Introduction To Digital Audio Coding And Standards The Springer International Series In Engineering And Computer Science

When Speech and Audio Signal Processing published in 1999,it stood out from its competition in its breadth of coverage andits accessible, intutiont-based style. This book was aimed atindividual students and engineers excited about the broad span ofaudio processing and curious to understand the availabletechniques. Since then, with the advent of the iPod in 2001,the field of digital audio and music has exploded, leading to amuch greater interest in the technical aspects of audioprocessing. This Second Edition will update and revise the originalbook to augment it with new material describing both the enablingtechnologies of digital music distribution (most significantly theMP3) and a range of exciting new research areas in automatic musiccontent processing (such as automatic transcription, musicsimilarity, etc.) that have emerged in the past five years, drivenby the digital music revolution. New chapter topics include: Psychoacoustic Audio Coding, describing MP3 and relatedaudio coding schemes based on psychoacoustic masking ofquantization noise Music Transcription, including automatically derivingnotes, beats, and chords from music signals. Music Information Retrieval, primarily focusing onaudio-based genre classification, artist/style identification, andsimilarity estimation. Audio Source Separation, including multi-microphonebeamforming, blind source separation, and the perception-inspiredtechniques usually referred to as Computational Auditory SceneAnalysis (CASA).
This book offers comprehensive coverage on the most important aspects of audio watermarking, from classic techniques to the latest advances, from commonly investigated topics to emerging research subdomains, and from the research and development achievements to date, to current limitations, challenges, and future directions. It also addresses key topics such as reversible audio watermarking, audio watermarking with encryption, and imperceptibility control methods. The book sets itself apart from the existing literature in three main ways. Firstly, it not only reviews classical categories of audio watermarking techniques, but also provides detailed descriptions, analysis and experimental results of the latest work in each category. Secondly, it highlights the emerging research topic of reversible audio watermarking, including recent research trends, unique features, and the potentials of this subdomain. Lastly, the joint consideration of audio watermarking and encryption is also reviewed. With the help of this concept, more secure audio watermarking systems can be developed, which meet the requirements for security and privacy in cloud-based networks and systems. Accordingly, the book serves as a tutorial suitable for readers with a general knowledge of audio signal processing as well as experts in related areas, helping these readers understand the basic principles and the latest advances, concepts and applications of audio watermarking.
A digital filter can be pictured as a "black box" that accepts a sequence of numbers and emits a new sequence of numbers. In digital audio signal processing applications, such number sequences usually represent sounds. For example, digital filters are used to implement graphic equalizers and other digital audio effects. This book is a gentle introduction to digital filters, including mathematical theory, illustrative examples, some audio applications, and useful software starting points. The theory treatment begins at the high-school level, and covers fundamental concepts in linear systems theory and digital filter analysis. Various "small" digital filters are analyzed as examples, particularly those commonly used in audio applications. Matlab programming examples are emphasized for illustrating the use and development of digital filters in practice.
Jonathan Sterne shows that understanding the historical meaning of the MP3, the world's most common format for recorded audio, involves rethinking the place of digital technologies in the broader universe of twentieth-century communication history.
