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THE definitive, authoritative book on DSP -- ideal for those with an introductory-level knowledge of signals and systems. Written by prominent, DSP pioneers, it provides thorough treatment of the fundamental theorems and properties of discrete-time linear systems, filtering, sampling, and discrete-time Fourier Analysis. By focusing on the general and universal concepts in discrete-time signal processing, it remains vital and relevant to the new challenges arising in the field -- "without" limiting itself to specific technologies with relatively short life spans. FEATURES NEW--Provides a new chapter organization. NEW--Material on: Multi-rate filtering banks. The discrete cosine transform. Noise-shaping sampling strategies. NEW--Includes several dozen new problem-solving examples that not only illustrate key points, but demonstrate approaches to typical problems related to the material. NEW--Contains a wealth of "combat tested" problems which are the best produced over decades of undergraduate and graduate signal processing classes at MIT and Georgia Tech. NEW--Problems are completely reorganized by level of difficulty into separate categories: Basic Problems with Answers to allow the user to check their results, but not solutions (20 per chapter). Basic Problems -- without answers. Advanced Problems. Extension Problems -- start from the discussion in the book and lead the reader beyond to glimpse some advanced areas of signal processing. Covers the history of discrete-time signal processing as well as contemporary developments in the field. Discusses the wide range of present and future applications of the technology. Focuses on the general and universal concepts in discrete-time signal processing. Offers a wealth of problems and examples. This text presents a definitive treatise on discrete-time signal processing. It provides thorough treatment of the fundamental theorems and properties of discrete-time linear systems, filtering, sampling, and discrete-time Fourier Analysis. A valuable introduction to the fundamentals of continuous and discrete-time signal processing, this book is intended for the reader with little or no background in this subject. The emphasis is on development from basic principles. With this book the reader can become knowledgeable about both the theoretical and practical aspects of digital signal processing. Some special features of this book are: (1) gradual and step-by-step development of the mathematics for signal processing, (2) numerous examples and homework problems, (3) evolutionary development of Fourier series, Discrete Fourier Transform, Fourier Transform, Laplace Transform, and Z-Transform, (4) emphasis on the relationship between continuous and discrete time signal processing, (5) many examples of using the computer for applying the theory, (6) computer based assignments to gain practical insight, (7) a set of computer programs to aid the reader in applying the theory. The second edition of this well received text continues to provide coherent and comprehensive coverage of digital signal processing. It is designed for undergraduate students of Electronics and Communication engineering, Telecommunication engineering, Electronics and Instrumentation engineering, Electrical and Electronics engineering, Electronics and Computers engineering, Biomedical engineering and Medical Electronics engineering. This book will also be useful to AMIE and IETE students. Written with student-centred, pedagogically-driven approach, the text provides a self-contained introduction to the theory of digital signal processing. It covers topics ranging from basic discrete-time signals and systems, discrete convolution and correlation, Z-transform and its applications, realization of discrete-time systems, discrete-time Fourier transform, discrete Fourier series, discrete Fourier transform to fast Fourier transform. In addition to this, various design techniques for design of IIR and FIR filters are discussed. Multi-rate digital signal processing and introduction to digital signal processors and finite word length effects on digital filters are also covered. All the solved and unsolved problems in this book are designed to illustrate the topics in a clear way. MATLAB programs and the results for typical examples are also included at the end of chapters for the benefit of the students. New to This Edition A chapter on Finite Word Length Effects in Digital Filters Key Features • Numerous worked-out examples in each chapter • Short questions with answers help students to prepare for examinations and interviews • Fill in the blanks, review questions, objective type questions and unsolved problems at the end of each chapter to test the level of understanding of the subjectThis book clearly explains digital signal processing principles and shows how they can be used to build DSP systems. The aim is to give enough insight and practical guidance to enable an engineer to construct DSP systems. The book's programs are written in C, the language used in DSP. 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.Digital Signal Processing With the TMSC20C25 Hardware Use in This text Ralph Chassagnac and Darrell W. Hornung offer detailed, state-of-the-art guidance on using digital signal processing tools. Through the development of actual programming examples, the book demonstrates how DSP theory is put to practical use. Current problems in digital signal filtering, such as finite impulse response filters, infinite impulse response filters, and fast Fourier transform are addressed through the step-by-step implementation of assembly language code for a modern, real-time digital signal processor, the TMS320C25. Hardware considerations specific to the TMS320C25, such as memory organization, addressing modes and representation of fixed- and floating-point numbers are discussed in relation to software development. 