Digital Coding Of Audio And Video Link Springer

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

Combines both the DSP principles and real-timeimplementations and applications, and now updated with the neweZdsp USB Stick, which is very low cost, portable and widelyemployed at many DSP labs. Real-Time Digital Signal Processing introducesfundamental digital signal processing (DSP) principles and will beupdated to include the latest DSP applications, introduce newsoftware development tools and adjust the software design processto reflect the latest advances in the field. In the 3rdedition of the book, the key aspect of hands-on experiments will beenhanced to make the DSP principles more interesting and directlyinteract with the real-world applications. All of the programs willbe carefully updated using the most recent version of softwaredevelopment tools and the new TMS320VC5505 eZdsp USB Stick forreal-time experiments. Due to its lower cost and portability, thenew software and hardware tools are now widely used in universitylabs and in commercial industrial companies to replace the olderand more expensive generation. The new edition will have a renewedfocus on real-time applications and will offer step-by-stephands-on experiments for a complete design cycle starting fromfloating-point C language program to fixed-point C implementation,code optimization using INTRINSICS, and mixed C-and-assemblyprogramming on fixed-point DSP processors. This new methodologyenables readers to concentrate on learning DSP fundamentals andinnovative applications by relaxing the intensive programmingefforts, namely, the traditional DSP assembly coding efforts. Thebook is organized into two parts; Part One introduces thedigital signal processing principles and theories, and PartTwo focuses on practical applications. The topics for theapplications are the extensions of the theories in Part One with anemphasis placed on the hands-on experiments, systematic design andimplementation approaches. The applications provided in the bookare carefully chosen to reflect current advances of DSP that are ofmost relevance for the intended readership. Combines both the DSP principles and real-time implementationsand applications using the new eZdsp USB Stick, which is very lowcost, portable and widely employed at many DSP labs is now used inthe new edition Places renewed emphasis on C-code experiments and reduces theexercises using assembly coding; effective use of C programming,fixed-point C code and INTRINSICS will become the main focus of thenew edition. Updates to application areas to reflect latest advances such asspeech coding techniques used for next generation networks (NGN),audio coding with surrounding sound, wideband speech codec (ITUG.722.2 Standard), fingerprint for image processing, and biomedicalsignal processing examples. Contains new addition of several projects that can be used assemester projects; as well as new many new real-time experimentsusing TI's binary libraries – the experiments areprepared with flexible interface and modular for readers to adaptand modify to create other useful applications from the providedbasic programs. Consists of more MATLAB experiments, such as filter design,algorithm evaluation, proto-typing for C-code architecture, andsimulations to aid readers to learn DSP fundamentals. Includes supplementary material of program and data files forexamples, applications, and experiments hosted on a companionwebsite. A valuable resource for Postgraduate students enrolled on DSPcourses focused on DSP implementation & applications as well asSenior undergraduates studying DSP; engineers and programmers whoneed to learn and use DSP principles and development tools fortheir projects.

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.

Information Technology, Coding of Moving Pictures and Associated Audio for Digital Storage Media Up to about 1,5 Mbit/s

Compressing Audio Signals Using Python

Introduction to Digital Audio

Advanced Technologies and Models

Designing Audio Effect Plug-ins in C++ with Digital Audio Signal Processing Theory

Information Technology - Coding of Moving Pictures and Associated Audio for Digital Storage Media at Up to about 1,5 Mbit/s

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.

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.

