Signal Processing First Mcclellan Solutions Manual

BeagleBone is an inexpensive web server, Linux desktop, and electronics hub that includes all the tools you need to create your own projects—whether it's robotics, gaming, drones, or software-defined radio. If you're new to BeagleBone Black, or want to explore more of its capabilities, this cookbook provides scores of recipes for connecting and talking to the physical world with this credit-card-sized computer. All you need is minimal familiarity with computer programming and electronics. Each recipe includes clear and simple wiring diagrams and example code to get you started. If you don't know what BeagleBone Black is, you might decide to get one after scanning these recipes. Learn how to use BeagleBone to interact with the physical world Connect force, light, and distance sensors Spin servo motors, stepper motors, and DC motors Flash single LEDs, strings of LEDs, and matrices of LEDs Manage real-time input/output (I/O) Work at the Linux I/O level with shell commands, Python, and C Compile and install Linux kernels Work at a high level with JavaScript and the BoneScript library Expand BeagleBone's functionality by adding capes Explore the Internet of Things

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.

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

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. 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

Now in a new edition—the most comprehensive, hands-on introduction to digital signal processing The first edition of Digital Signal Processing and Applications with the TMS320C6713 and TMS320C6416 DSK is widely accepted as the most extensive text available on the hands-on teaching of Digital Signal Processing (DSP). Now, it has been fully updated in this valuable Second Edition to be compatible with the latest version (3.1) of Texas Instruments Code Composer Studio (CCS) development environment. Maintaining the original's comprehensive, hands-on approach that has made it an instructor's favorite, this new edition also features: Added program examples that illustrate DSP concepts in real-time and in the laboratory Expanded coverage of analog input and output New material on frame-based processing A revised chapter on IIR, which includes a number of floating-point example programs that explore IIR filters more comprehensively More extensive coverage of DSP/BIOS All programs listed in the text—plus additional applications—which are available on a companion CD-ROM No other book provides such an extensive or comprehensive set of program examples to aid instructors in teaching DSP in a laboratory using audio frequency signals—making this an ideal text for DSP courses at the senior undergraduate and postgraduate levels. It also serves as a valuable resource for researchers, DSP developers, business managers, and technology solution providers who are looking for an overview and examples of DSP algorithms implemented using the TMS320C6713 and TMS320C6416 DSK.

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.

Efficient signal processing algorithms are important for embedded and power-limited applications since, by reducing the number of computations, power consumption can be reduced significantly. Similarly, efficient algorithms are also critical to very large scale applications such as video processing and four-dimensional medical imaging. This self-contained guide, the only one of its kind, enables engineers to find the optimum fast algorithm for a specific application. It presents a broad range of computationally-efficient algorithms, describes their structure and implementation, and compares their relative strengths for given problems. All the necessary background mathematics is included and theorems are rigorously proved, so all the information needed to learn and apply the techniques is provided in one convenient guide. With this practical reference, researchers and practitioners in electrical engineering, applied mathematics, and computer science can reduce power dissipation for low-end applications of signal processing, and extend the reach of high-end applications.

Signal Processing FirstSignal Processing FirstPearson College Division

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

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

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.

Signal Processing: A Mathematical Approach is designed to show how many of the mathematical tools the reader knows can be used to understand and employ signal processing techniques in an applied environment. Assuming an advanced undergraduate- or graduate-level understanding of mathematics-including familiarity with Fourier series, matrices, probab

A fully updated second edition of the excellent Digital Audio Signal Processing Well established in the consumer electronics industry, Digital Audio Signal Processing (DASP) techniques are used in audio CD, computer music and multi-media components. In addition, the applications afforded by this versatile technology now range from real-time signal processing to room simulation. Digital Audio Signal Processing, Second Edition covers the latest signal processing algorithms for audio processing. Every chapter has been completely revised with an easy to understand introduction into the basics and exercises have been included for self testing. Additional Matlab files and Java Applets have been provided on an accompanying website, which support the book by easy to access application examples. Key features include: A thoroughly updated and revised second edition of the popular Digital Audio Signal Processing, a comprehensive coverage of the topic as whole Provides basic principles and fundamentals for Quantization, Filters, Dynamic Range Control, Room Simulation, Sampling Rate Conversion, and Audio Coding Includes detailed accounts of studio technology, digital transmission systems, storage media and audio components for home entertainment Contains precise algorithm description and applications Provides a full account of the techniques of DASP showing their theoretical foundations and practical solutions Includes updated computer-based exercises, an accompanying website, and features Web-based Interactive JAVA-Applets for audio processing This essential guide to digital audio signal processing will serve as an invaluable reference to audio engineering professionals, R&D engineers, researchers in consumer electronics industries and academia, and Hardware and Software developers in IT companies. Advanced students studying multi-media courses will also find this guide of interest.

