in Signal ProcessingKeywords:Space-Time Algorithms;Adaptive Beamforming;Transmit Beamforming;Robust Beamforming;Digital Beamforming;Spatiotemporal Beamforming;Forward-Looking STAP Radar;Airborne Radar;Synthetic Aperture Radar;MIMO Radar;Array Calibration;Array Uncertainties

CD-ROM contains: Demonstrations -- Problem solutions.

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

This book serves as an ideal starting point for newcomers and an excellent reference source for people already working in the field. Researchers and graduate students in signal processing, computer science, acoustics and music will primarily benefit from this text. It could be used as a textbook for advanced courses in music signal processing. Since it only requires a basic knowledge of signal processing, it is accessible to undergraduate students.

This book presents the investigation of special type of IIR polyphase filter structures combined with frequency transformation techniques, and their application for custom fixed-point implementation. Featuring a wealth of design and analysis techniques, it includes sufficient introductory material to enable non-experts to understand the topics.

For senior or introductory graduate-level courses in digital signal processing. Developed by a group of six eminent scholars and teachers, this book offers a rich collection of exercises and projects which guide students in the use of MATLAB v5 to explore major topical areas in digital signal processing.

Mathematical morphology (MM) is a powerful methodology for the quantitative analysis of geometrical structures. It consists of a broad and coherent collection of theoretical concepts, nonlinear signal operators, and algorithms aiming at extracting, from images or other geometrical objects, information related to their shape and size. Its mathematical origins stem from set theory, lattice algebra, and integral and stochastic geometry. MM was initiated in the late 1960s by G. Matheron and J. Serra at the Fontainebleau School of Mines in France. Originally it was applied to analyzing images from geological or biological specimens. However, its rich theoretical framework, algorithmic efficiency, easy implementability on special hardware, and suitability for many shape- oriented problems have propelled its widespread diffusion and adoption by many academic and industry groups in many countries as one among the dominant image analysis methodologies. The purpose of Mathematical Morphology and its Applications to Image and Signal Processing is to provide the image analysis community with a sampling from the current developments in the theoretical (deterministic and stochastic) and computational aspects of MM and its applications to image and signal processing. The book consists of the papers presented at the ISMM'96 grouped into the following themes: Theory Connectivity Filtering Nonlinear System Related to Morphology Algorithms/Architectures Granulometries, Texture Segmentation Image Sequence Analysis Learning Document Analysis Applications

Briefly describes the physical characteristics, the habitat, and the behavior of the Alaskan brown bear.

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Electronics and Communications for Scientists and Engineers, Second Edition, offers a valuable and unique overview on the basics of electronic technology and the internet. Class-tested over many years with students at Northwestern University, this useful text covers the essential electronics and communications topics for students and practitioners in engineering, physics, chemistry, and other applied sciences. It describes the electronic underpinnings of the World Wide Web and explains the basics of digital technology, including computing and communications, circuits, analog and digital electronics, as well as special topics such as operational amplifiers, data compression, ultra high definition

TV, artificial intelligence, and quantum computers. Incorporates comprehensive updates and expanded material in all chapters where appropriate Includes new problems added throughout the text Features an updated section on RLC circuits Presents revised and new content in Chapters 7, 8, and 9 on digital systems, showing the many changes and rapid progress in these areas since 2000

With updates and enhancements to the incredibly successful first edition, Probability and Random Processes for Electrical and Computer Engineers, Second Edition retains the best aspects of the original but offers an even more potent introduction to probability and random variables and processes. Written in a clear, concise style that illustrates the subject’s relevance to a wide range of areas in engineering and physical and computer sciences, this text is organized into two parts. The first focuses on the probability model, random variables and transformations, and inequalities and limit theorems. The second deals with several types of random processes and queuing theory. New or Updated for the Second Edition: A short new chapter on random vectors that adds some advanced new material and supports topics associated with discrete random processes Reorganized chapters that further clarify topics such as random processes (including Markov and Poisson) and analysis in the time and frequency domain A large collection of new MATLAB®-based problems and computer projects/assignments Each Chapter Contains at Least Two Computer Assignments Maintaining the simplified, intuitive style that proved effective the first time, this edition integrates corrections and improvements based on feedback from students and teachers. Focused on strengthening the reader’s grasp of underlying mathematical concepts, the book combines an abundance of practical applications, examples, and other tools to simplify unnecessarily difficult solutions to varying engineering problems in communications, signal processing, networks, and associated fields.