1990 (0 471-51066-1) 464 pp. Digital Filter Design T. W. Parks and C. S. Burrus "The book is excellently written and fully illustrated ... it will soon become a reference book in the area of digital filter design." —Mathematics Abstracts With coverage from basic theory to working programs, this clear, practical text addresses frequency-domain analysis, design, and implementation of linear constant-coefficient digital filters on general purpose computers and special-purpose signal processors. Offering a complete, self-contained treatment of both FIR and IIR filters, a feature unique to this book, the text examines their underlying design theory, design formulas, and algorithms. Detailed coverage also includes a discussion of filter properties, the approximation problem, and implementation of the filter with fixed-point arithmetic. The book also includes detailed examples that illustrate the design and implementation of a typical filter as well as listings for nine FORTRAN programs for filter design. 1987 (0 471-82896-3) 342 pp. DFT/FFT And Convolution Algorithms Theory and Implementation C. S. Burrus and T. W. Parks Written for the scientist or engineer conversant with continuous-time signals and discrete-time signal analysis, this book details the Fourier transform of a discrete-time signal. Efficient algorithms for computing the Discrete Fourier Transform (DFT) are given special emphasis. Coverage includes continuous and discrete-time transform analysis of signals and properties of the DFT; methods of computing the DFT at a few frequencies (direct, Goertzel, and chirp transforms); and the three main approaches to an FFT (Cooley-Tukey, prime-factor, and Winograd transforms). The book also includes a discussion of DFT's use in the design of digital filters; a basis for custom DFT codes; and FORTRAN programs for special applications. 1985 (0 471-81932-8) 232 pp.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. For senior/graduate-level courses in Digital-Time Signal Processing, THE definitive, authoritative text on DSP — ideal for those with an introductory-level knowledge of signals and systems. Written by prominent DSP pioneers, it provides thorough treatment of the fundamental theorems and properties of discrete-time linear systems, filtering, sampling, and discrete-time Fourier Analysis. By focusing on the general and universal concepts in discrete-time signal processing, it remains vital and relevant to the new challenges arising in the field. A significant revision of a best-selling text for the introductory digital signal processing course. This book presents the fundamentals of discrete-time signals, systems, and modern digital processing and applications for students in electrical engineering, computer engineering, and computer science. The book is suitable for either a one-semester or a two-semester undergraduate level course in discrete systems and digital signal processing. It is also intended for use in a one-semester first-year graduate-level course in digital signal processing. This book uses MATLAB as a computing tool to explore traditional DSP topics and solve problems. This greatly expands the range and complexity of problems that students can effectively study in signal processing courses. A large number of worked examples, computer simulations and applications are provided, along with theoretical aspects that are essential in order to gain a good understanding of the main topics. Practicing engineers may also find it useful as an introductory text on the subject. Covers the analysis and representation of discrete-time signals and systems, including discrete-time convolution, difference equations, the z-transform, and the discrete-time Fourier transform. Emphasis is placed on the similarities and distinctions between discrete-time and continuous-time signals and systems. It also covers digital network structures for implementation fo both recursive
(infinite impulse response) and nonrecursive (finite impulse response) digital filters with four videocassettes devoted to digital filter design for recursive and nonrecursive filters. Concludes with a discussion of the fast Fourier transform algorithm for computation of the discrete Fourier transform. This comprehensive and up-to-date book focuses on an algebraic approach to the analysis and design of discrete-time signal processors, including material applicable to numeric and symbolic computation programs such as MATLAB. Written with clarity, it contains the latest detailed research results. This new, fully-revised edition covers all the major topics of digital signal processing (DSP) design and analysis in a single, all-inclusive volume, interweaving theory with real-world examples and design trade-offs. Building on the success of the original, this edition includes new material on random signal processing, a new chapter on spectral estimation, greatly expanded coverage of filter banks and wavelets, and new material on the solution of difference equations. Additional steps in mathematical derivations make them easier to follow, and an important new feature is the do-it-yourself section at the end of each chapter, where readers get hands-on experience of solving practical signal processing problems in a range of MATLAB experiments. With 120 worked examples, 20 case studies, and almost 400 homework exercises, the book is essential reading for anyone taking an introductory courses in DSP or an advanced course in