Digital Filters And Signal Processing In Electronic Engineering Theory Applications Architecture Code Woodhead Publishing Series In Electronic And Optical Materials

Digital Signal Processing 101: Everything You Need to Know to Get Started provides a basic tutorial on digital signal processing (DSP). Beginning with discussions of numerical representation and complex numbers and exponentials, it goes on to explain difficult concepts such as sampling, aliasing, imaginary numbers, and frequency response. It does so using easy-to-understand examples with minimum mathematics. In addition, there is an overview of the DSP functions and implementation used in several DSP-intensive fields or applications, from error correction to CDMA mobile communication to airborne radar systems. This book has been updated to include the latest developments in Digital Signal Processing, and has eight new chapters on: Automotive Radar Signal Processing Space-Time Adaptive Processing Radar Field Orientated Motor Control Matrix Inversion algorithms GPUs for computing Machine Learning Entropy and Predictive Coding Video compression Features eight new chapters on Automotive Radar Signal Processing, Space-Time Adaptive Processing Radar, Field Orientated Motor Control, Matrix Inversion algorithms, GPUs for computing, Machine Learning, Entropy and Predictive Coding, and Video compression Provides clear examples and a non-mathematical approach to get you up to speed quickly Includes an overview of the DSP functions and implementation used in typical DSP-intensive applications, including error correction, CDMA mobile communication, and radar systems

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.

Introductory text examines role of digital filtering in many applications, particularly computers. Focus on linear signal processing; some consideration of roundoff effects, Kalman filters. Only calculus, some statistics required.

Digital signal processing is commonplace in most electronics including MP3 players, HDTVs, and phones, just to name a few of the applications. The engineers creating these devices are in need of essential information at a moment's notice. The Instant Access Series provides all the critical content that a signal or communications engineer needs in his or her daily work. This book provides an introduction to DSPs as well as succinct overviews of linear systems, digital filters, and digital compression. This book is filled with images, figures, tables, and easy to find tips and tricks for the engineer that needs material fast to complete projects to deadline. Tips and tricks feature that will help engineers get info fast and move on to the next issue Easily searchable content complete with tabs, chapter table of contents, bulleted lists, and boxed features Just the essentials, no need to page through material not needed for the current project

&Quot;With a strong focus on basic principles and applications, this thoroughly up-to-date text provides a solid foundation in the concepts, methods, and algorithms of digital signal processing. Key topics such as spectral analysis, discrete-time systems, the sampling process, and digital filter design are all covered in well-illustrated detail.". "Filled with examples and problems that can be worked in MATLAB or the author's DSP software, D-Filter, Digital Signal Processing offers a fully interactive approach to successfully mastering DSP.". "Accessible and comprehensive, this resource covers the essentials of DSP theory and practice."--BOOK JACKET.

This text provides a broad introduction to the field of digital signal processing and contains sufficient material for a two-semester sequence in this multifaceted subject. It is also written with the practicing engineer or scientist in mind, having many observations and examples of practical significance drawn from the author's industrial experience. The first semester, at the junior, senior, or first-year graduate level, could cover chapters 2 through 7 with topics perhaps from chapters 8 and 9, depending upon the background of the students. The only requisite background is linear systems theory for continuous-time systems, including Fourier and Laplace trans forms. Many students will also have had some previous exposure to discrete-time systems, in which case chapters 2 through 4 may serve to review and expand that preparation. Note, in particular, that knowledge of probability theory and random processes is not required until chapters 10 and 11, except for section 7. 6 on the periodogram. A second, advanced course could utilize material from chapters 8 through 13. A comprehensive one-semester course for suitably prepared graduate students might cover chapters 4 through 9 and additional topics from chapters 10 through 13. Sections marked with a dagger Ct) cover advanced or specialized topics and may be skipped without loss of continuity. Notable features of the book include the following: 1. Numerous useful filter examples early in the text in chapters 4 and 5. 2. State-space representation and structures in chapters 4 and 11.

Sampled Data Systems - ADCs for DSP Applications - DACs for DSP Applications - Fast Fourier Transforms - Digital Filters - DSP Hardware - Interfacing to DSPs - DSP Applications - Hardware Design Techniques.

