Digital Signal Compression Principles And Practice

This book describes algorithmic methods and hardware implementations that aim to help realize the promise of Compressed Sensing (CS), namely the ability to reconstruct high-dimensional signals from a properly chosen low-dimensional “portrait”. The authors describe a design flow and some low-resource physical realizations of sensing systems based on CS. They highlight the pros and cons of several design choices from a pragmatic point of view, and show how a lightweight and mild but effective form of adaptation to the target signals can be the key to consistent resource saving. The basic principle of the devised design flow can be applied to almost any CS-based sensing system, including analog-to-information converters, and has been proven to fit an extremely diverse set of applications. Many practical aspects required to put a CS-based sensing system to work are also addressed, including saturation, quantization, and leakage phenomena.

This book presents various contributions of splines to signal and image processing from a unified perspective that is based on the Zak transform (ZT). It expands the methodology from periodic splines, which were presented in the first volume, to non-periodic splines. Together, these books provide a universal toolbox accompanied by MATLAB software for manipulating polynomial and discrete splines, spline-based wavelets, wavelet packets and wavelet frames for signal/ image processing applications. In this volume, we see that the ZT provides an integral representation of discrete and polynomial splines, which, to some extent, is similar to Fourier integral. The authors explore elements of spline theory and design, and consider different types of polynomial and discrete splines. They describe applications of spline-based wavelets to data compression. These splines are useful for real-time signal processing and, in particular, real-time wavelet and frame transforms. Further topics addressed in this volume include: "global" splines, such as interpolating, self-dual and smoothing, whose supports are infinite; the compactly supported quasi-interpolating and smoothing splines including quasi-interpolating splines on non-uniform grids; and cubic Hermite splines as a source for the design of multiwavelets and multiwavelet frames. Readers from various disciplines including engineering, computer science and mathematical information technology will find the descriptions of algorithms, applications and software in this book especially useful.

Combining clear explanations of elementary principles, advanced topics and applications with step-by-step mathematical derivations, this textbook provides a comprehensive yet accessible introduction to digital signal processing. All the key topics are covered, including discrete-time Fourier transform, z-transform, discrete Fourier transform and FFT, A/D conversion, and FIR and IIR filtering algorithms, as well as more advanced topics such as multirate systems, the discrete cosine transform and spectral signal processing. Over 600 full-color illustrations, 200 fully worked examples, hundreds of end-of-chapter homework problems and detailed computational examples of DSP algorithms implemented in MATLAB® and C aid understanding, and help put knowledge into practice. A wealth of supplementary material accompanies the book online, including interactive programs for instructors, a full set of solutions and MATLAB® laboratory exercises, making this the ideal text for senior undergraduate and graduate courses on digital signal processing.

Summary: "Bridging the gap between theory and application, this text covers all the main areas of modern DSP. Principles, applications, and hardware implementation issues are presented, and a wealth of worked examples and end of chapter exercises provide the opportunity for self-learning. Throughout the book emphasis is placed on applications to signal, image, and video processing, and real time implementation of DSP algorithms using DSP processors is highlighted."--Back Cover.

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.

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

Digital Signal Processing: Fundamentals and Applications, Third Edition, not only introduces students to the fundamental principles of DSP, it also provides a working knowledge that they take with them into their engineering careers. Many instructive, worked examples are used to illustrate the material, and the use of mathematics is minimized for an easier grasp of concepts. As such, this title is also useful as a reference for non-engineering students and practicing engineers. The book goes beyond DSP theory, showing the 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, ?-law, ADPCM, and multi-rate DSP, over-sampling ADC subband coding, and wavelet transform. Covers DSP principles with an emphasis on communications and control applications Includes chapter objectives, worked examples, and end-of-chapter exercises that aid the reader in grasping key concepts and solving related problems Provides an accompanying website with MATLAB programs for simulation and C programs for real-time DSP Presents new problems of varying types and difficulties

Covers all recognised coding algorithms

Describes various wavelet image coding systems that use set partitioning primarily, such as SBHP (Subband Block Hierarchical Partitioning), SPIHT, and EZBC (Embedded Zero-Block Coder).

A self-contained approach to DSP techniques and applications in radar imaging The processing of radar images, in general, consists of three major fields: Digital Signal

Processing (DSP); antenna and radar operation; and algorithms used to process the radar images. This book brings together material from these different areas to allow readers to gain a thorough understanding of how radar images are processed. The book is divided into three main parts and covers: * DSP principles and signal characteristics in both analog and digital domains, advanced signal sampling, and interpolation techniques * Antenna theory (Maxwell equation, radiation field from dipole, and linear phased array), radar fundamentals, radar modulation, and target-detection techniques (continuous wave, pulsed Linear Frequency Modulation, and stepped Frequency Modulation) * Properties of radar images, algorithms used for radar image processing, simulation examples, and results of satellite image files processed by Range-Doppler and Stolt interpolation algorithms The book fully utilizes the computing and graphical capability of MATLAB? to display the signals at various processing stages in 3D and/or cross-sectional views. Additionally, the text is complemented with flowcharts and system block diagrams to aid in readers' comprehension. Digital Signal Processing Techniques and Applications in Radar Image Processing serves as an ideal textbook for graduate students and practicing engineers who wish to gain firsthand experience in applying DSP principles and technologies to radar imaging.