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

The Only DSP Book 100% Focused on Step-by-Step Design and Implementation of Real Devices and Systems in Hardware and Software Practical Applications in Digital Signal Processing is the first DSP title to address the area that even the excellent engineering textbooks of today tend to omit. This book fills a large portion of that omission by addressing circuits and system applications that most design engineers encounter in the modern signal processing industry. This book includes original work in the areas of Digital Data Locked Loops (DLLs), Digital Automatic Gain Control (dAGC), and the design of fast elastic store memory used for synchronizing independently clocked asynchronous data bit streams. It also contains detailed design discussions on Cascaded Integrator Comb (CIC) filters, including the seldom-covered topic of bit pruning. Other topics not extensively covered in other modern textbooks, but detailed here, include analog and digital signal tuning, complex-to-real conversion, the design of digital channelizers, and the techniques of digital frequency synthesis. This book also contains an appendix devoted to the techniques of writing mixed-language C\C++ Fortran programs. Finally, this book contains very extensive review material covering important engineering mathematical tools such as the Fourier series, the Fourier transform, the z transform, and complex variables. Features of this book include • Thorough coverage of the complex-to-real conversion of digital signals • A complete tutorial on digital frequency synthesis • Lengthy discussion of analog and digital tuning and signal translation • Detailed coverage of the design of elastic store memory • A comprehensive study of the design of digital data locked loops • Complete coverage of the design of digital channelizers • A detailed treatment on the design of digital automatic gain control • Detailed techniques for the design of digital and multirate filters • Extensive coverage of the CIC filter, including the topic of bit pruning • An extensive review of complex variables • An extensive review of the Fourier series, and continuous and discrete Fourier transforms • An extensive review of the z transform

With the emergence of compressive sensing and sparse signal reconstruction, approaches to urban radar have shifted toward relaxed constraints on signal sampling schemes in time and space, and to effectively address logistic difficulties in data acquisition. Traditionally, these challenges have hindered high resolution imaging by restricting both bandwidth and aperture, and by imposing uniformity and bounds on sampling rates. Compressive Sensing for Urban Radar is the first book to focus on a hybrid of two key areas: compressive sensing and urban sensing. It explains how reliable imaging, tracking, and localization of indoor targets can be achieved using compressed observations that amount to a tiny percentage of the entire data volume. Capturing the latest and most important advances in the field, this state-of-the-art text: Covers both ground-based and airborne synthetic aperture radar (SAR) and uses different signal waveforms Demonstrates successful applications of compressive sensing for target detection and revealing building interiors Describes problems facing urban radar and highlights sparse reconstruction techniques applicable to urban environments Deals with both stationary and moving indoor targets in the presence of wall clutter and multipath exploitation Provides numerous supporting examples using real data and computational electromagnetic modeling Featuring 13 chapters written by leading researchers and experts, Compressive Sensing for Urban Radar is a useful and authoritative reference for radar engineers and defense contractors, as well as a seminal work for graduate students and academia.