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.

Considering the rapid evolution of digital signal processing (DSP), those studying this field require an easily understandable text that complements practical software and hardware applications with sufficient coverage of theory. Designed to keep pace with advancements in the field and elucidate lab work, Digital Signal Processing Laboratory, Second Edition was developed using material and student input from courses taught by the author. Contains a new section on digital filter structure Honed over the past several years, the information presented here reflects the experience and insight the author gained on how to convey the subject of DSP to senior undergraduate and graduate students coming from varied subject backgrounds. Using feedback from those students and faculty involved in these courses, this book integrates simultaneous training in both theory and practical software/hardware aspects of DSP. The practical component of the DSP course curriculum has proven to greatly enhance understanding of the basic theory and principles. To this end, chapters in the text contain sections on:

Theory—Explaining the underlying mathematics and principles Problem solving—Offering an ample amount of workable problems for the reader Computer laboratory—Featuring programming examples and exercises in MATLAB® and Simulink® Hardware laboratory—Containing exercises that employ test and measurement equipment, as well as the Texas Instruments TMS320C6711DSP Starter Kit The text covers the progression of the Discrete and Fast Fourier transforms (DFT and FFT). It also addresses Linear Time-Invariant (LTI) discrete-time signals and systems, as well as the mathematical tools used to describe them. The author includes appendices that give detailed descriptions of hardware along with instructions on how to use the equipment featured in the book.

Quickly Engages in Applying Algorithmic Techniques to Solve Practical Signal Processing Problems With its active, hands-on learning approach, this text enables readers to master the underlying principles of digital signal processing and its many applications in industries such as digital television, mobile and broadband communications, and medical/scientific devices. Carefully developed MATLAB® examples throughout the text illustrate the mathematical concepts and use of digital signal processing algorithms. Readers will develop a deeper understanding of how to apply the algorithms by manipulating the codes in the examples to see their effect. Moreover, plenty of exercises help to put knowledge into practice solving real-world signal processing challenges. Following an introductory chapter, the text explores: Sampled signals and digital processing Random signals Representing signals and systems Temporal and spatial signal processing Frequency analysis of signals Discrete-time filters and recursive filters Each chapter begins with chapter objectives and an introduction. A summary at the end of each chapter ensures that one has mastered all the key concepts and techniques before progressing in the text. Lastly, appendices listing selected web resources, research papers, and related textbooks enable the investigation of individual topics in greater depth. Upon completion of this text, readers will understand how to apply key algorithmic techniques to address practical signal processing problems as well as develop their own signal processing algorithms. Moreover, the text provides a solid foundation for evaluating and applying new digital processing signal techniques as they are developed.

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.

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.
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
Get a working knowledge of digital signal processing for computer science applications The field of digital signal processing (DSP) is rapidly exploding, yet most books on the subject do not reflect the real world of algorithm development, coding for applications, and software engineering. This important new work fills the gap in the field, providing computer professionals with a comprehensive introduction to those aspects of DSP essential for working on today's cutting-edge applications in speech compression and recognition and modem design. The author walks readers through a variety of advanced topics, clearly demonstrating how even such areas as spectral analysis, adaptive and nonlinear filtering, or communications and speech signal processing can be made readily accessible through clear presentations and a practical hands-on approach. In a light, reader-friendly style, Digital Signal Processing: A Computer Science Perspective provides: * A unified treatment of the theory and practice of DSP

at a level sufficient for exploring the contemporary professional literature * Thorough coverage of the fundamental algorithms and structures needed for designing and coding DSP applications in a high level language * Detailed explanations of the principles of digital signal processors that will allow readers to investigate assembly languages of specific processors * A review of special algorithms used in several important areas of DSP, including speech compression/recognition and digital communications * More than 200 illustrations as well as an appendix containing the essential mathematical background

Master the basic concepts and methodologies of digital signal processing with this systematic introduction, without the need for an extensive mathematical background. The authors lead the reader through the fundamental mathematical principles underlying the operation of key signal processing techniques, providing simple arguments and cases rather than detailed general proofs. Coverage of practical implementation, discussion of the limitations of particular methods and plentiful MATLAB illustrations allow readers to better connect theory and practice. A focus on algorithms that are of theoretical importance or useful in real-world applications ensures that students cover material relevant to engineering practice, and equips students and practitioners alike with the basic principles necessary to apply DSP techniques to a variety of applications. Chapters include worked examples, problems and computer experiments, helping students to absorb the material they have just read. Lecture slides for all figures and solutions to the numerous problems are available to instructors.

