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

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.

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.

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.

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 new Digital Radio system DAB (Digital Audio Broadcasting) is a highly innovative and universal multimedia broadcast system that will replace the existing AM and FM audio broadcast services in many parts of the world in the immediate future. It is designed for excellent mobile reception, is highly robust against multipath reception and allows the use of single frequency networks (SFN) for high frequency efficiency. In addition to several high-quality digital audio services, DAB is able to transmit programme associated data and a host of other data services including travel and traffic information and still and moving pictures. Dynamic multiplex management on the network side opens up new possibilities for flexible programming. Written in an accessible style, Digital Audio Broadcasting provides an excellent guide for developers in industry, planning engineers together with broadcasters, network providers and service and content providers. For students and those wishing to get to grips with the new concepts of digital broadcasting it will serve as a comprehensive introduction to the field. * Explains the basic concepts of DAB including audio processing, data transmission and modulation schemes and how the system can be implemented and operated * Features new broadcasting components such as perceptual audio coding (MPEG-1 and MPEG-2), OFDM channel coding and modulation, multiplex management (STI) and data transmission protocols (MOT) * Focuses on the practical implications for service provision and coverage planning and the new infrastructure required in studios and broadcasting houses for multiplex and network management * Provides an insight into current receiver development strategies
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

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.

Multimedia signals include different data types (text, sound, graphics, picture, animations, video, etc.), which can be time-dependent (sound, video and animation) or spatially-dependent (images, text and graphics). Hence, the multimedia systems represent an interdisciplinary cross-section of the following areas: digital signal processing, computer architecture, computer networks and telecommunications. Multimedia Signals and Systems is an introductory text, designed for students or professionals and researchers in other fields, with a need to learn the basics of signals and systems. A considerable emphasis is placed on the analysis and processing of multimedia signals (audio, images, video). Additionally, the book connects these principles to other important elements of multimedia systems such as the analysis of optical media, computer networks, QoS, and digital watermarking.

Modelling and simulation in acoustics is currently gaining importance. In fact, with the development and improvement of innovative computational techniques and with the growing need for predictive models, an impressive boost has been observed in several research and application areas, such as noise control, indoor acoustics, and industrial applications. This led us to the proposal of a special issue about “Modelling, Simulation and Data Analysis in Acoustical Problems”, as we believe in the importance of these topics in modern acoustics’ studies. In total, 81 papers were submitted and 33 of them were published, with an acceptance rate of 37.5%. According to the number of papers submitted, it can be affirmed that this is a trending topic in the scientific and academic community and this special issue will try to provide a future reference for the research that will be developed in coming years.

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.

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.

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.

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.

Continuous media streaming systems will shape the future of information infrastructure. The challenge is to design systems and networks capable of supporting millions of concurrent users. Key to this is the integration of fault-tolerant mechanisms to prevent individual component failures from disrupting systems operations. These are just some of the hurdles that need to be overcome before large-scale continuous media services such as video-on-demand can be deployed with maximum efficiency. The author places the subject in context, drawing together findings from the past decade of research whilst examining the technology's present status and its future potential. The approach adopted is comprehensive, covering topics – notably the scalability and fault-tolerance issues - that previously have not been treated in depth. Provides an accessible introduction to the technology, presenting the basic principles for media streaming system design, focusing on the need for the correct and timely delivery of data. Explores the use of parallel server architectures to tackle the two key challenges of scalability and fault-tolerance. Investigates the use of network multicast streaming algorithms to further increase the scalability of very-large-scale media streaming systems. Illustrates all findings using real-world examples and case studies gleaned from cutting-edge worldwide research. Combining theory and practice, this book will appeal to industry specialists working in content distribution in general and continuous media streaming in particular. The introductory materials and basic building blocks complemented by amply illustrated, more advanced coverage provide essential reading for senior undergraduates, postgraduates and researchers in these fields.

This book illustrates the commonly used and novel approaches of audio watermarking for copyrights protection. The author examines the theoretical and practical step by step guide to the topic of data hiding in audio signal such as music, speech, broadcast. The book covers new techniques developed by the authors are fully explained and MATLAB programs, for audio watermarking and audio quality assessments and also discusses methods for objectively predicting the perceptual quality of the watermarked audio signals. Explains the theoretical basics of the commonly used audio watermarking techniques Discusses the methods used to objectively and subjectively assess the quality of the audio signals Provides a comprehensive well tested MATLAB programs that can be used efficiently to watermark any audio media

This authoritative guide to multimedia networking balances just the right amount of theory with practical design and integration knowledge.

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

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

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.