signal processing. This book is also makes it an ideal reference for practitioners. This book is useful as a Textbook for undergraduate students of Electronics and Telecommunication Engineering and allied disciplines, as well as diploma and science courses. Undoubtedly one of the key factors influencing recent technology has been the advent of high speed computational tools. Virtually every advanced engineering system we come in contact with these days depends upon some form of sampling and digital signal processing. Well-known examples are digital tele phone systems, digital recording of audio signals, and computer control. These developments have been matched by the appearance of a plethora of books which explain a variety of analysis, synthesis and design tools applicable to sampled-data systems. The reader might therefore wonder what is distinct tive about the current book. Our observation of the existing literature is that the underlying continuous-time system is usually forgotten once the samples are taken. The alternative point of view, adopted in this book, is to formulate the analysis in such a way that the user is constantly reminded of the presence of the underlying continuous-time signals. We thus give emphasis to two aspects of sampled-data analysis: Firstly, we formulate the various algorithms so that the appropriate continuous-time system is approached as the sampling rate increases. Secondly we place emphasis on the continuous-time output response rather than simply focusing on the sampled response. 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. Taking a novel, less classical approach to the subject, the authors have written this book with the conviction that signal processing should be fun. Their treatment is less focused on the mathematics and more on the practical aspects, allowing the reader to become proficient and helpful for more advanced courses. The last chapter pulls together the individual topics into an in-depth look at the development of an end-to-end communication system. Richly illustrated with examples and exercises in each chapter, the book offers a fresh approach to the teaching of signal processing to upper-level undergraduates. Converting analog to digital signals and vice versa — Time-domain representation of discrete-time signals and systems — Frequency-domain representation of discrete-time signals — DSP application examples — Finite impulse response filter design — Infinite impulse response (IIR) filter design — Digital filter realizations — Digital signal processors — Hardware and software development tools. This excellent Senior undergraduate/graduate textbook offers an unprecedented measurement of science perspective on DSP theory and applications, a wealth of definitions and real-life examples making it invaluable for students, while practical. The field of digital signal processing (DSP) has spurred developments from basic theory of discrete-time signals and processing tools to diverse applications in telecommunications, speech and acoustics, radar, and video. This volume provides an accessible reference, offering theoretical and practical information to the audience of DSP users. This immense compilation outlines both introductory and specialized aspects of information-bearing signals in digital form, creating a resource relevant to the expanding needs of the engineering community. It also explores the use of computers and special-purpose digital hardware in extracting information or transforming signals in advantageous ways. Impacted areas presented include: Telecommunications Computer engineering Acoustics Seismic data analysis DSP software and hardware Image and video processing Remote sensing Multimedia applications Medical technology Radar and sonar applications. This authoritative collaboration, written by the foremost researchers and practitioners in their fields, comprehensively presents the range of DSP: from theory to application, in a practical and accessible manner, but which nonetheless include all essential foundation mathematics. As the series title implies, the scripts (of which there are more than 200) described in the text and supplied in code form (available at www.morganclaypool.com/page/isen) will run on both MATLAB and LabVIEW. Volume I consists of four chapters. The first chapter gives a brief overview of the field of digital signal processing. This is followed by a chapter detailing many useful signals and concepts, including convolution, recursion, difference equations, LTI systems, etc. The third chapter covers conversion from the continuous to discrete domain and back (i.e., analog-to-digital and digital-to-analog conversion), aliasing, the Nyquist rate, normalized frequency, conversion from one sample rate to another, waveform generation at various sample rates from stored wave data, and Mu-law compression. The fourth and final chapter of the present volume introduces the reader to many important principles of signal processing, including correlation, the correlation sequence, the Real DFT, correlation by convolution, matched filtering, simple FIR filters, and simple IIR filters. Chapter 4, in particular, provides an intuitive or “first principle” understanding of how digital filtering and frequency transforms work, preparing the reader for Volumes II and III, which provide, respectively, detailed