Applied Underwater Acoustics meets the needs of scientists and engineers working in underwater acoustics and graduate students solving problems in, and preparing theses on, topics in underwater acoustics. The book is structured to provide the basis for rapidly assimilating the essential underwater acoustic knowledge base for practical application to daily research and analysis. Each chapter of the book is self-supporting and focuses on a single topic and its relation to underwater acoustics. The chapters start with a brief description of the topic’s physical background, necessary definitions, and a short description of the applications, along with a roadmap to the chapter. The subtopics covered within individual subchapters include most frequently used equations that describe the topic. Equations are not derived, rather, assumptions behind equations and limitations on the applications of each equation are emphasized. Figures, tables, and illustrations related to the sub-topic are presented in an easy-to-use manner, and examples on the use of the equations, including appropriate figures and tables are also included. Provides a complete and up-to-date treatment of all major subjects of underwater acoustics Presents chapters written by recognized experts in their individual field Covers the fundamental knowledge scientists and engineers need to solve problems in underwater acoustics Illuminates, in shorter sub-chapters, the modern applications of underwater acoustics that are described in worked examples Demands no prior knowledge of underwater acoustics, and the physical principles and mathematics are designed to be readily understood by scientists, engineers, and graduate students of underwater acoustics Includes a comprehensive list of literature references for each chapter

This textbook provides engineering students with instruction on processing signals encountered in speech, music, and wireless communications using software or hardware by employing basic mathematical methods. The book starts with an overview of signal processing, introducing readers to the field. It goes on to give instruction in converting continuous time signals into digital signals and discusses various methods to process the digital signals, such as filtering. The author uses MATLAB throughout as a user-friendly software tool to perform various digital signal processing algorithms and to simulate real-time systems. Readers learn how to convert analog signals into digital signals; how to process these signals using software or hardware; and how to write algorithms to perform useful operations on the acquired signals such as filtering, detecting digitally modulated signals, correcting channel distortions, etc. Students are also shown how to convert MATLAB codes into firmware codes. Further, students will be able to apply the basic digital signal processing techniques in their workplace. The book is based on the author's popular online course at University of California, San Diego.

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.

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.

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.

This new book by Ken Steigliz offers an informal and easy-to-understand introduction to digital signal processing, emphasizing digital audio and applications to computer music. A DSP Primer covers important topics such as phasors and tuning forks; the wave equation; sampling and quantizing; feedforward and feedback filters; comb and string filters; periodic sounds; transform methods; and filter design. Steiglitz uses an intuitive and qualitative approach to develop the mathematics critical to understanding DSP. A DSP Primer is written for a broad audience including: Students of DSP in Engineering and Computer Science courses. Composers of computer music and those who work with digital sound. WWW and Internet developers who work with multimedia. General readers interested in science that want an introduction to DSP. Features: Offers a simple and uncluttered step-by-step approach to DSP for first-time users, especially beginners in computer music. Designed to provide a working knowledge and understanding of frequency domain methods, including FFT and digital filtering. Contains thought-provoking questions and suggested experiments that help the reader to understand and apply DSP theory and techniques.

This updated edition gives readers hands-on experience in real-time DSP using a practical, step-by-step framework that also incorporates demonstrations, exercises, and problems, coupled with brief overviews of applicable theory and MATLAB applications. Organized in three sections that cover enduring fundamentals and present practical projects and invaluable appendices, this new edition provides support for the most recent and powerful of the inexpensive DSP development boards currently available from Texas Instruments: the OMAP-L138 LCDK. It includes two new real-time DSP projects, as well as three new appendices: an introduction to the Code Generation tools available with MATLAB, a guide on how to turn the LCDK into a portable battery-operated device, and a comparison of the three DSP boards directly supported by this edition.

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.

CD-ROM contains: Demonstrations -- Problem solutions.

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

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

Sensors arrays are used in diverse applications across a broad range of disciplines. Regardless of the application, however, the tools of sensor array signal processing remain the same. Furthermore, whether your interest is in acoustic, seismic, mechanical, or electromagnetic wavefields, they all have a common mathematical framework. Mastering this framework and those tools lays a strong foundation for more specialized study and research. Sensor Array Signal Processing helps build that foundation. It unravels the underlying principles of the subject without reference to any particular application. Instead, the author focuses on the common threads that exist in wavefield analysis. After introducing the basic equations governing different wavefields, the treatment includes topics from simple beamformation, spatial filtering, and high resolution DOA estimation to imaging and reflector mapping. It studies different types of sensor configurations, but focuses on the uniform linear and circular arrays-the most useful configurations for understanding array systems in practice. Unique in its approach, depth, and quantitative focus, Sensor Array Signal Processing offers the ideal starting point and an outstanding reference for those working or interested in medical imaging, astronomy, radar, communications, sonar, seismology-any field that studies propagating wavefields. Its clear exposition, numerical examples, exercises, and wide applicability impart a broad picture of array signal processing unmatched by any other text on the market.