Sound source localization is an important research field that has attracted researchers' efforts from many technical and biomedical sciences. Sound source localization (SSL) is defined as the determination of the direction from a receiver, but also includes the distance from it. Because of the wave nature of sound propagation, phenomena such as refraction, diffraction, diffusion, reflection, reverberation and interference occur. The wide spectrum of sound frequencies that range from infrasounds through acoustic sounds to ultrasounds, also introduces difficulties, as different spectrum components have different penetration properties through the medium. Consequently, SSL is a complex computation problem and development of robust sound localization techniques calls for different approaches, including multisensor schemes, null-steering beamforming and time-difference arrival techniques. The book offers a rich source of valuable material on advances on SSL techniques and their applications that should appeal to researches representing diverse engineering and scientific disciplines.

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.

The NAB Engineering Handbook provides detailed information on virtually every aspect of the broadcast chain, from news gathering, program production and postproduction through master control and distribution links to transmission, antennas, RF propagation, cable and satellite. Hot topics covered include HD Radio, HDTV, 2 GHz broadcast auxiliary services, EAS, workflow, metadata, digital asset management, advanced video and audio compression, audio and video over IP, and Internet broadcasting. A wide range of related topics that engineers and managers need to understand are also covered, including broadcast administration, FCC practices, technical standards, security, safety, disaster planning, facility planning, project management, and engineering management. Basic principles and the latest technologies and issues are all addressed by respected professionals with first-hand experience in the broadcast industry and manufacturing. This edition has been fully revised and updated, with 104 chapters and over 2000 pages. The Engineering Handbook provides the single most comprehensive and accessible resource available for engineers and others working in production, postproduction, networks, local stations, equipment manufacturing or any of the associated areas of radio and television.

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.

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.

A comprehensive guide that addresses the theory and practice of spatial audio This book provides readers with the principles and best practices in spatial audio signal processing. It describes how sound fields and their perceptual attributes are captured and analyzed within the time-frequency domain, how essential representation parameters are coded, and how such signals are efficiently reproduced for practical applications. The book is split into four parts starting with an overview of the fundamentals. It then goes on to explain the reproduction of spatial sound before offering an examination of signal-dependent spatial filtering. The book finishes with coverage of both current and future applications and the direction that spatial audio research is heading in. Parametric Time-frequency Domain Spatial Audio focuses on applications in entertainment audio, including music, home cinema, and gaming—covering the capturing and reproduction of spatial sound as well as its generation, transduction, representation, transmission, and perception. This book will teach readers the tools needed for such processing, and provides an overview to existing research. It also shows recent up-to-date projects and commercial applications built on top of the systems. Provides an in-depth presentation of the principles, past developments, state-of-the-art methods, and future research directions of spatial audio technologies Includes contributions from leading researchers in the field Offers MATLAB codes with selected chapters An advanced book aimed at readers who are capable of digesting mathematical expressions about digital signal processing and sound field analysis, Parametric Time-frequency Domain Spatial Audio is best suited for researchers in academia and in the audio industry.

A comprehensive and mathematically accessible introduction to digital signal processing, covering theory, advanced topics, and applications.

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.

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.

This book is a printed edition of the Special Issue "Sound and Music Computing" that was published in Applied Sciences

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.

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.

This book contains a complete and accurate mathematical treatment of the sounds of music with an emphasis on musical timbre. The book spans the range from tutorial introduction to advanced research and application to speculative assessment of its various techniques. All the contributors use a generalized additive sine wave model for describing musical timbre which gives a conceptual unity, but is of sufficient utility to be adapted to many different tasks.

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.

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 video player.

The unprecedented growth in the range of multimedia services offered these days by modern telecommunication systems has been made possible only because of the advancements in signal processing technologies and algorithms. In the area of telecommunications, application of signal processing allows for new generations of systems to achieve performance close to theoretical limits, while in the area of multimedia, signal processing the underlying technology making possible realization of such applications that not so long ago were considered just a science fiction or were not even dreamed about. We all learnt to adopt those achievements very quickly, but often the research enabling their introduction takes many years and a lot of efforts. This book presents a group of invited contributions, some of which have been based on the papers presented at the International Symposium on DSP for Communication Systems held in Coolangatta on the Gold Coast, Australia, in December 2003. Part 1 of the book deals with applications of signal processing to transform what we hear or see to the form that is most suitable for transmission or storage for a future retrieval. The first three chapters in this part are devoted to processing of speech and other audio signals. The next two chapters consider image coding and compression, while the last chapter of this part describes classification of video sequences in the MPEG domain.

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.