Digital Signal CompressionPrinciples and PracticeCambridge University Press

This book presents tools and algorithms required tocompress/uncompress signals such as speech and music. Thesealgorithms are largely used in mobile phones, DVD players, HDTVsets, etc. In a first rather theoretical part, this book presents thestandard tools used in compression systems: scalar and vectorquantization, predictive quantization, transform quantization,entropy coding. In particular we show the consistency between thesedifferent tools. The second part explains how these tools are usedin the latest speech and audio coders. The third part gives Matlabprograms simulating these coders.

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.

Digital image business applications are expanding rapidly, driven by recent advances in the technology and breakthroughs in the price and performance of hardware and firmware. This ever increasing need for the storage and transmission of images has in turn driven the technology of image compression: image data rate reduction to save storage space and reduce transmission rate requirements. Digital image compression offers a solution to a variety of imaging applications that require a vast amount of data to represent the images, such as document imaging management systems, facsimile transmission, image archiving, remote sensing, medical imaging, entertainment, HDTV, broadcasting, education and video teleconferencing. Digital Image Compression: Algorithms and Standards introduces the reader to compression algorithms, including the CCITT facsimile standards T.4 and T.6, JBIG, CCITT H.261 and MPEG standards. The book provides comprehensive explanations of the principles and concepts of the algorithms, helping the readers' understanding and allowing them to use the standards in business, product development and R&D. Audience: A valuable reference for the graduate student, researcher and engineer. May also be used as a text for a course on the subject.

Presents trends and techniques for successful intelligent decision-making andtransfer of products through digital signal processing.

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

Sixth in the book series, Advances in Image Communication, which documents the rapid advancements of recent years in image communication technologies, this volume provides a comprehensive exploration of subband coding. Originally, subband coding and transform coding were developed separately. The former, however, benefitted considerably from the earlier evolution of transform coding theory and practice. Retaining their own terminology and views, the two methods are closely related and this book indeed aims to unify the approaches. Specifically, the volume contributes effectively to the understanding of frequency domain coding techniques. Many images from coding experiments are presented, enabling the reader to consider the properties of different coders. Chapter 1 introduces the problem of image compression in general terms. Sampling of images and other fundamental concepts, such as entropy and the rate distortion function, are briefly reviewed. The idea of viewing coding techniques as series expansions is also introduced. The second chapter presents signal decomposition and the conditions for perfect reconstruction from minimum representations. Chapter 3 deals with filter bank structures, primarily those displaying the perfect reconstruction property. Quantization techniques and the efficient exploitation of the bit resources are discussed from a theoretical perspective in Chapter 4 and this issue is further examined in Chapter 6, from a more practical point of view. Chapter 5 provides a development of gain formulas, i.e. quantitative measures of the performance of filter banks in a subband coding context, and these are then employed in a search for optimal filter banks. A number of examples of coded images using different subband coders are presented in Chapter 7, these indicating that subband coders give rise to some characteristic types of image degradations. Accordingly, Chapter 8 presents several techniques for minimizing these artifacts. The theory and practice of subband coding of video, at several target bit rates, is discussed in the last chapter.

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

This comprehensive and engaging textbook introduces the basic principles and techniques of signal processing, from the fundamental ideas of signals and systems theory to real-world applications. Students are introduced to the powerful foundations of modern signal processing, including the basic geometry of Hilbert space, the mathematics of Fourier transforms, and essentials of sampling, interpolation, approximation and compression The authors discuss real-world issues and hurdles to using these tools, and ways of adapting them to overcome problems of finiteness and localization, the limitations of uncertainty, and computational costs. It includes over 160 homework problems and over 220 worked examples, specifically designed to test and expand students' understanding of the fundamentals of signal processing, and is accompanied by extensive online materials designed to aid learning, including Mathematica® resources and interactive demonstrations.

"Digital Compression for Multimedia" captures in a single reference the current standards for speech, audio, video, image, fax and file compression. It is intended for engineers and computer scientists designing and implementing compression techniques, system integrators, technical managers, and researchers. The essential ideas and motivation behind the various compression methods are presented and insight is provided into the evolution of the standards.