From industrial and teaching experience the authors provide a blend of theory and practice of digital signal processing (DSP) for advanced undergraduate and post-graduate engineers reading electronics. This fast-moving, developing area is driven by the information technology revolution. It is a source book in research and development for embedded system design engineers, designers in real-time computing, and applied mathematicians who apph DSP techniques in telecommunications, aerospace (control systems), satellite communications, instrumentation, and medical technology (ultrasound and magnetic resonance imaging). The book is particularly useful at the hardware end of DSP, with its emphasis on practical I)SP devices and the integration of basic processes with appropriate software. It is unique to find in one volume the implementation of the equations as algorithms, not only in \IATLAB but right up to a working DSP-based scheme. Other relevant architectural features include number representations, multiply-accumulate, special addressing modes, zero overhead iteration schemes. and single and multiple nlicroprocessors which will allow the readers to compare and understand both current processors and future DSP developments. Fundamental signal processing procedures are introduced and developed: also convolution. correlation, the Discrete Fourier Transform and its fast computation algorithms. Then follo finite impulse response (FIR) filters, infinite impulse response (IIR) filters, multirate filters, adaptive filters, and topics from communication and control. I)esign examples are given in all of these cases, taken through an algorithm testing stage using MATLAB.

The design of the latter. using C language models, is explained together with the experimental results of real time integer implementations. Academic prerequisites are first and second year university mathematics, an introductor knowledge of circuit theor 'and microprocessors. and C Language. Provides an unusual blend of theory and practice of digital signal processing (DSP) Discusses fundamental signal processing procedures, convolution, correlation, the Discrete Fourier Transform and its fast computation algorithms Includes number representations, multiply-accumulate, special addressing modes, zero overhead iteration schemes, and single and multiple instructions

This text emphasizes the intricate relationship between adaptive filtering and signal analysis - highlighting stochastic processes, signal representations and properties, analytical tools, and implementation methods. This second edition includes new chapters on adaptive techniques in communications and rotation-based algorithms. It provides practical applications in information, estimation, and circuit theories.

This book presents recent advances in DSP to simplify, or increase the computational speed of, common signal processing operations. The topics describe clever DSP tricks of the trade not covered in conventional DSP textbooks. This material is practical, real-world, DSP tips and tricks as opposed to the traditional highly-specialized, math-intensive, research subjects directed at industry researchers and university professors. This book goes well beyond the standard DSP fundamentals textbook and presents new, but tried-and-true, clever implementations of digital filter design, spectrum analysis, signal generation, high-speed function approximation, and various other DSP functions.

Introduction to digital filters. Finite impulse-response filters. Design of linear-phase finite impulse-response. Minimum-phas and complex approximation. Implementation of finite impulse-response filters. Properties of infinite impulse-response filters. Design of infinite impulse-response filters. Implementation of infinite impulse-response filters. Programs.

A complete up-to-date reference for advanced analog and digital IIR filter design rooted in elliptic functions. "Revolutionary" in approach, this book opens up completely new vistas in basic analog and digital IIR filter design--regardless of the technology. By introducing exceptionally elegant and creative mathematical stratagems (e.g., accurate replacement of Jacobi elliptic functions by functions comprising polynomials, square roots, and logarithms), optimization routines carried out with symbolic analysis by "Mathematica," and the advance filter design software of MATLAB, it shows readers how to design many types of filters that cannot be designed using conventional techniques. The filter design algorithms can be directly programed in any language or environment such as Visual BASIC, Visual C, Maple, DERIVE, or MathCAD. Signals; Systems; Transforms; Classical Analog Filter Design; Advanced Analog Filter Design Case Studies; Advanced Analog Filter Design Algorithms; Multi-criteria Optimization of Analog Filter Designs; Classical Digital Filter Design; Advanced Digital Filter Design Case Studies; Advanced Digital Filter Design Algorithms; Multi-criteria Optimization of Digital Filter Designs; Elliptic Functions; Elliptic Rational Function.

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

A mathematically rigorous but accessible treatment of digital signal processing that intertwines basic theoretical techniques with hands-on laboratory instruction is provided by this book. The book covers various aspects of the digital signal processing (DSP) "problem". It begins with the analysis of discrete-time signals and explains sampling and the use of the discrete and fast Fourier transforms. The second part of the book — covering digital to analog and analog to digital conversion — provides a practical interlude in the mathematical content before Part III lays out a careful development of the Z-transform and the design and analysis of digital filters.