The definitive guide to digital engineering--fully updated Gain a thorough understanding of digital audio tools, techniques, and practices from this completely revised and expanded resource. Written by industry pioneer and Audio Engineering Society Fellow Ken C. Pohlmann, Principles of Digital Audio, Sixth Edition, describes the technologies behind today's audio equipment in a clear, practical style. Covering basic theory to the latest technological advancements, the book explains how to apply digital conversion, processing, compression, storage, streaming, and transmission concepts. New chapters on Blu-ray, speech coding, and low bit-rate coding are also included in this bestselling guide. Learn about discrete time sampling, quantization, and signal processing Examine details of CD, DVD, and Blu-ray players and discs Encode and decode AAC, MP3, MP4, Dolby Digital, and other files Prepare content for distribution via the Internet and digital radio and television Learn the critical differences between music coding and speech coding Design low bit-rate codecs to optimize memory capacity while preserving fidelity Develop methodologies to evaluate the sound quality of music and speech files Study audio transmission via HDMI, VoIP, Wi-Fi, and Bluetooth Handle digital rights management, fingerprinting, and watermarking Understand how one-bit conversion and high-order noise shaping work

Watermarking in Audio

Audio Signal Processing and Coding

The Art of Digital Audio

Theory and Applications

Digital Audio Compression Using Subband Coding and Perceptual Masking

Signal Compression

The availability of increased computational power and the proliferation of the Internet have facilitated the production and distribution of unauthorized copies of multimedia information. As a result, the problem of multimedia copyright protection has attracted the interest of the worldwide scientific and the business communities. The most promising solution seems to be the watermarking process where the original data is marked with ownership information hidden in an imperceptible manner in the original signal. Watermarking in Audio: Key Techniques and Technologies is an inclusive compilation of the most important and fundamental theories and techniques in digital audio watermarking. It includes a comprehensive overview of the state-of-the-art techniques used in digital audio watermarking and focuses on two key issues in digital audio watermarking: psychoacoustic modeling and synchronization. The fundamental theories and the innovative techniques introduced in this book can be directly applied not only to digital audio watermarking, but also to perceptual digital audio coding. Watermarking in Audio will serve as an essential reference to the scientists and researchers in digital audio and related fields, including engineering and information technology.

With the advent of `multimedia', digital signal processing (DSP) of sound has emerged from the shadow of bandwidth limited speech processing to become a research field of its own. To date, most research in DSP applied to sound has been concentrated on speech, which is bandwidth limited to about 4 kilohertz. Speech processing is also limited by the low fidelity typically expected in the telephone network. Today, the main applications of audio DSP are high quality audio coding and the digital generation and manipulation of music signals. They share common research topics including perceptual measurement techniques and analysis/synthesis methods. Additional important topics are hearing aids using signal processing technology and hardware architectures for digital signal processing of audio. In all these areas the last decade has seen a significant amount of application-oriented research. The frequency range of wideband audio has an upper limit of 20 kilohertz and the resulting difference in frequency range and Signal to Noise Ratio (SNR) due to sample size must be taken into account when designing DSP algorithms. There are whole classes of algorithms that the speech community is not interested in pursuing or using. These algorithms and techniques are revealed in this book. This book is suitable for advanced level courses and serves as a valuable reference for researchers in the field. Interested and informed engineers will also find the book useful in their work.

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.

Introduction to Digital Audio Coding and Standards

Digital Audio Theory

Creating music with ChucK

Key Techniques and Technologies

Coding of Speech, Audio, Text, Image and Video

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

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

Optimal Audio and Video Reproduction at Home is a comprehensive guide that will help every reader set up a modern audio-video system in a small room such as a home theater or studio control room. Verdult covers everything the reader needs to know to optimize the reproduction of multichannel audio and high-resolution video. The book provides concrete advice on equipment setup, display calibration, loudspeaker positioning, room acoustics, and much more. Detailed, easy-to-grasp explanations of the underlying principles ensure the reader will make the right choices, find alternatives, and separate the rigid from the more flexible requirements to achieve the best possible results.

