

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

An authoritative reference on all aspects of audio engineering and technology including basic mathematics and formulae, acoustics and psychoacoustics, microphones, loudspeakers and studio installations. Compiled by an international team of experts, the second edition was updated to keep abreast of fast-moving areas such as digital audio and transmission technology. Much of the material has been revised, updated and expanded to cover the very latest techniques. This is a new paperback version. 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.

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.

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),

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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 innovative 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.

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.

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

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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.

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.

Digital Audio Forensics Fundamentals offers an accessible introduction to both the theory and practical skills behind this emerging field of forensic science. Beginning with an overview of the history of the discipline, the reader is guided through forensic principles and key audio concepts, before being introduced to practical areas such as audio enhancement, audio authentication, and the presentation of reports. Covering all aspects of audio forensics from the capture to the courtroom, this book is pivotal reading for beginners entering the field, as well as experienced professionals looking to develop their knowledge of the practice.

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.

Described as "the most comprehensive book on digital audio to date", it is widely acclaimed as an industry "bible". Covering the very latest developments in digital audio technology, it provides an thorough introduction to the theory as well as acting as an authoritative and comprehensive professional reference source. Everything you need is here from the fundamental principles to the latest applications, written in an award-winning style with clear explanations from first principles. New material covered includes internet audio, PC audio technology, DVD, MPEG audio compression, digital audio broadcasting and audio networks. Whether you are in the field of audio engineering, sound

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recording, music technology, broadcasting and communications media or audio design and installation, this book has it all. Written by a leading international audio specialist, who conducts professional seminars and workshops around the world, the book has been road tested for many years by professional seminar attendees and students to ensure their needs are taken into account, and all the right information is covered. This new edition now includes: Internet audio PC Audio technology DVD MPEG Audio compression Digital Audio Broadcasting Audio networks Digital audio professionals will find everything they need here, from the fundamental principles to the latest applications, written in an award-winning style with clear explanations from first principles. John Watkinson is an international consultant in audio, video and data recording. He is a Fellow of the AES, a member of the British Computer Society and a chartered information systems practitioner. He presents lectures, seminars, conference papers and training courses worldwide. He is the author of many other Focal Press books, including: the Kraszna-Krausz award winning MPEG-2; The Art of Digital Audio; An Introduction to Digital Video; The Art of Sound Reproduction; An Introduction to Digital Audio; TV Fundamentals and Audio for Television. He is also co-author, with Francis Rumsey, of The Digital Interface Handbook, and contributor to the Loudspeaker and Headphone Handbook, 3rd edition.

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.

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 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

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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.

This essential text for any technician in broadcasting deals with all the most important digital television, sound radio and multimedia standards. The book provides an in-depth look at these subjects in terms of practical experience. In addition it contains chapters on the basics of technologies such as analog television, digital modulation, COFDM or mathematical transformations between time and frequency domains. The attention in each respective field under discussion is focused on aspects of measuring techniques and of measuring practice, in each case consolidating the knowledge imparted with numerous practical examples. Since the entire field of electrical communications technology is traversed in a wide arc, those who are students in this field are not excluded either.

Introduction to Digital Speech Processing highlights the central role of DSP techniques in modern speech communication research and applications. It presents a comprehensive overview of digital speech processing that ranges from the basic nature of the speech signal, through a variety of methods of representing speech in digital form, to applications in voice communication and automatic synthesis and recognition of speech. Introduction to Digital Speech Processing provides the reader with a practical introduction to the wide range of important concepts that comprise the field of digital speech processing. It serves as an invaluable reference for students embarking on speech research as well as the experienced researcher already working in the field, who can utilize the book as a reference guide.

Explains how to use the portable music player with a Windows PC or a Macintosh computer to perform functions including play music, store personal contact and calendar information, and use as a portable FireWire drive.

Digital Video offers comprehensive coverage of the MPEG-2 audio/visual digital compression standard. The treatment includes the specifics needed to implement an MPEG-2 Decoder, including the syntax and semantics of the coded bitstreams. Since the MPEG-2 Encoders are not specified by the standard, and are actually closely held secrets of many vendors, the book only outlines the fundamentals of encoder design and algorithm optimization.

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.

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.

This textbook presents an introduction to signal processing for audio applications. The author's approach posits that math is at the heart of audio processing and that it should not be simplified. He thus retains math as the core of signal processing and includes concepts of difference equations, convolution, and the Fourier Transform. Each of these is presented in a context where they make sense to the student and can readily be applied to build artifacts. Each chapter in the book builds on the previous ones, building a linear, coherent story. The book starts with a definition of sound and goes on to discuss digital audio signals, filters, The Fourier Transform, audio effects, spatial effects, audio equalizers, dynamic range control, and pitch estimation. The exercises in each chapter cover the application of the concepts to audio signals. The exercises are made specifically for Pure Data (Pd) although traditional software, such as MATLAB, can be

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used. The book is intended for students in media technology bachelor programs. The book is based on material the author developed teaching on the topic over a number of years.

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.

"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"--

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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. 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

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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.

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

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.

When Speech and Audio Signal Processing published in 1999, it stood out from its competition in its breadth of coverage and its accessible, intuition-based style. This book was aimed at individual students and engineers excited about the broad span of audio processing and curious to understand the available techniques. Since then, with the advent of the iPod in 2001, the field of digital audio and music has exploded, leading to a much greater interest in the technical aspects of audio processing. This Second Edition will update and revise the original book to augment it with new material describing both the enabling technologies of digital music distribution (most significantly the MP3) and a range of exciting new research areas in automatic music content processing (such as automatic transcription, music similarity, etc.) that have emerged in the past five years, driven by the digital music revolution. New chapter topics include: Psychoacoustic Audio Coding, describing MP3 and related audio coding schemes based on psychoacoustic masking of quantization noise Music Transcription, including automatically deriving notes, beats, and chords from music signals. Music Information Retrieval, primarily focusing on audio-based genre classification,

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artist/style identification, and similarity estimation. Audio Source Separation, including multi-microphone beamforming, blind source separation, and the perception-inspired techniques usually referred to as Computational Auditory Scene Analysis (CASA).

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

The MPEG-1 Layer III (MP3) algorithm is one of the most successful audio formats for consumer audio storage and for transfer and playback of music on digital audio players. The MP3 compression standard along with the AAC (Advanced Audio Coding) algorithm are associated with the most successful music players of the last decade. This book describes the fundamentals and the MATLAB implementation details of the MP3 algorithm. Several of the tedious processes in MP3 are supported by demonstrations using MATLAB software. The book presents the theoretical concepts and algorithms used in the MP3 standard. The implementation details and simulations with MATLAB complement the theoretical principles. The extensive list of references enables the reader to perform a more detailed study on specific aspects of the algorithm and gain exposure to advancements in perceptual coding. Table of Contents: Introduction / Analysis Subband Filter Bank / Psychoacoustic Model II / MDCT / Bit Allocation, Quantization and Coding / Decoder

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.

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.

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

Summary Programming for Musicians and Digital Artists: Creating Music with ChuckK offers a complete introduction to programming in the open source music language ChuckK. In it, you'll learn the basics of digital sound creation and manipulation while you discover the ChuckK 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. ChuckK 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, ChuckK 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 ChuckK. 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 ChuckK 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