Basic Digital Signal Processing describes the principles of digital signal processing and experiments with BASIC programs involving the fast Fourier theorem (FFT). The book reviews the fundamentals of the BASIC program, continuous and discrete time signals including analog signals, Fourier analysis, discrete Fourier transform, signal energy, power. The text also explains digital signal processing involving digital filters, linear time-variant systems, discrete time unit impulse, discrete-time convolution, and the alternative structure for second order infinite impulse response (IIR) sections. The text notes the importance of the effects of analogue/digital interfaces, of the aspects such as sampling and quantization of the analogue input, as well as the reconstruction of an analogue output from the processed digital signal. Digital filter design consists of two separate operations: 1) approximation—the determination of a realizable system function from some idealized 'target'; and 2) realization—the formulation of a signal flow graph and its implementation in hardware or software. Digital signal processing employs the FFT, a number of efficient algorithms that compute the discrete Fourier transform and the inverse discrete Fourier transform. The programmer can run the FFT methods using some BASIC programs. The book can prove useful for programmers, computer engineers, computer technicians, and computer instructors dealing with many aspects of computers such as networking, engineering or design.

The subject of Digital Signal Processing (DSP) is enormously complex, involving many concepts, probabilities, and signal processing that are woven together in an intricate manner. To cope with this scope and complexity, many DSP texts are often organized around the “numerical examples” of a communication system. With such organization, readers can see through the complexity of DSP, they learn about the distinct concepts and protocols in one part of the communication system while seeing the big picture of how all parts fit together. From a pedagogical perspective, our personal experience has been that such approach indeed works well. Based on the authors’ extensive experience in teaching and research, Digital Signal Processing: A Breadth-First Approach is written with the reader in mind. The book is intended for a course on digital signal processing, for seniors and undergraduate students. The subject has high popularity in the field of electrical and computer engineering, and the authors consider all the needs and tools used in analysis and design of discrete time systems for signal processing. Key features of the book include: • The extensive use of MATLAB based examples to illustrate how to solve signal processing problems. The textbook includes a wealth of problems, with solutions • Worked-out examples have been included to explain new and difficult concepts, which help to expose the reader to real-life signal processing problems • The inclusion of FIR and IIR filter design further enrich the contents.

Intended to anyone interested in numerical computing and data science: students, researchers, teachers, engineers, analysts, hobbyists... Basic knowledge of Python/NumPy is recommended. Some skills in mathematics will help you understand the theory behind the computational methods.

In addition to its thorough coverage of DSP design and programming techniques, Smith also covers the operation and usage of DSP chips. He uses Analog Devices' popular DSP chip family as design examples. Covers all major DSP topics Full of insider information and shortcuts Basic techniques and algorithms explained without complex numbers

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

The second, strongly enlarged edition of the textbook gives a substantial insight into the characteristics and the design of digital filters. It briefly introduces to the theory of continuous-time systems and the design methods for analog filters. Time-discrete systems, the basic structures of digital filters, sampling theorem, and the design of IIR filters are widely discussed. The author devotes important parts to the design of non-recursive filters and the effects of finite register length. The explanation of techniques like oversampling and noise shaping conclude the book. The author has substantially updated all chapters and added some important topics like Allpass filters. With an emphasize put on the practical implementation of theoretical concepts, the book is a reference for advanced students as well as practicing engineers.

This textbook provides comprehensive coverage for courses in the basics of design and implementation of digital filters. The book assumes only basic knowledge in digital signal processing and covers state-of-the-art methods for digital filter design and provides a simple route for the readers to design their own filters. The advanced mathematics that is required for the filter design is minimized by providing an extensive MATLAB toolbox with over 300 files. The book presents over 200 design examples with MATLAB code and over 300 problems to be solved by the reader. The students can design and modify the code for their use. The book and the design examples cover almost all known design methods of frequency-selective digital filters as well as some of the authors’ own, unique techniques.

The book is not an exposition on digital signal processing (DSP) but rather a treatise on digital filters. The material and coverage is comprehensive, presented in a consistent that first develops topics and subtopics in terms it their purpose, relationship to other core ideas, theoretical and conceptual framework, and finally instruction in the implementation of digital filter devices. Each major study is supported by Matlab-enabled activities and examples, with each Chapter culminating in a comprehensive design case study.