Amazon.com’s Top-Selling DSP Book for Seven Straight Years—Now Fully Updated! Understanding Digital Signal Processing, Third Edition, is quite simply the best resource for engineers and other technical professionals who want to master and apply today’s latest DSP techniques. Richard G. Lyons has updated and expanded his best-selling second edition to reflect the newest technologies, building on the exceptionally readable coverage that made it the favorite of DSP professionals worldwide. He has also added hands-on problems to every chapter, giving students even more of the practical experience they need to succeed. Comprehensive in scope and clear in approach, this book achieves the perfect balance between theory and practice, keeps math at a tolerable level, and makes DSP exceptionally accessible to beginners without ever oversimplifying it. Readers can thoroughly grasp the basics and quickly move on to more sophisticated techniques. This edition adds extensive new coverage of FIR and IIR filter analysis techniques, digital differentiators, integrators, and matched filters. Lyons has significantly updated and expanded his discussions of multirate processing techniques, which are crucial to modern wireless and satellite communications. He also presents nearly twice as many DSP Tricks as in the second edition—including techniques even seasoned DSP professionals may have overlooked. Coverage includes New homework problems that deepen your understanding and help you apply what you've learned Practical, day-to-day DSP implementations and problem-solving throughout Useful new guidance on generalized digital networks, including discrete differentiators, integrators, and matched filters Clear descriptions of statistical measures of signals, variance reduction by averaging, and real-world signal-to-noise ratio (SNR) computation A significantly expanded chapter on sample rate conversion (multirate systems) and associated filtering techniques New guidance on implementing fast convolution, IIR filter scaling, and more Enhanced coverage of analyzing digital filter behavior and performance for diverse communications and biomedical applications Discrete sequences/systems, periodic sampling, DFT, FFT, finite/infinite impulse response filters, quadrature (I/Q) processing, discrete Hilbert transforms, binary number formats, and much more

This textbook on signals and systems provides a complete array of MATLAB tools specifically designed for the course, compatible with MATLAB 3.5 or 4.0. This software tool is used in the context of a presentation of systems concepts and analysis techniques. Use of MATLAB helps students to understand what the mathematical abstractions represent, which helps them to understand the behavior of a variety of systems. In response to a wide range of signal inputs.The software provides students with instantaneous feedback which encourages them to explore problems further. Topics covered in the text include signals, systems, convolution, Fourier series and transforms, Laplace transforms, analog filters, sampling, the discrete-time Fourier transform (DTFT), FFT, z-transforms and digital filters. All basic concepts are illustrated by worked examples. End-of-chapter problems include simple drills as well as more challenging exercises that develop or extend the concepts covered. A unique (but optional) feature of this text is the software supplied on disk which contains ready-to-run demonstrations, interactive programs and full-fledged general purpose programs. ..The software runs under MATLAB and includes routines developed for plotting functions, generating random signals, regular and periodic convolution, analytical and numerical solution of differential and difference equations, Fourier analysis, frequency response, asymptotic Bode plots, closed form expressions for Laplace and z-transforms and inverse transforms, classical analog filter design, sampling, quantization, interpolation, FIR and IIR filter design using various methods, and more. So as not to affect the continuity and logical flow of the text material, the programs are described and used only in the accompanying documentation on disk. A MATLAB appendix to each chapter lists the appropriate programs, and each section that can be tied to the software is marked.

Digital Signal Processing for Communication Systems examines the plans for the future and the progress that has already been made, in the field of DSP and its applications to communication systems. The book pursues the progression from communication and information theory through to the implementation, evaluation and performance enhancing of practical communication systems using DSP technology. Digital Signal Processing for Communication Systems looks at various types of coding and modulation techniques, describing different applications of Turbo-Codes, BCH codes and general block codes, pulse modulations, and combined modulation and coding in order to improve the overall system performance. The book examines DSP applications in measurements performed for channel characterisation, pursues the use of DSP for design of effective channel simulators, and discusses equalization and detection of various signal formats for different channels. A number of system design issues are presented where digital signal processing is involved, reporting on the successful implementation of the system components using DSP technology, and including the problems involved with implementation of some DSP algorithms. Digital Signal Processing for Communication Systems serves as an excellent resource for professionals and researchers who deal with digital signal processing for communication systems, and may serve as a text for advanced courses on the subject.

For introductory courses (freshman and sophomore courses) in Digital Signal Processing and Signals and Systems. Text may be used before the student has taken a course in circuits. DSP First and it's accompanying digital assets are the result of more than 20 years of work that originated from, and was guided by, the premise that signal processing is the best starting point for the study of electrical and computer engineering. The "DSP First" approach introduces the use of mathematics as the language for thinking about engineering problems, lays the groundwork for subsequent courses, and gives students hands-on experiences with MATLAB. The Second Edition features three new chapters on the Fourier Series, Discrete-Time Fourier Transform, and the The Discrete Fourier Transform as well as updated labs, visual demos, an update to the existing chapters, and hundreds of new homework problems and solutions.