Summary Programming for Musicians and Digital Artists: Creating Music with ChucK offers a complete introduction to programming in the open source music language ChucK. In it, you'll learn the basics of digital sound creation and manipulation while you discover the ChucK language. As you move example-by-example through this easy-to-follow book, you'll create meaningful and rewarding digital compositions and "instruments" that make sound and music in direct response to program logic, scores, gestures, and other systems connected via MIDI or the network. Purchase of the print book includes a free eBook in PDF, Kindle, and ePub formats from Manning Publications. About this Book A digital musician must manipulate sound precisely. ChucK is an audio-centric programming language that provides precise control over time, audio computation, and user interface elements like track pads and joysticks. Because it uses the vocabulary of sound, ChucK is easy to learn even for artists with little or no exposure to computer programming. Programming for Musicians and Digital Artists offers a complete introduction to music programming. In it, you'll learn the basics of digital sound manipulation while you learn to program using ChucK. Example-by-example, you'll create meaningful digital compositions and "instruments" that respond to program logic, scores, gestures, and other systems connected via MIDI or the network. You'll also experience how ChucK enables the on-the-fly musical improvisation practiced by communities of "live music coders" around the world. Written for readers familiar with the vocabulary of sound and music. No experience with computer programming is required. What's Inside Learn ChucK and digital music creation side-by-side Invent new sounds, instruments, and modes of performance Written by the creators of the ChucK language About the Authors Perry Cook, Ajay Kapur, Spencer Salazar, and Ge Wang are pioneers in the area of teaching and programming digital music. Ge is the creator and chief architect of the ChucK language. Table of Contents Introduction: ChucK programming for artistsPART 1 INTRODUCTION TO PROGRAMMING IN CHUCK Basics: sound, waves, and ChucK programming Libraries: ChucK's built-in tools Arrays: arranging and accessing your compositional data Sound files and sound manipulation Functions: making your own tools PART 2 NOW IT GETS REALLY INTERESTING! Unit generators: ChucK objects for sound synthesis and processing Synthesis ToolKit instruments Multithreading and concurrency: running many programs at once Objects and classes: making your own ChucK power tools Events: signaling between shreds and syncing to the outside world Integrating with other systems via MIDI, OSC, serial, and more
A comprehensive and mathematically accessible introduction to digital signal processing, covering theory, advanced topics, and applications.
A best-seller in its print version, this comprehensive CD-ROM reference contains unique, fully searchable coverage of all major topics in digital signal processing (DSP), establishing an invaluable, time-saving resource for the engineering community. Its unique and broad scope includes contributions from all DSP specialties, including: telecommunications, computer engineering, acoustics, seismic data analysis, DSP software and hardware, image and video processing, remote sensing, multimedia applications, medical technology, radar and sonar applications
Speech Coding
Signal Compression
The MPEG-4 Book
An Introduction to Computer Programming and Digital Signal Processing in MATLAB
The Audio Programming Book
Audio Coding
Digital Signal Processing
Processing and Perception of Speech and Music
Speech and Audio Signal Processing
Advanced Technologies and Models
The Art of Sound Reproduction

This book presents an introduction to the principles of the fast Fourier transform. This book covers FFTs, frequency domain filtering, and applications to video and audio signal processing. As fields like communications, speech and image processing, and related areas are rapidly developing, the FFT as one of essential parts in digital signal processing has been widely used. Thus there is a pressing need from instructors and students for a book dealing with the latest FFT topics. This book provides thorough and detailed explanation of important or up-to-date FFTs. It also has adopted modern approaches like MATLAB examples and projects for better understanding of diverse FFTs.
An in-depth treatment of algorithms and standards for perceptual coding of high-fidelity audio, this self-contained reference surveys and addresses all aspects of the field. Coverage includes signal processing and perceptual (psychoacoustic) fundamentals, details on relevant research and signal models, details on standardization and applications, and details on performance measures and perceptual measurement systems. It includes a comprehensive bibliography with over 600 references, computer exercises, and MATLAB-based projects for use in EE multimedia, computer science, and DSP courses. An ftp site containing supplementary material such as wave files, MATLAB programs and workspaces for the students to solve some of the numerical problems and computer exercises in the book can be found at ftp://ftp.wiley.com/public/sci_tech_med/audio_signal
Digital Audio Theory: A Practical Guide bridges the fundamental concepts and equations of digital audio with their real-world implementation in an accessible introduction, with dozens of programming examples and projects. Starting with digital audio conversion, then segueing into filtering, and finally real-time spectral processing, Digital Audio Theory introduces the uninitiated reader to signal processing principles and techniques used in audio effects and virtual instruments that are found in digital audio workstations. Every chapter includes programming snippets for the reader to hear, explore, and experiment with digital audio concepts. Practical projects challenge the reader, providing hands-on experience in designing real-time audio effects, building FIR and IIR filters, applying noise reduction and feedback control, measuring impulse responses, software synthesis, and much more. Music technologists, recording engineers, and students of these fields will welcome Bennett's approach, which targets readers with a background in music, sound, and recording. This guide is suitable for all levels of knowledge in mathematics, signals and systems, and linear circuits. Code for the programming examples and accompanying videos made by the author can be found on the companion website, DigitalAudioTheory.com.