Digital Filters and Signal Processing, Third Edition ... with MATLAB Exercises presents a general survey of digital signal processing concepts, design methods, and implementation considerations, with an emphasis on digital filters. It is suitable as a textbook for senior undergraduate or first-year graduate courses in digital signal processing. While mathematically rigorous, the book stresses an intuitive understanding of digital filters and signal processing systems, with numerous realistic and relevant examples. Hence, practicing engineers and scientists will also find the book to be a most useful reference. The Third Edition contains a substantial amount of new material including, in particular, the addition of MATLAB exercises to deepen the students' understanding of basic DSP principles and increase their proficiency in the application of these principles. The use of the exercises is not mandatory, but is highly recommended. Other new features include: normalized frequency utilized in the DTFT, e.g., X(ejomega); new computer generated drawings and MATLAB plots throughout the book; Chapter 6 on sampling the DTFT has been completely rewritten; expanded coverage of Types I-IV linear-phase FIR filters; new material on power and doubly-complementary filters; new section on quadrature-mirror filters and their application in filter banks; new section on the design of maximally-flat FIR filters; new section on roundoff-noise reduction using error feedback; and many new problems added throughout.

Nonlinear Digital Filters provides an easy to understand overview of nonlinear behavior in digital filters, showing how it can be utilized or avoided when operating nonlinear digital filters. It gives techniques for analyzing discrete-time systems with discontinuous linearity, enabling the analysis of other nonlinear discrete-time systems, such as sigma delta modulators, digital phase lock loops, and turbo coders. It uses new methods based on symbolic dynamics, enabling the engineer to easily operate reliable nonlinear digital filters. It gives practical, 'real-world' applications of nonlinear digital filters and contains many examples. The book is ideal for professional engineers working with signal processing applications, as well as advanced undergraduates and graduates conducting a nonlinear filter analysis project. Uses new methods based on symbolic dynamics, enabling the engineer more easily to operate reliable nonlinear digital filters Gives practical, "real-world" applications of nonlinear digital filter Includes many examples.

Analysis, design, and realization of digital filters have experienced major developments since the 1970s, and have now become an integral part of the theory and practice in the field of contemporary digital signal processing. Digital Filter Design and Realization is written to present an up-to-date and comprehensive account of the analysis, design, and realization of digital filters. It is intended to be used as a text for graduate students as well as a reference book for practitioners in the field. Prerequisites for this book include basic knowledge of calculus, linear algebra, signal analysis, and linear system theory. Technical topics discussed in the book include: Discrete-Time Systems and z-TransformationStability and Coefficient SensitivityState-Space ModelsFIR Digital Filter DesignFrequency-Domain Digital Filter DesignTime-Domain Digital Filter DesignInterpolated and Frequency-Response-Masking FIR Digital Filter DesignComposite Digital Filter DesignFinite Word Length EffectsCoefficient Sensitivity Analysis and MinimizationError Spectrum ShapingRoundoff Noise Analysis and MinimizationGeneralized Transposed Direct-Form IIBlock-State Realization

Provides a thorough and accessible introduction to the fast-growing area of multirate digital signal processing covering both the fundamental theory and the practical applications. The key characteristic of multirate algorithms is their high computational efficiency, and hence their increasing implementation in a range of applications from digital audio broadcasting to multi-carrier data transmission and subband speech coding. This book gives a comprehensive analysis of the subject and features include: * A summary of the key properties of those filters which employ multirate techniques including cascaded multirate filters, multirate complementary filters, and interpolated FIR filters * An assessment of the properties of various digital filter banks, such as quadratur mirror, parunitary, biorthogonal, modulated, polyphase, and multicomplementary filter banks * Design methodologies for multirate filters and filter banks * An examination of the discrete wavelet transform using filter banks, the construction of wavelets and examples of wavelet systems * A complete overview of current applications and a look ahead towards the future developments in the field This book will be invaluable for advanced students in electronics and computer science. It will also be useful for practising electronics and communications engineers and physicists working in industry.

Digital signal processing is ubiquitous. It is an essential ingredient in many of today’s electronic devices, ranging from medical equipment to weapon systems. It makes the difference between dumb and intelligent systems. This book is organized into five parts: (1) Introduction, which contains an account of Prof. Constantinides’ contribution to the field and brief summaries of the remaining chapters of this festschrift, (2) Digital Filters and Transforms, which covers efficient digital filtering techniques for improving signal quality, (3) Signal Processing, which provides an insight into fundamental theories, (4) Communications, which deals with some important applications of signal processing techniques, and (5) Finale, which contains a discussion on the impact of digital signal processing on our society and the closing remarks on this festschrift.

