Digital Signal Processing A Computer Based Approach 4th Edition Solution Manual

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 Signal Processing covers the basic theory and practice of digital signal processing (DSP) at an introductory level. As with all volumes in the Essential Electronics Series, this book retains the unique formula of minimal mathematics and straightforward explanations. The author has included examples throughout of the standard software design package, MATLAB and screen dumps are used widely throughout to illustrate the text. Ideal for students on degree and diploma level courses in electric and electronic engineering, 'Introduction to Digital Signal Processing' contains numerous worked examples throughout as well as further problems with solutions to enable students to work both independently and in conjunction with their course. Assumes only minimum knowledge of mathematics and electronics Concise and written in a straightforward and accessible style Packed with worked examples, exercises and self-assesment questions

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.

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.

Mneney's text focuses on basic concepts of digital signal processing, MATLAB simulation, and implementation on selected DSP hardware.

A uniquely practical DSP text, this book gives a thorough understanding of the principles and applications of DSP with a minimum of mathematics, and provides the reader with an introduction to DSP applications in telecoms, control engineering and measurement and data analysis systems. The new edition contains: • Expanded coverage of the basic concepts to aid understanding • New sections on filter sysnthesis, control theory and contemporary topics of speech and image recognition • Full solutions to all questions and exercises in the book Assuming the reader already has some prior knowledge of signal theory, this textbook will be highly suitable for undergraduate and postgraduate students in electrical and electronic engineering taking introductory and advanced courses in DSP, as well as courses in communications and control systems engineering. It will also prove an invaluable introduction to DSP and its applications for the professional engineer. Expanded coverage of the basic concepts to aid understanding, along with a wide range of DSP applications New textbook features included throughout, including learning objectives, summary sections, exercises and worked examples to increase accessibility of the text Full solutions to all questions and exercises included in the book

Explains digital and analog signals and DSP applications using everyday examples and simple diagrams, including digital signal collection, filtering, analysis, and how digital signal processing works in modern electronic devices.

This well-written introduction exposes students to digital signal processing in a computer environment. Through a series of projects and exercises, the text helps students develop confidence in manipulating discrete-time signals without having to write and debug large programs. Included are a summary of many concepts basic to signal processing and a library of DSP computer functions that run on personal computers. Included with the text is a disk with stand-alone programs that perform elementary signal processing functions. The software is easy to use with an on-line help function that explains the usage of all DSP functions. This allows students to concentrate on DSP principles instead of details related to software usage. The topics covered include practical applications that help students relate DSP concepts to real-world problems. Exercises can be modified easily in many cases to provide variety from year to year. This encourages independent work and discourages copying the lab work from students in previous classes. Exercises are designed for easy grading; for example, many exercises call for a hand-drawn sketch that can often be graded by inspection. All exercises have been student tested at Georgia Institute of Technology. Chapter 7 includes 9 projects that provide an opportunity for students to apply a variety of DSP concepts and exercise their creativity. A title in the Georgia Institute of Technology Digital Signal Processing Laboratory Series.

Digital Signal Processing: Principles, Algorithms and System Design is used in a wide range of applications, including voice processing, image processing, digital communications, the transfer of data over the Internet, and image and data compression. Engineers who develop DSP applications today, and in the future, need to understand the fundamental theories and mathematical algorithms, and will need to address implementation issues, like mapping algorithms to hardware, computational efficiency, and the effects of finite precision arithmetic. Alexander and Williams cover all these topics at a level appropriate for senior undergraduates or first year graduate students, making this text the ideal bridge between the theory and analytical procedures that form the basis for modern DSP and practical implementation. 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

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

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.

