Signal Processing First James H Mcclellan

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.

This is the 22nd Volume in the series Memorial Tributes compiled by the National Academy of Engineering as a personal remembrance of the lives and outstanding achievements of its members and foreign associates. These volumes are intended to stand as an enduring record of the many contributions of engineers and engineering to the benefit of humankind. In most cases, the authors of the tributes are contemporaries or colleagues who had personal knowledge of the interests and the engineering accomplishments of the deceased. Through its members and foreign associates, the Academy carries out the responsibilities for which it was established in 1964. Under the charter of the National Academy of Sciences, the National Academy of Engineering was formed as a parallel organization of outstanding engineers. Members are elected on the basis of significant contributions to engineering theory and practice and to the literature of engineering or on the basis of demonstrated unusual accomplishments in the pioneering of new and developing fields of technology. The National Academies share a responsibility to advise the federal government on matters of science and technology. The expertise and credibility that the National Academy of Engineering brings to that task stem directly from the abilities, interests, and achievements of our members and foreign associates, our colleagues and friends, whose special gifts we remember in this book.

Presents the Bayesian approach to statistical signal processing for a variety of useful model sets This book aims to give readers a unified Bayesian treatment starting from the basics (Baye’s rule) to the more advanced (Monte Carlo sampling), evolving to the next-generation model-based techniques (sequential Monte Carlo sampling). This next edition incorporates a new chapter on “Sequential Bayesian Detection,” a new section on “Ensemble Kalman Filters” as well as an expansion of Case Studies that detail Bayesian solutions for a variety of applications. These studies illustrate Bayesian approaches to real-world problems incorporating detailed particle filter designs, adaptive particle filters and sequential Bayesian detectors. In addition to these major developments a variety of sections are expanded to “fill-in-the gaps” of the first edition. Here metrics for particle filter (PF) designs with emphasis on classical “sanity testing” lead to ensemble techniques as a basic requirement for performance analysis. The expansion of information theory metrics and their application to PF designs is fully developed and applied. These expansions of the book have been updated to provide a more cohesive discussion of Bayesian processing with examples and applications enabling the comprehension of alternative approaches to solving estimation/detection problems. The second edition of Bayesian Signal Processing features: “Classical” Kalman filtering for linear, linearized, and nonlinear systems; “modern” unscented and ensemble Kalman filters: and the “next-generation” Bayesian particle filters Sequential Bayesian detection techniques incorporating model-based schemes for a variety of real-world problems Practical Bayesian processor designs including comprehensive methods of performance analysis ranging from simple sanity testing and ensemble techniques to sophisticated information metrics New case studies on adaptive particle filtering and sequential Bayesian detection are covered detailing more Bayesian approaches to applied problem solving MATLAB® notes at the end of each chapter help readers solve complex problems using readily available software commands and point out other software packages available Problem sets included to test readers’ knowledge and help them put their new skills into practice Bayesian Signal Processing, Second Edition is written for all students, scientists, and engineers who investigate and apply signal processing to their everyday problems.

This supplement to any standard DSP text is one of the first books to successfully integrate the use of MATLAB® in the study of DSP concepts. In this book, 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. This updated second edition includes new homework problems and revises the scripts in the book, available functions, and m-files to MATLAB® V7.

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.

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.

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.

This book is intended to fill the gap between the "ideal precision" digital signal processing (DSP) that is widely taught, and the limited precision implementation skills that are commonly required in fixed-point processors and field programmable gate arrays (FPGAs). These skills are often neglected at the university level, particularly for undergraduates. We have attempted to create a resource both for a DSP elective course and for the practicing engineer with a need to understand fixed-point implementation. Although we assume a background in DSP, Chapter 2 contains a review of basic theory and Chapter 3 reviews random processes to support the noise model of quantization error. Chapter 4 details the binary arithmetic that underlies fixed-point processors and then introduces fractional format for binary numbers. Chapter 5 covers the noise model for quantization error and the effects of coefficient quantization in filters. Because of the numerical sensitivity of IIR filters, they are used extensively as an example system in both Chapters 5 and 6. Fortunately, the principles of dealing with limited precision can be applied to a wide variety of numerically sensitive systems, not just IIR filters. Chapter 6 discusses the problems of product roundoff error and various methods of scaling to avoid overflow. Chapter 7 discusses limit cycle effects and a few common methods for minimizing them. There are a number of simple exercises integrated into the text to allow you to test your understanding. Answers to the exercises are included in the footnotes. A number of MATLAB examples are provided in the text. They generally assume access to the Fixed-Point Toolbox. If you lack access to this software, consider either purchasing or requesting an evaluation license from The Mathworks. The code listed in the text and other helpful MATLAB code is also available at http://www.morganclaypool.com/page/padgett and http://www.rose-hulman.edu/padgett/fpsp. You will also find MATLAB exercises designed to demonstrate each of the four types of error discussed in Chapters 5 and 6. Simulink examples are also provided on the web site. Table of Contents: Getting Started / DSP Concepts / Random Processes and Noise / Fixed Point Numbers / Quantization Effects: Data and Coefficients / Quantization Effects - Round-Off Noise and Overflow / Limit Cycles

CD-ROM contains: Demonstrations -- Problem solutions.