"Engineering Computations and Modeling in MATLAB/Simulink" provides a broad overview of The

This book presents the state of the art in sparse and multiscale image and signal processing, covering linear multiscale transforms, such as wavelet, ridgelet, or curvelet transforms, and non-linear multiscale transforms based on the median and mathematical morphology operators. Recent concepts of sparsity and morphological diversity are described and exploited for various problems such as denoising, inverse problem regularization, sparse signal decomposition, blind source separation, and compressed sensing. This book weds theory and practice in examining applications in areas such as astronomy, biology, physics, digital media, and forensics. A final chapter explores a paradigm shift

in signal processing, showing that previous limits to information sampling and extraction can be overcome in very significant ways. Matlab and IDL code accompany these methods and applications to reproduce the experiments and illustrate the reasoning and methodology of the research are available for download at the associated web site.
A practical and accessible guide to understanding digital signal processing Introduction to Digital Signal Processing and Filter Design was developed and fine-tuned from the author's twenty-five years of experience teaching classes in digital signal processing. Following a step-by-step approach, students and professionals quickly master the fundamental concepts and applications of discrete-time signals and systems as well as the synthesis of these systems to meet specifications in the time and frequency domains. Striking the right balance between mathematical derivations and theory, the book features: * Discrete-time signals and systems * Linear difference equations * Solutions by recursive algorithms * Convolution * Time and frequency domain analysis * Discrete Fourier series * Design of FIR and IIR filters * Practical methods for hardware implementation A unique feature of this book is a complete chapter on the use of a MATLAB(r) tool, known as the FDA (Filter Design and Analysis) tool, to investigate the effect of finite word length and different formats of quantization, different realization structures, and different methods for filter design. This chapter contains material of practical importance that is not found in many books used in academic courses. It introduces students in digital signal processing to what they need to know to design digital systems using DSP chips currently available from industry. With its unique, classroom-tested approach, Introduction to Digital Signal Processing and Filter Design is the ideal text for students in electrical and electronic engineering, computer science, and applied mathematics, and an accessible introduction or refresher for engineers and scientists in the field.
Provides easy learning and understanding of DWT from a signal processing point of view Presents DWT from a digital signal processing point of view, in contrast to the usual mathematical approach, making it highly accessible Offers a comprehensive coverage of related topics, including convolution and correlation, Fourier transform, FIR filter, orthogonal and biorthogonal filters Organized systematically, starting from the fundamentals of signal processing to the more advanced topics of DWT and Discrete Wavelet Packet Transform. Written in a clear and concise manner with abundant examples, figures and detailed explanations Features a companion website that has several MATLAB programs for the implementation of the DWT with commonly used filters “This well-written textbook is an introduction to the theory of discrete wavelet transform (DWT) and its applications in digital signal and image processing.” -- Prof. Dr. Manfred Tasche - Institut für Mathematik, Uni Rostock Full review at https://zbmath.org/?q=an:06492561
In this supplementary text, MATLAB is used as a computing tool to explore traditional DSP topics and solve problems to gain insight. This greatly expands the range and complexity of problems that students can effectively study in the course. Since DSP applications are primarily algorithms implemented on a DSP processor or software, a fair amount of programming is required. Using interactive software such as MATLAB makes it possible to place more emphasis on learning new and difficult concepts than on programming algorithms. Interesting practical examples are discussed and useful problems are explored. Important Notice: Media content referenced within the product description or the product text may not be available in the ebook version.
Signal Processing FirstPearson College Division
With a novel, less classical approach to the subject, the authors have written a book with the conviction that signal processing should be taught to be fun. The treatment is therefore less focused on the mathematics and more on the conceptual aspects, the idea being to allow the readers to think about the subject at a higher conceptual level, thus building the foundations for more advanced topics. The book remains an engineering text, with the goal of helping students solve real-world problems. In this vein, the last chapter pulls together the individual topics as discussed throughout the book into an in-depth look at the development of an end-to-end communication system, namely, a modem for communicating digital information over an analog channel.