Computers are at the center of almost everything related to audio. Whether for synthesis in music production, recording in the studio, or mixing in live sound, the computer plays an essential part. Audio effects plug-ins and virtual instruments are implemented as software computer code. Music apps are computer programs run on a mobile device. All these tools are created by programming a computer. Hack Audio: An Introduction to Computer Programming and Digital Signal Processing in MATLAB provides an introduction for musicians and audio engineers interested in computer programming. It is intended for a range of readers including those with years of programming experience and those ready to write their first line of code. In the book, computer programming is used to create audio effects using digital signal processing. By the end of the book, readers implement the following effects: signal gain change, digital summing, tremolo, auto-pan, mid/side processing, stereo widening, distortion, echo, filtering, equalization, multi-band processing, vibrato, chorus, flanger, phaser, pitch shifter, auto-wah, convolution and algorithmic reverb, vocoder, transient designer, compressor, expander, and de-esser. Throughout the book, several types of test signals are synthesized, including: sine wave, square wave, sawtooth wave, triangle wave, impulse train, white noise, and pink noise. Common visualizations for signals and audio effects are created including: waveform, characteristic curve, goniometer, impulse response, step response, frequency spectrum, and spectrogram. In total, over 200 examples are provided with completed code demonstrations.

'Digital Signal Processing' introduces the tools used in the analysis and design of discrete-time systems for signal processing.

Digital signal processing continues to be of importance in fields as diverse as communications, radar, seismology, sonar, biomedicine, data analysis and aerospace. Some of the main problems of digital signal processing are identified in this book and interpreting the results of signal processing are detailed. Principles, applications and design methods are illustrated using a range of computer programs which have been translated into C for this edition. The accompanying disk provides a graphic display of the results of processing example signals and allows the reader to experiment with the programs.

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

The book provides a comprehensive exposition of all major topics in digital signal processing (DSP). With numerous illustrative examples for easy understanding of the topics, it also includes MATLAB-based examples with codes in order to encourage the readers to become more confident of the fundamentals and to gain insights into DSP. Further, it presents real-world signal processing design problems using MATLAB and programmable DSP processors. In addition to problems that require analytical solutions, it discusses problems that require solutions using MATLAB at the end of each chapter. Divided into 13 chapters, it addresses many emerging topics, which are not typically found in advanced texts on DSP. It includes a chapter on adaptive digital filters used in the signal processing problems for faster acceptable results in the presence of changing environments and changing system requirements. Moreover, it offers an overview of wavelets, enabling readers to easily understand the basics and applications of this powerful mathematical tool for signal and image processing. The final chapter explores DSP processors, which is an area of growing interest for researchers. A valuable resource for undergraduate and graduate students, it can also be used for self-study by researchers, practicing engineers and scientists in electronics, communications, and computer engineering as well as for teaching one- to two-semester courses.

"An excellent introductory book" (Review of the First Edition in the International Journal of Electrical Engineering Education) " it will serve as a reference book in this area for a long time" (Review of Revised Edition in Zentralblatt für Mathematik (Germany)) Firmly established as the essential introductory Digital Signal Processing (DSP) text, this second edition reflects the growing importance of random digital signals and random DSP in the undergraduate syllabus by including two new chapters. The authors' practical, problem-solving approach to DSP continues in this new material, which is backed up by additional worked examples and computer programs. The book now features: * fundamentals of digital signals and systems * time and frequency domain analysis and processing, including digital convolution and the Discrete and Fast Fourier Transforms * design and practical application of digital filters * description and processing of random signals, including correlation, filtering, and the detection of signals in noise Programs in C and equivalent PASCAL are listed in an Appendix. Typical results and graphic plots from all the programs are illustrated and discussed in the main text. The overall approach assumes no prior knowledge of electronics, computing, or DSP. An ideal text for