A digital filter can be pictured as a "black box" that accepts a sequence of numbers and emits a new sequence of numbers. In digital audio signal processing applications, such number sequences usually represent sounds. For example, digital filters are used to implement graphic equalizers and other digital audio effects. This book is a gentle introduction to digital filters, including mathematical theory, illustrative examples, some audio applications, and useful software starting points. The theory treatment begins at the high-school level, and covers fundamental concepts in linear systems theory and digital filter analysis. Various "small" digital filters are analyzed as examples, particularly those commonly used in audio applications. Matlab programming examples are emphasized for illustrating the use and development of digital filters in practice.

An engineer's introduction to concepts, algorithms, and advancements in Digital Signal Processing. This lucidly written resource makes extensive use of real-world examples as it covers all the important design and engineering references.

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.

Dealing with digital filtering methods for 1-D and 2-D signals,this book provides the theoretical background in signal processing,covering topics such as the z-transform, Shannon sampling theoremand fast Fourier transform. An entire chapter is devoted to thedesign of time-continuous filters which provides a usefulpreliminary step for analog-to-digital filter conversion. Attention is also given to the main methods of designing finiteimpulse response (FIR) and infinite impulse response (IIR) filters.Bi-dimensional digital filtering (image filtering) is investigatedand a study on stability analysis, a very useful tool whenimplementing IIR filters, is also carried out. As such, it willprovide a practical and useful guide to those engaged in signalprocessing.

Up-to-date digital filter design principles, techniques, and applications Written by a Life Fellow of the IEEE, this comprehensive textbook teaches digital filter design, realization, and implementation and provides detailed illustrations and real-world applications of digital filters to signal preocessing. Digital Filters: Analysis, Design, and Signal Processing Applications provides a solid foundation in the fundamentals and concepts of DSP and continues with state-of-the-art methodologies and algorithms for the design of digital filters. You will get clear explanations of key topics such as spectral analysis, discrete-time systems, and the sampling process.. This hands-on resource is supported by a rich collection of online materials which include PDF presentations, detailed solutions of the end-of-chapter problems, MATLAB programs that can be used to analyze and design digital filters of professional quality, and also the author's DSP software D-Filter. Coverage includes: •Discrete-time systems •The Fourier series and transform •The Z transform •Application of transform theory to systems •The sampling process •The discrete Fourier transform •The window technique •Realization of digital filters •Design of recursive and nonrecursive filters •Approximations for analog filters •Recursive filters satisfying prescribed specifications •Effects of finite word length on digital filters •Design of recursive and nonrecursive filters using optimization methods •Wave digital filters •Signal processing applications

Digital signal processing (DSP) has been applied to a very wide range of applications. This includes voice processing, image processing, digital communications, the transfer of data over the internet, image and data compression, etc. Engineers who develop DSP applications today, and in the future, will need to address many implementation issues including mapping algorithms to computational structures, computational efficiency, power dissipation, the effects of finite precision arithmetic, throughput and hardware implementation. It is not practical to cover all of these in a single text. However, this text emphasizes the practical implementation of DSP algorithms as well as the fundamental theories and analytical procedures that form the basis for modern DSP applications. Digital Signal Processing: Principles, Algorithms and System Design provides an introduction to the principals of digital signal processing along with a balanced analytical and practical treatment of algorithms and applications for digital signal processing. It is intended to serve as a suitable text for a one semester junior or senior level undergraduate course. It is also intended for use in a following one semester first-year graduate level course in digital signal processing. It may also be used as a reference by professionals involved in the design of embedded computer systems, application specific integrated circuits or special purpose computer systems for digital signal processing, multimedia, communications, or image processing. Covers fundamental theories and analytical procedures that form the basis of modern DSP Shows practical implementation of DSP in software and hardware Includes Matlab for design and implementation of signal processing algorithms and related discrete time systems Bridges the gap between reference texts and the knowledge needed to implement DSP applications in software or hardware

James D. Broesch is a staff engineer for General Atomics, where he is responsible for the design and development of several advanced control systems used on fusion control programs. He also teaches classes in signal processing and hardware design at the University of California-San Diego. · Integrated book/software package allows readers to simulate digital signal processing (DSP) situations and experiment with effects of different DSP techniques. · Gives an applications-oriented approach to DSP instead of a purely mathematical one. · The accompanying CD includes a DSP "calculator" to help solve design problems