Introduction -- Foundations of television -- Digital video and audio coding -- Digital signal processing -- Video data compression -- Audio data compression -- Digital audio production -- Digital video production -- The MPEG multiplex -- Broadcasting digital video -- Consumer digital technology -- The future.
The requirements for multimedia (especially video and audio) communications increase rapidly in the last two decades in broad areas such as television, entertainment, interactive services, telecommunications, conference, medicine, security, business, traffic, defense and banking. Video and audio coding standards play most important roles in multimedia communications. In order to meet these requirements, series of video and audio coding standards have been developed such as MPEG-2, MPEG-4, MPEG-21 for audio and video by ISO/IEC, H.26x for video and G.72x for audio by ITU-T, Video Coder 1 (VC-1) for video by the Society of Motion Picture and Television Engineers (SMPTE) and RealVideo (RV) 9 for video by Real Networks. AVS China is the abbreviation for Audio Video Coding Standard of China. This new standard includes four main technical areas, which are systems, video, audio and digital copyright management (DRM), and some supporting documents such as consistency verification. The second part of the standard known as AVS1-P2 (Video - Jizhun) was approved as the national standard of China in 2006, and several final drafts of the standard have been completed, including AVS1-P1 (System - Broadcast), AVS1-P2 (Video - Zengqiang), AVS1-P3 (Audio - Double track), AVS1-P3 (Audio - 5.1), AVS1-P7 (Mobile Video), AVS-S-P2 (Video) and AVS-S-P3 (Audio). AVS China provides a technical solution for many applications such as digital broadcasting (SDTV and HDTV), high-density storage media, Internet streaming media, and will be used in the domestic IPTV, satellite and possibly the cable TV market. Comparing with other coding standards such as H.264 AVC, the advantages of AVS video standard include similar performance, lower complexity, lower implementation cost and licensing fees. This standard has attracted great deal of attention from industries related to television, multimedia communications and even chip manufacturing from around the world. Also many well known companies have joined the AVS Group to be Full Members or Observing Members. The 163 members of AVS Group include Texas Instruments (TI) Co., Agilent Technologies Co. Ltd., Envivio Inc., NDS, Philips Research East Asia, Aisino Corporation, LG, Alcatel Shanghai Bell Co. Ltd., Nokia (China) Investment (NCIC) Co. Ltd., Sony (China) Ltd., and Toshiba (China) Co. Ltd. as well as some high level universities in China. Thus there is a pressing need from the instructors, students, and engineers for a book dealing with the topic of AVS China and its performance comparisons with similar standards such as H.264, VC-1 and RV-9.
This authoritative guide to multimedia networking balances just the right amount of theory with practical design and integration knowledge.
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.
Video, Speech, and Audio Signal Processing and Associated Standards
Introduction to Digital Audio
Multimedia Networking
Theory and Applications
Principles of Digital Audio
Hack Audio
Digital Audio Broadcasting
with Code-Excited Linear Prediction
Digital Audio Watermarking Techniques and Technologies: Applications and Benchmarks
Fundamentals, Techniques and Challenges
Speech and Audio Coding for Wireless and Network Applications

This invaluable monograph addresses the specific needs of audio-engineering students and researchers who are either learning about the topic or using it as a reference book on multichannel audio compression. This book covers a wide range of knowledge on perceptual audio coding, from basic digital signal processing and data compression techniques to advanced audio coding standards and innovate coding tools. It is the only book available on the market that solely focuses on the principles of high-quality audio codec design for multichannel sound sources. This book includes three parts. The first part covers the basic topics on audio compression, such as quantization, entropy coding, psychoacoustic model, and sound quality assessment. The second part of the book highlights the current most prevalent low-bit-rate high-performance audio coding standards-MPEG-4 audio. More space is given to the audio standards that are capable of supporting multichannel signals, that is, MPEG advance audio coding (AAC), including the original MPEG-2 AAC technology, additional MPEG-4 toolsets, and the most recent aacPlus standard. The third part of this book introduces several innovate multichannel audio coding tools, which have been demonstrated to further improve the coding performance and expand the available functionalities of MPEG AAC, and is more suitable for graduate students and researchers in the advanced level. Dai Tracy Yang is currently Postdoctoral Research Fellow, Chris Kyriakakis is Associated Professor, and C.-C. Jay Kuo is Professor, all affiliated with the Integrated Media Systems Center (IMSC) at the University of Southern California.