undergraduate students in electrical, electronic and other branches of engineering, computer science, applied mathematics and physics. Practising engineers and scientists will also find this a highly accessible introduction to an increasingly important field.
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
This book provides a comprehensive overview of digital signal processing for a multi-disciplinary audience. It posits that though the theory involved in digital signal processing stems from electrical, electronics, communication, and control engineering, the topic has use in other disciplinary areas like chemical, mechanical, civil, computer science, and management. This book is written about digital signal processing in such a way that it is suitable for a wide ranging audience. Readers should be able to get a grasp of the field, understand the concepts easily, and apply as needed in their own fields. It covers sampling and reconstruction of signals; infinite impulse response filter; finite impulse response filter; multi rate signal processing; statistical signal processing; and applications in multidisciplinary domains. The book takes a functional approach and all techniques are illustrated using Matlab.
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.
Digital signal processing has progressed rapidly from a specialist research topic to one with practical applications in many disciplines, including branches of engineering and science which involve data acquisition, such as meterology, physics and information systems. This book aims to provide students with an introductory, one-term course in the subject, using a considerable number of computer programmes to illustrate the text. A number of worked examples have been included in order to illustrate and develop important ideas and design techniques. Problems designed to test and consolidate work already undertaken are supplied at the end of each chapter, and selected answers are given at the end of the book.
About the Book : - Digital Signal Processing Fundamentals Digital Signal Processing (DSP), as the term suggests, is the processing of signals using digital computers. These signals might be anything transferred from an analog domain to a digital form (e.g., temperature and pressure sensors, voices over a telephone, images from a camera, or data transmittal though computes). As a result, understanding the whole spectrum of DSP technology can be a daunting task for electrical engineering professionals and students alike. Digital Signal Processing Fundamentals provides a comprehensive look at DSP by introducing the important mathematical processes and then providing several application-specific tutorials for practicing the techniques learned. Beginning with general theory, including Fourier Analysis, the mathematics of complex numbers, Fourier transforms, differential equations, analog and digital filters, and much more; the book then delves into Matlab and Scilab tutorials with examples on solving practical engineering problems, followed by software applications on image processing and audio processing - complete with all the algorithms and source code. This is an invaluable resource for anyone seeking to understand how DSP works. Features: Provides a comprehensive overview and introduction of digital signal processing technology. Provides application with software algorithms Explains the concept of Nyquist frequency, orthogonal functions and method of finding Fourier coefficients Includes a CD-ROM with the source code for the projects plus Matlab and Scilab that generate graphs, figures in the book, and third party application software Discusses the techniques of digital filtering and windowing of input data, including: Butterwoth, Chebyshev, and elliptic filter formulation. Table Of Contents : Fourier Analysis Complex Number Arithmetic The Fourier Transform Solutions of Differential Equations Laplace Transforms and z-Tranforms Filter Design Digital Filters The FIR Filters Appendix A : Matlab Tutorial Appendix B : Scilab Tutorial Appendix C : Digital Filter Applications Appendix D : About the CD-ROM Appendix E : Software Licenses Appendix F : Bibliography Index About Author :- Ashfaq A. Khan (Baton Rouge, LA) is a senior software engineer for LIGO Livingston Observatory, with over 20 years of experience in system design. He has conducted several workshop and is the author of Practical Linux Programming: Device Drivers, Embedded Systems, and the Internet.
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
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

If you understand basic mathematics and know how to program with Python, you're ready to dive into signal processing. While most resources start with theory to teach this complex subject, this practical book introduces techniques by showing you how they're applied in the real world. In the first chapter alone, you'll be able to decompose a sound into its harmonics, modify the harmonics, and generate new sounds. Author Allen Downey explains techniques such as spectral decomposition, filtering, convolution, and the Fast Fourier Transform. This book also provides exercises and code examples to help you understand the material. You'll explore: Periodic signals and their spectrums Harmonic structure of simple waveforms Chirps and other sounds whose spectrum changes over time Noise signals and natural sources of noise The autocorrelation function for estimating pitch The discrete cosine transform (DCT) for compression The Fast Fourier Transform for spectral analysis Relating operations in time to filters in the frequency domain Linear time-invariant (LTI) system theory Amplitude modulation (AM) used in radio Other books in this series include Think Stats and Think Bayes, also by Allen Downey.