This book provides comprehensive, graduate-level treatment of analog and digital signal analysis suitable for course use and self-guided learning. This expert text guides the reader from the basics of signal theory through a range of application tools for use in acoustic analysis, geophysics, and data compression. Each concept is introduced and explained step by step, and the necessary mathematical formulae are integrated in an accessible and intuitive way. The first part of the book explores how analog systems and signals form the basics of signal analysis. This section covers Fourier series and integral transforms of analog signals, Laplace and Hilbert transforms, the main analog filter classes, and signal modulations. Part II covers digital signals, demonstrating their key advantages. It presents z and Fourier transforms, digital filtering, inverse filters, deconvolution, and parametric modeling for deterministic signals. Wavelet decomposition and reconstruction of non-stationary signals are also discussed. The third part of the book is devoted to random signals, including spectral estimation, parametric modeling, and Tikhonov regularization. It covers statistics of one and two random variables and the principles and methods of spectral analysis. Estimation of signal properties is discussed in the context of ergodicity conditions and parameter estimations, including the use of Wiener and Kalman filters. Two appendices cover the basics of integration in the complex plane and linear algebra. A third appendix presents a basic Matlab toolkit for computer signal analysis. This expert text provides both a solid theoretical understanding and tools for real-world applications.

Devices overview. Discrete signal and systems. Z transforms. The discrete Fourier transform. FIR and IIR filter design methods. Kalman filters. Implementation of digital control algorithms. Review of architectures. Microcontrollers. Systolic arrays. Case studies.

In a field as rapidly expanding as digital signal processing, even the topics relevant to the basics change over time both in their nature and their relative importance. It is important, therefore, to have an up-to-date text that not only covers the fundamentals, but that also follows a logical development that leaves no gaps readers must somehow bridge by themselves. Digital Signal Processing with Examples in MATLAB® is just such a text. The presentation does not focus on DSP in isolation, but relates it to continuous signal processing and treats digital signals as samples of physical phenomena. The author also takes care to introduce important topics not usually addressed in signal processing texts, including the discrete cosine and wavelet transforms, multirate signal processing, signal coding and compression, least squares systems design, and adaptive signal processing. He also uses the industry-standard software MATLAB to provide examples of signal processing, system design, spectral analysis, filtering, coding and compression, and exercise solutions. All of the examples and functions used in the text are available online at www.crcpress.com. Designed for a one-semester upper-level course but also ideal for self-study and reference, Digital Signal Processing with Examples in MATLAB is complete, self-contained, and rigorous. For basic DSP, it is quite simply the only book you need.

Based on fundamental principles from mathematics, linear systems, and signal analysis, digital signal processing (DSP) algorithms are useful for extracting information from signals collected all around us. Combined with today's powerful computing capabilities, they can be used in a wide range of application areas, including engineering, communicati

Here is a fully readable introduction to the basic technologies, infrastructures, costs, and applications for digital audio and video compression. Delivering a concise account of compression's terms, techniques, and tricks in an easy-to-read style, it covers the basic principles underlying digital signal processing and compression; how human beings see and hear; how audio and video are reproduced; all of the existing and emerging compression standards; video and audio compression techniques; and compression and reproduction requirements of different applications, including videoconferencing.

The purpose of Transporting Compressed Digital Video is to introduce fundamental principles and important technologies used in design and analysis of video transport systems for many video applications in digital networks. In the past two decades, progress in digital video processing, transmission, and storage technologies, such as video compression, digital modulation, and digital storage disk, has proceeded at an astounding pace. Digital video compression is a field in which fundamental technologies were motivated and driven by practical applications so that they often lead to many useful advances. Especially, the digital video-compression standards, developed by the Moving Pictures Expert Group (MPEG) of the International Organization for Standardization (ISO) and the International Electrotechnical Commission (IEC), have enabled many successful digital-video applications. These applications range from digital-video disk (DVD) and multimedia CDs on a desktop computer, interactive digital cable television, to digital satellite networks. MPEG has become the most recognized standard for digital video compression. MPEG video is now an integral part of most digital video transmission and storage systems. Nowadays, video compression technologies are being used in almost all modern digital video systems and networks. Not only is video compression equipment being implemented to increase the bandwidth efficiency of communication systems, but video compression also provides innovative solutions to many related vid- networking problems. The subject of Transporting Compressed Digital Video includes several important topics, in particular video buffering, packet scheduling, multiplxing and synchronization.