This book provides scientific understanding of the most central techniques used in speech coding both for advanced students as well as professionals with a background in speech audio and or digital signal processing. It provides a clear connection between the Why ' s?, How ' s?, and What ' s, such that the necessity, purpose and solutions provided by tools should be always within sight, as well as their strengths and weaknesses in each respect. Equivalently, this book sheds light on the following perspectives for each technology presented: Objective: What do we want to achieve and especially why is this goal important? Resource / Information: What information is available and how can it be useful? Resource / Platform: What kind of platforms are we working with and what are the capabilities/restrictions of those platforms? This includes properties such as computational, memory, acoustic and transmission capacity of devices used. Solutions: Which solutions have been proposed and how can they be used to reach the stated goals? Strengths and weaknesses: In which ways do the solutions fulfill the objectives and where are they insufficient? Are resources used efficiently? This book concentrates solely on code excited linear prediction and its derivatives since mainstream speech codecs are based on linear prediction It also concentrates exclusively on time domain techniques because frequency domain tools are to a large extent common with audio codecs.
Now the standardisation work of DAB (Digital Audio Broadcasting) system is finished many broadcast organisations, network providers and receiver manufacturers in European countries and outside of Europe (for example Canada and the Far East) will be installing DAB broadcast services as pilot projects or public services. In addition some value added services (data and video services) are under development or have already started as pilot projects. The new digital broadcast system DAB distinguishes itself from existing conventional broadcast systems, and the various new international standards and related documents (from ITU-R, ISO/IEC, ETSI, EBU, EUREKA147, and others) are not readily available and are difficult to read for users. Therefore it is essential that a well structured technical handbook should be available. The Second Edition of Digital Audio Broadcasting has been fully updated with new sections and chapters added to reflect all the latest developments and advances. Digital Audio Broadcasting: Provides a fully updated comprehensive overview of DAB Covers international standards, applications and other technical issues Combines the expertise of leading researchers in the field of DAB Now covers such new areas as: IP-Tunneling via DAB; Electronic Programme Guide for DAB; and Metadata A comprehensive overview of DAB specifically written for planning and system engineers, developers for professional and domestic equipment manufacturers, service providers, as well as postgraduate students and lecturers in communications technology.
In this book, two leaders of the MPEG-4 standards community offer an in-depth, targeted guide to the MPEG-4 standard and its use in real, cutting-edge applications. The authors demonstrate how MPEG-4 addresses the rapidly evolving needs of telecommunications, broadcast, interactive, and converged applications more successfully than any previous standard.
Presents digital audio watermarking as a new and alternative method to enforce intellectual property rights and protect digital audio from tampering. Provides theoretical frameworks, recent research findings, and practical applications.
Introduction to Digital Audio Coding and Standards provides a detailed introduction to the methods, implementations, and official standards of state-of-the-art audio coding technology. In the book, the theory and implementation of each of the basic coder building blocks is addressed. The building blocks are then fit together into a full coder and the reader is shown how to judge the performance of such a coder. Finally, the authors discuss the features, choices, and performance of the main state-of-the-art coders defined in the ISO/IEC MPEG and HDTV standards and in commercial use today. The ultimate goal of this book is to present the reader with a solid enough understanding of the major issues in the theory and implementation of perceptual audio coders that they are able to build their own simple audio codec. There is no other source available where a non-professional has access to the true secrets of audio coding.