Applications of Digital Signal Processing to Audio and Acoustics

Filter Banks and Audio Coding

Digital Audio Signal Processing

Digital Coding of Waveforms

Optimal Audio and Video Reproduction at Home

Information Technology

Digital Audio Signal Processing The fully revised new edition of the popular textbook, featuring additional MATLAB exercises and new algorithms for processing digital audio signals Digital Audio Signal Processing (DASP) techniques are used in a variety of applications, ranging from audio streaming and computer-generated music to real-time signal processing and virtual sound processing. Digital Audio Signal Processing provides clear and accessible coverage of the fundamental principles and practical applications of digital audio processing and coding. Throughout the book, the authors explain a wide range of basic audio processing techniques and highlight new directions for automatic tuning of different algorithms and discuss state- of-the-art DASP approaches. Now in its third edition, this popular guide is fully updated with the latest signal processing algorithms for audio processing. Entirely new chapters cover nonlinear processing, Machine Learning (ML) for audio applications, distortion, soft/hard clipping, overdrive, equalizers and delay effects, sampling and reconstruction, and more. Covers the fundamentals of quantization, filters, dynamic range control, room simulation, sampling rate conversion, and audio coding Describes DASP techniques, their theoretical foundations, and their practical applications Discusses modern studio technology, digital transmission systems, storage media, and home entertainment audio components Features a new introductory chapter and extensively revised content throughout Provides updated application examples and computer-based activities supported with MATLAB exercises and interactive JavaScript applets via an author-hosted companion website Balancing essential concepts and technological topics, Digital Audio Signal Processing, Third Edition remains the ideal textbook for advanced music technology and engineering students in audio signal processing courses. It is also an invaluable reference for audio engineers, hardware and software developers, and researchers in both academia and industry.

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.

Here is a thorough, not-overly-complex introduction to the three technical foundations for multimedia applications across the Internet: communications (principles, technologies and networking); compressive encoding of digital media; and Internet protocol and services. All the contributing systems elements are explained through descriptive text and numerous illustrative figures; the result is a book well-suited toward non-specialists, preferably with technical background, who need well-composed tutorial introductions to the three foundation areas. The text discusses the latest advances in digital audio and video encoding, optical and wireless communications technologies, high-speed access networks, and IP-based media streaming, all crucial enablers of the multimedia Internet.

Principles of Digital Audio

Digital Video and Audio Compression

Fundamentals, Implementations and Applications

Principles and Applications of Digital Radio

Coding of Moving Pictures and Associated Audio for Digital Storage Media at Up to about 1,5 Mbit/s. Video. Vid é o

Transform Coding of Stereophonic Digital Audio Signals

Introduction to Digital Audio Coding and StandardsSpringer Science & Business Media

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

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.

Real-Time Digital Signal Processing

Coding for Digital Recording

Introduction To Digital Audio Coding And Standards

Principles of Digital Audio, Sixth Edition

Information Technology, Coding of Moving Pictures and Associated Audio for Digital Storage Media Up to about 1,5 Mbit/s: Audio

Coding of Moving Pictures and Associated Audio for Digital Storage Media at Up to about 1.5 Mbit/s : Part 3: Audio

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.
First Published in 2000. Routledge is an imprint of Taylor & Francis, an informa company.
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.
Improving the Listening and Viewing Experience
Signal Processing, Perceptual Coding, and Watermarking of Digital Audio
Audio Coding
High-fidelity Multichannel Audio Coding
Principles and Applications to Speech and Video
Perceptual Coding of Digital Audio
More and more information, audio and video but also a range of other information type, is generated, processed and used by machines today, even though the end user may be a human. The result over the past 15 years has been a substantial increase in the type of information and change in the way humans generate, classify, store, search, access and consume information. Conversion of information to digital form is a prerequisite for this enhanced machine role, but must be done having in mind requirements such as compactness, fidelity, interpretability etc. This book presents new ways of dealing with digital information and new types of digital information underpinning the evolution of society and business.
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.
"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"--
Digital Audio Broadcasting
The Audio Programming Book
Efficient Digital Coding Schemes for Audio Signals
A Practical Guide
Programming for Musicians and Digital Artists
The Multimedia Internet