coverage of discrete frequency transforms (including the Discrete Time Fourier Transform, the Discrete Fourier Transform, and the z-Transform) and digital filter design (FIR design using Windowing, Frequency Sampling, and Optimum Equiripple techniques, and Classical IIR design). Volume IV, the culmination of the series, is an introductory treatment of LMS Adaptive Filtering and applications. The text for all volumes contains many examples, and many useful computational scripts, augmented by demonstration scripts and LabVIEW Virtual Instruments (VIs) that can be run to illustrate various signal processing concepts graphically on the user’s computer screen. Table of Contents: An Overview of DSP / Discrete Signals and Concepts / Sampling and Binary Representation / Transform and Filtering Principles” In three parts, this book contributes to the advancement of engineering education and that serves as a general reference on digital signal processing. Part I presents the basics of analog and digital signals and systems in the time and frequency domain. It covers the core topics: convolution, transforms, filters, and random signal analysis. It also treats important applications including signal detection in noise, radar range estimation for airborne targets, binary communication systems, channel estimation, banking and financial applications, and audio effects production. Part II considers selected signal processing systems and techniques. Core topics covered are the Hilbert transformer, binary signal transmission, phase-locked loops, sigma-delta modulation, noise shaping, quantization, adaptive filters, and non-stationary signal analysis. Part III presents some selected advanced DSP topics. Commercial applications of speech processing and recognition are fast becoming a growth industry that will shape the next decade. Now students and practicing engineers of signal processing can find in a single volume the fundamentals essential to understanding this rapidly developing field. IEEE Press is pleased to publish a classic reissue of Discrete-Time Processing of Speech Signals. Specially featured in this reissue is the addition of valuable World Wide Web links to the latest speech data references. This landmark book offers a balanced discussion of both the mathematical theory of digital speech signal processing and critical contemporary applications. The authors provide a comprehensive view of all major modern speech processing areas: speech production physiology and modeling, signal analysis techniques, coding, enhancement, quality assessment, and recognition. You will learn the principles needed to understand advanced technologies in speech processing – from speech coding for communications systems to biomedical applications of speech analysis and recognition. Ideal for self-study or as a course text, this far-reaching reference book offers an extensive historical context for concepts under discussion, end-of-chapter problems, and practical algorithms. Discrete-Time Processing of Speech Signals is the definitive resource for students, engineers, and scientists in the speech processing field. An Instructor’s Manual presenting detailed solutions to all the problems in the book is available upon request from the Wiley Marketing Department. This book covers the basics of processing and spectral analysis of monovariate discrete-time signals. The approach is practical, the aim being to acquaint the reader with the indications for and drawbacks of the various methods and to highlight possible misuses. The book is rich in original ideas, visualized in new and illuminating ways, and is structured so that parts can be skipped without loss of continuity. Many examples are included, based on synthetic data and real measurements from the fields of physics, biology, medicine, macroeconomics etc., and a complete set of MATLAB exercises requiring no previous experience of programming is provided. Prior advanced mathematical skills are not needed in order to understand the contents: a good command of basic mathematical analysis is sufficient. Where more advanced mathematical tools are necessary, they are included in an Appendix and presented in an easy-to-follow way. With this book, digital signal processing leaves the domain of engineering to address the needs of students and scholars in traditionally less quantitative disciplines, now facing increasing amounts of data. The book provides an introduction to digital signal processing for intermediate level students of electronic and/or electrical engineering and is also relevant to other disciplines which deal with time-series analysis: these include acoustics, mathematics, statistics, psychology and economics. ’Digital Signal Processing’ introduces the tools used in the analysis and design of discrete-time systems for signal processing. This textbook offers a fresh approach to digital signal processing (DSP) that combines heuristic reasoning and physical appreciation with sound mathematical methods to illuminate DSP concepts and practices. It uses metaphors, analogies and creative explanations, along with examples and exercises to provide deep and intuitive insights into DSP concepts. Practical DSP requires hybrid systems including both discrete- and continuous-time components. This book follows a holistic approach and presents discrete-time processing as a seamless continuation of continuous-time signals and systems, beginning with a review of continuous-time signals and systems, frequency response, and filtering. The synergistic combination of continuous-time and discrete-time perspectives leads to a deeper appreciation and understanding of DSP concepts and practices. • For upper-level undergraduates • Illustrates concepts with 500 high-quality figures, more than 170 fully worked examples, and hundreds of end-of-chapter problems, more than 150 drill exercises, including complete and detailed solutions • Seamlessly integrates MATLAB throughout the text to enhance learning

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Page 4/4