The professional recording industry is rapidly moving from a hardware paradigm (big studios with expensive gear) to a software paradigm, in which lots of expensive hardware is replaced with a single computer loaded with software plug-ins. Complete albums are now being recorded and engineered "inside the box"-all within a computer without hardware processing or mixing gear. Audio effect plug-ins, which are small software modules that work within audio host applications, like Avid Pro Tools, Apple Logic, Ableton Live, and Steinberg Cubase, are big business. Designing Audio Effect Plug-Ins in C++ gives readers everything they need to know to create real-world, working plug-ins in the widely used C++ programming language. Beginning with the necessary theory behind audio signal processing, author Will Pirkle quickly gets into the heart of this implementation guide, with clearly-presented, previously unpublished algorithms, tons of example code, and practical advice. From the companion website, readers can download free software for the rapid development of the algorithms, many of which have never been revealed to the general public. The resulting plug-ins can be compiled to snap in to any of the above host applications. Readers will come away with the knowledge and tools to design and implement their own audio signal processing designs. Learn to build audio effect plug-ins in a widely used, implementable programming language-C++ Design plug-ins for a variety of platforms (Windows and Mac) and popular audio applications Companion site gives you fully worked-out code for all the examples used, free development software for download, video tutorials for the software, and examples of student plug-ins complete with theory and code
Designing Audio Effect Plug-ins in C++ with Digital Audio Signal Processing Theory
MP3
AVS China, H.264/MPEG-4 PART 10, HEVC, VP6, DIRAC and VC-1
Newnes Guide to Digital TV
Introduction to Digital Filters
Introduction To Digital Audio Coding And Standards
A Practical Guide
Digital Audio Signal Processing
From Theory to Practice
With Audio Applications
Video coding standards

Master the basics from first principles: the physics of sound, principles of hearing etc, then progress onward to fundamental digital principles, conversion, compression and coding and then onto transmission, digital audio workstations, DAT and optical disks. Get up to speed with how digital audio is used within DVD, Digital Audio Broadcasting, networked audio and MPEG transport streams. All of the key technologies are here: compression, DAT, DAB, DVD, SACD, oversampling, noise shaping and error correction theories are treated in a simple yet accurate form. Thoroughly researched, totally up-to-date and technically accurate this is the only book you need on the subject.
Introduction to Data Compression, Fifth Edition, builds on the success of what is widely considered the best introduction and reference text on the art and science of data compression. Data compression techniques and technology are ever-evolving with new applications in image, speech, text, audio and video. This new edition includes all the latest developments in the field. Khalid Sayood provides an extensive introduction to the theory underlying today's compression techniques, with detailed instruction for their applications using several examples to explain the concepts. Encompassing the entire field of data compression, the book includes lossless and lossy compression, Huffman coding, arithmetic coding, dictionary techniques, context based compression, and scalar and vector quantization. The book provides a comprehensive working knowledge of data compression, giving the reader the tools to develop a complete and concise compression package. Explains established and emerging standards in- depth, including JPEG 2000, JPEG-LS, MPEG-2, H.264, JBIG 2, ADPCM, LPC, CELP, MELP, iLBC and the new HEVC standard Includes more coverage of lattices in vector quantization Contains improved and expanded end-of-chapter problems Source code is provided via a companion website that gives readers the opportunity to build their own algorithms and choose and implement techniques in their own applications
Audio Coding: Theory and Applications provides succinct coverage of audio coding technologies that are widely used in modern audio coding standards. Delivered from the perspective of an engineer, this book articulates how signal processing is used in the context of audio coding. It presents a detailed treatment of contemporary audio coding technologies and then uses the DRA audio coding standard as a practical example to illustrate how numerous technologies are integrated into a fully-fledged audio coding algorithm. Drawing upon years of practical experience and using numerous examples and illustrations Dr. Yuli You, gives a description of practical audio coding technologies including: • Designing high-performance algorithms that can be readily implemented on fixed-point or integer microprocessors. • How to properly implement an audio decoder on various microprocessors. Transient detection and adaptation of time-frequency resolution of subband filters. • Psychoacoustic models and optimal bit allocation. Audio Coding: Theory and Applications will be a valuable reference book for engineers in the consumer electronics industry, as well as students and researchers in electrical engineering.