The topic of the proposed book is signal compression. The compression (or low bit rate coding) of speech, audio, image and video signals is a key technology for rapidly emerging opportunities in multimedia products and services.The book contains chapters dedicated to the subtopics of data, speech, audio and visual signal coding, together with an introductory overview chapter on signal compression. The overview article summarizes current capabilities and future trends. The signal-specific chapters that follow focus on the latest technologies and coding standards, while including self-contained introductions to the respective signal domains. The authors of the book chapters are recognized experts in the field of signal processing, compression in particular.Signal compression dealing with both audio and visual signals technology has progressed very rapidly. The proposed book fills a clear void, and should prove to be a valuable reference, both to the practicing professional and to the relatively uninitiated student.

This book explains the stages necessary to create a wavelet compression system for images and describes state-of-the-art systems used in image compression standards and current research. It starts with a high level discussion of the properties of the wavelet transform, especially the decomposition into multi-resolution subbands. It continues with an exposition of the null-zone, uniform quantization used in most subband coding systems and the optimal allocation of bitrate to the different subbands. Then the image compression systems of the FBI Fingerprint Compression Standard and the JPEG2000 Standard are described in detail. Following that, the set partitioning coders SPECK and SPIHT, and EZW are explained in detail and compared via a fictitious wavelet transform in actions and number of bits coded in a single pass in the top bit plane. The presentation teaches that, besides producing efficient compression, these coding systems, except for the FBI Standard, are capable of writing bit streams that have attributes of rate scalability, resolution scalability, and random access decoding. Many diagrams and tables accompany the text to aid understanding. The book is generous in pointing out references and resources to help the reader who wishes to expand his knowledge, know the origins of the methods, or find resources for running the various algorithms or building his own coding system.

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

Learn how diagnostic ultrasound works, and find out how to properly handle artifacts, scan safely, evaluate instrument performance, and prepare for registry examinations, with the market-leading Sonography Principles and Instruments, 9th Edition. It concisely and comprehensively covers the essential aspects of ultrasound physics and instrumentation like Doppler, artifacts, safety, quality assurance, and the newest technology — all in a dynamic, highly visual format for easy review of key information. Dr. Kremkau, unlike others, uses extensive exam questions, over 1,000 high-quality illustrations, and only the most basic equations to simplify complicated concepts, making this text a highly respected reference for sonography students and professionals. Essential coverage of physics and sonography prepares you for the physics portion of the American Registry for Diagnostic Medical Sonography (ARDMS) certification exam. Current technology content, including the continuing progression of contrast agents and 3D and the more general aspects of transducers and instruments, helps you better comprehend the text. Straightforward explanations simplify complicated concepts. Learning objectives at the beginning of every chapter give you a measurable outcome to achieve. Key terms provide you with a list of the most important terms at the beginning of each chapter. Key Points, called out with an icon and special type, highlight the most important information to help you study more efficiently. Bulleted reviews at the end of each chapter identify key concepts covered in that chapter. End-of-chapter exercises test your knowledge and understanding with a mix of true/false, fill-in-the-blank, multiple choice, and matching questions. Glossary of key terms at the end of the book serves as a quick reference, letting you look up definitions without having to search through each chapter. Appendices, including a List of Symbols, Complication of Equations, and Mathematics Review, equip you with additional resources to help comprehend difficult concepts. An Evolve site with student resources enhances your learning experience. A full-color design depicts over 120 high-quality ultrasound scans similar to what you will encounter in the clinical setting. NEW! All-new content on elastography, shear wave imaging, acoustic radiation force impulse imaging (ARFI), volume imaging, power M-mode Doppler in TCD, miniaturization, and newer acquisition technique in Epic System keeps you in the know. NEW! Updated instrument output data and official safety statements ensure you are current with today’s technology. NEW! Updated art added to necessary chapters gives you an up-to-date representation of what you will encounter in the clinical setting.

Herb Caen, a popular columnist for the San Francisco Chronicle, recently quoted a Voice of America press release as saying that it was reorganizing in order to "eliminate duplication and redundancy. " This quote both states a goal of data compression and illustrates its common need: the removal of duplication (or redundancy) can provide a more efficient representation of data and the quoted phrase is itself a candidate for such surgery. Not only can the number of words in the quote be reduced without losing informa tion, but the statement would actually be enhanced by such compression since it will no longer exemplify the wrong that the policy is supposed to correct. Here compression can streamline the phrase and minimize the em barassment while improving the English style. Compression in general is intended to provide efficient representations of data while preserving the essential information contained in the data. This book is devoted to the theory and practice of signal compression, i. e. , data compression applied to signals such as speech, audio, images, and video signals (excluding other data types such as financial data or general purpose computer data). The emphasis is on the conversion of analog waveforms into efficient digital representations and on the compression of digital information into the fewest possible bits. Both operations should yield the highest possible reconstruction fidelity subject to constraints on the bit rate and implementation complexity.
Provides clear and easily understandable coverage of the fundamental concepts and coding methods, whilst retaining technical depth and rigor.