Although Digital Signal Processing (DSP) has long been considered an electrical engineering topic, recent developments have also generated significant interest from the computer science community. DSP applications in the consumer market, such as bioinformatics, the MP3 audio format, and MPEG-based cable/satellite television have fueled a desire to understand this technology outside of hardware circles. Designed for upper division engineering and computer science students as well as practicing engineers and scientists, Digital Signal Processing Using MATLAB & Wavelets, Second Edition emphasizes the practical applications of signal processing. Over 100 MATLAB examples and wavelet techniques provide the latest applications of DSP, including image processing, games, filters, transforms, networking, parallel processing, and sound. This Second Edition also provides the mathematical processes and techniques needed to ensure an understanding of DSP theory. Designed to be incremental in difficulty, the book will benefit readers who are unfamiliar with complex mathematical topics or those limited in programming experience. Beginning with an introduction to MATLAB programming, it moves through filters, sinusoids, sampling, the Fourier transform, the z-transform and other key topics. Two chapters are dedicated to the discussion of wavelets and their applications. A CD-ROM (platform independent) accompanies the book and contains source code, projects for each chapter, and the figures from the book.

This unique introduction to the foundational concepts of cyber-physical systems (CPS) describes key design principles and emerging research trends in detail. Several interdisciplinary applications are covered, with a focus on the wide-area management of infrastructures including electric power systems, air transportation networks, and health care systems. Design, control and optimization of cyber-physical infrastructures are discussed, addressing security and privacy issues of networked CPS, presenting graph-theoretic and numerical approaches to CPS evaluation and monitoring, and providing readers with the knowledge needed to operate CPS in a reliable, efficient, and secure manner. Exercises are included. This is an ideal resource for researchers and graduate

students in electrical engineering and computer science, as well as for practitioners using cyber-physical systems in aerospace and automotive engineering, medical technology, and large-scale infrastructure operations.
"This book covers basic and the advanced approaches in the design and implementation of multirate filtering"--Provided by publisher.
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
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.

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

Photonics has long been considered an attractive substrate for next generation implementations of machine-learning concepts. Reservoir Computing tremendously facilitated the realization of recurrent neural networks in analogue hardware. This concept exploits the properties of complex nonlinear dynamical systems, giving rise to photonic reservoirs implemented by semiconductor lasers, telecommunication modulators and integrated photonic chips.

In the past few years Biomedical Engineering has received a great deal of attention as one of the emerging technologies in the last decade and for years to come, as witnessed by the many books, conferences, and their proceedings. Media attention, due to the applications-oriented advances in Biomedical Engineering, has also increased. Much of the excitement comes from the fact that technology is rapidly changing and new technological adventures become available and feasible every day. For many years the physical sciences contributed to medicine in the form of expertise in radiology and slow but steady contributions to other more diverse fields, such as computers in surgery and diagnosis, neurology, cardiology, vision and visual prosthesis, audition and hearing aids, artificial limbs, biomechanics, and biomaterials. The list goes on. It is therefore hard for a person unfamiliar with a subject to separate the substance from the hype. Many of the applications of Biomedical Engineering are rather complex and difficult to understand even by the not so novice in the field. Much of the hardware and software tools available are either too simplistic to be useful or too complicated to be understood and applied. In addition, the lack of a common language between engineers and computer scientists and their counterparts in the medical profession, sometimes becomes a barrier to progress.

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

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

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.

This previously included a CD. The CD contents can be accessed via World Wide Web.

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.