Designed to make life a little easier by providing all the theoretical background necessary to understand sound reproduction, backed up with practical examples. Specialist terms - both musical and physical - are defined as they occur and plain English is used throughout. Analog and digital audio are considered as alternatives, and the advantages of both are stressed. Audio is only as good as the transducers employed, and consequently microphone and loudspeaker technology also feature heavily - making this the most comprehensive, up-to-date text currently available on all aspects of sound reproduction.
This textbook presents the fundamentals of audio coding, used to compress audio and music signals, using Python programs both as examples to illustrate the principles and for experiments for the reader. Together, these programs then form complete audio coders. The author starts with basic knowledge of digital signal processing (sampling, filtering) to give a thorough introduction to filter banks as used in audio coding, and their design methods. He then continues with the next core component, which are psycho-acoustic models. The author finally shows how to design and implement them. Lastly, the author goes on to describe components for more specialized coders, like the Integer-to-Integer MDCT filter bank, and predictive coding for lossless and low delay coding. Included are Python program examples for each section, which illustrate the principles and provide the tools for experiments. Comprehensively explains the fundamentals of filter banks and audio coding; Provides Python examples for each principle so that completed audio coders are obtained in the language; Includes a suite of classroom materials including exercises, experiments, and examples.
Speech and Audio Coding for Wireless and Network Applications contains 34 chapters, loosely grouped into six topical areas. The chapters in this volume reflect the progress and present the state of the art in low-bit-rate speech coding, primarily at bit rates from 2.4 kbit/s to 16 kbit/s. Together they represent important contributions from leading researchers in the speech coding community. Speech and Audio Coding for Wireless and Network Applications contains contributions describing technologies that are under consideration as standards for such applications as digital cellular communications (the half-rate American and European coding standards). A brief Introduction is followed by a section dedicated to low-delay speech coding, a research direction which emerged as a result of the CCITT requirement for a universal low-delay 16 kbit/s speech coding technology and now continues with the objective of achieving toll quality with moderate delay at a rate of 8 kbit/s. A section on the important topic of speech quality evaluation is then presented. This is followed by a section on speech coding for wireless transmission, and a section on audio coding which covers not only 7 kHz bandwidth speech, but also wideband coding applicable to high fidelity music. The book concludes with a section on speech coding for noisy transmission channels, followed by a section addressing future research directions. Speech and Audio Coding for Wireless and Network Applications presents a cross-section of the key contributions in speech and audio coding which have emerged recently. For this reason, the book is a valuable reference for all researchers and graduate students in the speech coding community.
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.
Introduction to Digital Audio Coding and Standards
Compressing Audio Signals Using Python
Coding of Speech, Audio, Text, Image and Video
Principles and Applications of Digital Radio
Audio Signal Processing for Next-Generation Multimedia Communication Systems
Digital Audio Watermarking
Digital Signal Processing: World Class Designs
Principles and Applications
Digital Signal Processing Handbook on CD-ROM
Audio Signal Processing and Coding
Machine Learning for Audio, Image and Video Analysis

"This book focuses on watermarking, in which data is marked with hidden ownership information, as a promising solution to copyright protection issues and deals with understanding human perception processes and including them in effective psychoacoustic models"--
An encyclopedic handbook on audio programming for students and professionals, with many cross-platform open source examples and a DVD covering advanced topics. This comprehensive handbook of mathematical and programming techniques for audio signal processing will be an essential reference for all computer musicians, computer scientists, engineers, and anyone interested in audio. Designed to be used by readers with varying levels of programming expertise, it not only provides the foundations for music and audio development but also tackles issues that sometimes remain mysterious even to experienced software designers. Exercises and copious examples (all cross-platform and based on free or open source software) make the book ideal for classroom use. Fifteen chapters and eight appendixes cover such topics as programming basics for C and C++ (with music-oriented examples), audio programming basics and more advanced topics, spectral audio programming; programming Csound opcodes, and algorithmic synthesis and music programming. Appendixes cover topics in compiling, audio and MIDI, computing, and math. An accompanying DVD provides an additional 40 chapters, covering musical and audio programs with micro-controllers, alternate MIDI controllers, video controllers, developing Apple Audio Unit plug-ins from Csound opcodes, and audio programming for the iPhone. The sections and chapters of the book are arranged progressively and topics can be followed from chapter to chapter and from section to section. At the same time, each section can stand alone as a self-contained unit. Readers will find The Audio Programming Book a trustworthy companion on their journey through making music and programming audio on modern computers.