Now available in a three-volume set, this updated and expanded edition of the bestselling The Digital Signal Processing Handbook continues to provide the engineering community with authoritative coverage of the fundamental and specialized aspects of information-bearing signals in digital form. Encompassing essential background material, technical details, standards, and software, the second edition reflects cutting-edge information on signal processing algorithms and protocols related to speech, audio, multimedia, and video processing technology associated with standards ranging from WiMax to MP3 audio, low-power/high-performance DSPs, color image processing, and chips on video. Drawing on the experience of leading engineers, researchers, and scholars, the three-volume set contains 29 new chapters that address multimedia and Internet technologies, tomography, radar systems, architecture, standards, and future applications in speech, acoustics, video, radar, and telecommunications. Emphasizing theoretical concepts, Digital Signal Processing Fundamentals provides comprehensive coverage of the basic foundations of DSP and includes the following parts: Signals and Systems; Signal Representation and Quantization; Fourier Transforms; Digital Filtering; Statistical Signal Processing; Adaptive Filtering; Inverse Problems and Signal Reconstruction; and Time–Frequency and Multirate Signal Processing.

&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 book forms the first part of a complete MSc course in an area that is fundamental to the continuing revolution in information technology and communication systems. Massively exhaustive, authoritative, comprehensive and reinforced with software, this is an introduction to modern methods in the developing field of Digital Signal Processing (DSP). The focus is on the design of algorithms and the processing of digital signals in areas of communications and control, providing the reader with a comprehensive introduction to the underlying principles and mathematical models. Provides an introduction to modern methods in the developing field of Digital Signal Processing (DSP) Focuses on the design of algorithms and the processing of digital signals in areas of communications and control Provides a comprehensive introduction to the underlying principles and mathematical models of Digital Signal Processing

Covers advances in the field of computer techniques and algorithms in digital signal processing.

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.

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

Informal, easy-to-understand introduction covers phasors and tuning forks, wave equation, sampling and quantizing, feedforward and feedback filters, comb and string filters, periodic sounds, transform methods, and filter design. 1996 edition.

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

"This book offers an introduction to digital signal processing (DSP) with an emphasis on audio signals and computer music ... This book is designed for both technically and musically inclined readers alike--folks with a common goal of exploring digital signal processing"--Cover, p. [4].

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.

Karlheinz Brandenburg and Mark Kahrs With the advent of multimedia, digital signal processing (DSP) of sound has emerged from the shadow of bandwidth limited speech processing. Today, the main appli cations of audio DSP are high quality audio coding and the digital generation and manipulation of music signals. They share common research topics including percep tual measurement techniques and analysis/synthesis methods. Smaller but nonetheless very important topics are hearing aids using signal processing technology and hardware architectures for digital signal processing of audio. In all these areas the last decade has seen a significant amount of application oriented research. The topics covered here coincide with the topics covered in the biannual work shop on “Applications of Signal Processing to Audio and Acoustics”. This event is sponsored by the IEEE Signal Processing Society (Technical Committee on Audio and Electroacoustics) and takes place at Mohonk Mountain House in New Paltz, New York. A short overview of each chapter will illustrate the wide variety of technical material presented in the chapters of this book. John Beerends: Perceptual Measurement Techniques. The advent of perceptual measurement techniques is a byproduct of the advent of digital coding for both speech and high quality audio signals. Traditional measurement schemes are bad estimates for the subjective quality after digital coding/decoding. Listening tests are subject to sta tistical uncertainties and the basic question of repeatability in a different environment.

An Introduction to the Digital Analysis of Stationary Signals: A Computer Illustrated Text directly illustrates the various techniques required to make accurate measurements of the properties of fluctuating signals. Emphasis is on qualitative ideas rather than detailed mathematical analysis for which the computer illustrated text format is ideally suited. The author reinforces normal figures and diagrams with computer-generated graphical displays produced dynamically by the student. This package of text and accompanying software is not specific to any particular microcomputer.