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 textbook gives a fresh approach to an introductory course in signal processing. Its unique feature is to alternate chapters on continuous-time (analog) and discrete-time (digital) signal processing concepts in a parallel and synchronized manner. This presentation style helps readers to realize and understand the close relationships between continuous and discrete time signal processing, and lays a solid foundation for the study of practical applications such as the analysis and design of analog and digital filters. The compendium provides motivation and necessary mathematical rigor. It generalizes the Fourier transform to Laplace and Z transforms, applies these transforms to linear system analysis, covers the time and frequency-domain analysis of differential and difference equations, and presents practical applications of these techniques to convince readers of their usefulness. MATLAB® examples are provided throughout, and over 100 pages of solved homework problems are included in the appendix. Contents: Introduction to Signal ProcessingDiscrete-Time Signals and OperationsContinuous-Time Signals and OperationsFrequency Analysis of Discrete-Time SignalsFrequency Analysis of Continuous-Time SignalsSampling Theory and PracticeFrequency Analysis of Discrete-Time SystemsFrequency Analysis of Continuous-Time SystemsZ-Domain Signal ProcessingS-Domain Signal ProcessingApplications of Z-Domain Signal ProcessingApplications of S-Domain Signal ProcessingAppendix: Solved Homework Problems Readership: Researchers, academics, professionals and undergraduate students in signal processing. Keywords: Signal Processing;Introduction;Analog and Digital;Practical;Applications;Solved Homework ProblemsReview:0

For the first time, a reference on the most relevant applications of adaptive filtering techniques. Top researchers in the field contributed chapters addressing applications in acoustics, speech, wireless and networking, where research is still very active and open.

This book describes the essential tools and techniques of statistical signal processing. At every stage theoretical ideas are linked to specific applications in communications and signal processing using a range of carefully chosen examples. The book begins with a development of basic probability, random objects, expectation, and second order moment theory followed by a wide variety of examples of the most popular random process models and their basic uses and properties. Specific applications to the analysis of random signals and systems for communicating, estimating, detecting, modulating, and other processing of signals are interspersed throughout the book. Hundreds of homework problems are included and the book is ideal for graduate students of electrical engineering and applied mathematics. It is also a useful reference for researchers in signal processing and communications.

Digital Signal Processing, Second Edition enables electrical engineers and technicians in the fields of biomedical, computer, and electronics engineering to master the essential fundamentals of DSP principles and practice. Many instructive worked examples are used to illustrate the material, and the use of mathematics is minimized for easier grasp of concepts. As such, this title is also useful to undergraduates in electrical engineering, and as a reference for science students and practicing engineers. The book goes beyond DSP theory, to show implementation of algorithms in hardware and software. Additional topics covered include adaptive filtering with noise reduction and echo cancellations, speech compression, signal sampling, digital filter realizations, filter design, multimedia applications, over-sampling, etc. More advanced topics are also covered, such as adaptive filters, speech compression such as PCM, u-law, ADPCM, and multi-rate DSP and over-sampling ADC. New to this edition: MATLAB projects dealing with practical applications added throughout the book New chapter (chapter 13) covering sub-band coding and wavelet transforms, methods that have become popular in the DSP field New applications included in many chapters, including applications of DFT to seismic signals, electrocardiography data, and vibration signals All real-time C programs revised for the TMS320C6713 DSK Covers DSP principles with emphasis on communications and control applications Chapter objectives, worked examples, and end-of-chapter exercises aid the reader in grasping key concepts and solving related problems Website with MATLAB programs for simulation and C programs for real-time DSP

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 statistical bootstrap is one of the methods that can be used to calculate estimates of a certain number of unknown parameters of a random process or a signal observed in noise, based on a random sample. Such situations are common in signal processing and the bootstrap is especially useful when only a small sample is available or an analytical analysis is too cumbersome or even impossible. This book covers the foundations of the bootstrap, its properties, its strengths and its limitations. The authors focus on bootstrap signal detection in Gaussian and non-Gaussian interference as well as bootstrap model selection. The theory developed in the book is supported by a number of useful practical examples written in MATLAB. The book is aimed at graduate students and engineers, and includes applications to real-world problems in areas such as radar and sonar, biomedical engineering and automotive engineering.

Confusing Textbooks? Missed Lectures? Not Enough Time? Fortunately for you, there's Schaum's Outlines. More than 40 million students have trusted Schaum's to help them succeed in the classroom and on exams. Schaum's is the key to faster learning and higher grades in every subject. Each Outline presents all the essential course information in an easy-to-follow, topic-by-topic format. You also get hundreds of examples, solved problems, and practice exercises to test your skills. This Schaum's Outline gives you Practice problems with full explanations that reinforce knowledge Coverage of the most up-to-date developments in your course field In-depth review of practices and applications Fully compatible with your classroom text, Schaum's highlights all the important facts you need to know. Use Schaum's to shorten your study time-and get your best test scores! Schaum's Outlines-Problem Solved.

FROM THE PREFACE: Many new useful ideas are presented in this handbook, including new finite impulse response (FIR) filter design techniques, half-band and multiplierless FIR filters, interpolated FIR (IFIR) structures, and error spectrum shaping.

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

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This book is a current, comprehensive design guide for your digital processing work with today's complex receiver systems. This book brings you up-to-date with the latest information on wideband electronic warfare receivers, the ADC testing procedure, frequency channelization and decoding schemes, and the operation of monobit receivers.