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. This volume, Video, Speech, and Audio Signal Processing and Associated Standards, provides thorough coverage of the basic foundations of speech, audio, image, and video processing and associated applications to broadcast, storage, search and retrieval, and communications.
All the design and development inspiration and direction an digital engineer needs in one blockbuster book! Kenton Williston, author, columnist, and editor of DSP DesignLine has selected the very best digital signal processing design material from the Newnes portfolio and has compiled it into this volume. The result is a book covering the gamut of DSP design'from design fundamentals to optimized multimedia techniques'with a strong pragmatic emphasis. In addition to specific design techniques and practices, this book also discusses various approaches to solving DSP design problems and how to successfully apply theory to actual design tasks. The material has been selected for its timelessness as well as for its relevance to contemporary embedded design issues. CONTENTS: Chapter 1 ADCs, DACs, and Sampling Theory Chapter 2 Digital Filters Chapter 3 Frequency Domain Processing Chapter 4 Audio Coding Chapter 5 Video Processing Chapter 6 Modulation Chapter 7 DSP Hardware Options Chapter 8 DSP Processors and Fixed-Point Arithmetic Chapter 9 Code Optimization and Resource Partitioning Chapter 10 Testing and Debugging DSP Systems *Hand-picked content selected by Kenton Williston, Editor of DSP DesignLine *Proven best design practices for image, audio, and video processing *Case histories and design examples get you off and running on your current project
A fully updated second edition of the excellent Digital Audio Signal Processing Well established in the consumer electronics industry, Digital Audio Signal Processing (DASP) techniques are used in audio CD, computer music and multi-media components. In addition, the applications afforded by this versatile technology now range from real-time signal processing to room simulation. Digital Audio Signal Processing, Second Edition covers the latest signal processing algorithms for audio processing.

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

This second edition focuses on audio, image and video data, the three main types of input that machines deal with when interacting with the real world. A set of appendices provides the reader with self-contained introductions to the mathematical background necessary to read the book. Divided into three main parts, From Perception to Computation introduces methodologies aimed at representing the data in forms suitable for computer processing, especially when it comes to audio and images. Whilst the second part, Machine Learning includes an extensive overview of statistical techniques aimed at addressing three main problems, namely classification (automatically assigning a data sample to one of the classes belonging to a predefined set), clustering (automatically grouping data samples according to the similarity of their properties) and sequence analysis (automatically mapping a sequence of observations into a sequence of human-understandable symbols). The third part Applications shows how the abstract problems defined in the second part underlie technologies capable to perform complex tasks such as the recognition of hand gestures or the transcription of handwritten data. Machine Learning for Audio, Image and Video Analysis is suitable for students to acquire a solid background in machine learning as well as for practitioners to deepen their knowledge of the state-of-the-art. All application chapters are based on publicly available data and free software packages, thus allowing readers to replicate the experiments.

Audio Signal Processing for Next-Generation Multimedia Communication Systems presents cutting-edge digital signal processing theory and implementation techniques for problems including speech acquisition and enhancement using microphone arrays, new adaptive filtering algorithms, multichannel acoustic echo cancellation, sound source tracking and separation, audio coding, and realistic sound stage reproduction. This book's focus is almost exclusively on the processing, transmission, and presentation of audio and acoustic signals in multimedia communications for telecollaboration where immersive acoustics will play a great role in the near future.