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Digital Coding of Waveforms Waveform Quantization and Coding Recursive Source Coding Waveform Quantization and Coding Image Coding Waveform Coding Using Wavelets Wavelets and Waveform Coding Low-Bit-Rate Predictive Coding of Speech Waveforms Based on Vector Quantization Linear Predictive Coding of Speech Waveforms Frequency Domain Coding of Speech Waveforms at Low Bit Rates Neural Waveform Coding Radar Signal Coding and Waveform Design WAVEFORM Quatization and Coding Waveform Coding of Speech and Voiceband Data Signals Integrated Coding and Waveform Design Study Waveform Coding of Voiceband Analog Modern Signals A Study of Adaptive Schemes for Waveform Coding of Speech Finite-state-vector Quantization for Waveform Coding Adaptive Residual Vector Quantization for Waveform Coding Digital Telephony and Network Integration Speech Coding and Synthesis L'Isle de Cazones Time-domain Waveform Coding by Simulated Annealing Prototype Waveform Interpolation Based Low Bit Rate Speech Coding Digital Speech Processing Pitch Detection Techniques for Prototype Waveform Coding An Investigation Into Glottal Waveform Based Speech Coding Exploitation of Advanced Waveform Coding Techniques to Enhance Stepped-frequency Radar Performance Quality of Compressed Speech Using Trellis Waveform Coding Audio Signal Processing and Coding Digital Transmission Systems Binaural Waveform Coding in the Inferior Colliculus of the Cat Coding of Excitation Signals in a Waveform Interpolation Speech Coder Pitch Synchronous Waveform Interpolation for Very Low Bit Rate Speech Coding Coding Theorems for Filtered-waveform Channels Principles of Speech Coding Vector Quantization and Scalar Linear Prediction for Waveform Coding of Speech at 16 Kb Waveform Design for Active Sensing Systems Handbook of Visual Communications Representing Voiced Speech Using Prototype Waveform Interpolation for Low-rate Speech Coding

Ideal for researchers and practitioners looking to develop and use computational algorithms for waveform design in diverse active sensing applications. "This thesis proposes an energy fixup to smoothen the synthesized speech envelope when the interpolation procedure fails to provide the smooth linear result that is desired. Further investigation, however, leads to the final proposal in this thesis that PWI should he performed on the clean speech signal instead of the excitation to achieve consistently reliable results for all voiced frames." -- Digital Transmission Systems, Third Edition, is a comprehensive overview of the theory and practices of digital transmission systems used in digital communication. This new edition has been completely updated to include the latest technologies and newest techniques in the transmission of digitized information as well as coverage of digital transmission design, implementation and testing. "The necessity of SEW phase information in the WI coder is also investigated in this thesis. Informal subjective test results demonstrate that transmission of SEW magnitude encoded by split/shape-gain VQ and inclusion of a fixed phase spectrum drawn from a voiced segment of a high-pitched male speaker obviates the need to send phase information." -- It is shown that error-correcting codes capable of correcting random/burst errors play a critical role in the design of reliable multiple-access and adaptive-array ECCM systems. A comprehensive investigation of effective block and convolutional coding techniques, as well as combinations of the above, is conducted for worst-case interference models which are applicable to a wide variety of digital communications systems. It is shown that essentially error-free communications are possible, even when significant portions of the received data are completely destroyed by interference, providing a proper code rate and decoding technique are selected. The results in this report may be used to select error-correcting codes to improve the performance of existing systems as well as to optimize the integration of the coding and waveform design in new digital communications network. (Author). What is "digital telephony"? To the authors, the term digital telephony de notes the technology used to provide a completely digital point-to-point voice communication system from end to end. This implies the use of digital technol ogy from one end instrument through the transmission facilities and switching centers to another end instrument. Digital telephony has become possible only because of the recent and ongoing surge of semiconductor developments allowing microminiaturization and high reliability along with reduced costs. This book deals with both the future and the present. Thus, the first chapter is entitled, "A Network in Transition." As baselines, Chapters 2, 3, and 10 provide the reader with the present status of telephone technology in terms of voice digitization as well as switching principles. The book is an outgrowth of the authors' continuing engineering education course, "Digital Telephony," which they have taught since January, 1980, to attendees from business, industry, government, common carriers, and tele phony equipment manufacturers. These attendees come from a wide variety of educational backgrounds. but generally have the equivalent of at least a bachelor's degree in electrical engineering. The book has been written to provide both the engineering student and the practicing engineer a working knowledge of the principles of present and future voice communication systems based upon the use of the public switched network. Problems or discussion questions have been included at the ends of the chapters to facilitate the book's use as a senior level or first year graduate level course text. 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 study of digital speech processing, synthesis and recognition. This second edition contains new sections on the international standardization of robust and flexible speech coding techniques, waveform unit concatenation-based speech synthesis, large vocabulary continuous-speech recognition based on statistical pattern recognition, and more. This volume is the most comprehensive reference work on visual communications to date. An international group of well-known experts in the field provide up-to-date and in-depth contributions on topics such as fundamental theory, international standards for industrial applications, high definition television, optical communications networks, and VLSI design. The book includes information for learning about both the fundamentals of image/video compression as well as more advanced topics in visual communications research. In addition, the Handbook of Visual Communications explores the latest developments in the field, such as model-based image coding, and provides readers with insight into possible future developments. Displays comprehensive coverage from fundamental theory to international standards and VLSI design Includes 518 pages of contributions from well-known experts Presents state-of-the-art knowledge--the most up-to-date and accurate information on various topics in the field Provides an extensive overview of international standards for industrial applications Filtered-waveform channels are continuous-time channels with waveform inputs satisfying a power constraint; the inputs are passed through a linear filter and corrupted by additive stationary Gaussian noise. Both a white-noise channel and a colored-noise channel with a matched filter are investigated. There exist equivalent channels (both of exactly the same form) which transmit coefficients of orthogonal expansions defined by the channel and are specified by its normal values. The direct half of the coding theorem is proved by Feinstein's fundamental lemma, using the asymptotic distributions of normal values and power distributions upon the input random variables. Weak and strong converses are proved. The capacity is also shown to be the supremum of asymptotic time-average mutual information over a class of stationary Gaussian inputs with spectral densities Orthogonal expansions in terms of normal functions in Hilbert spaces isometric to random signal and noise permit expression of mutual information in terms of normal values whose asymptotic distribution is found. The coding theorem is proved by random coding using the measure of the input process. (Author). Acoustic signal compression techniques, converting floating-point waveforms into a bitstream representation, serve a cornerstone in the current data storage and telecommunication infrastructure. Conventional digital signal processing (DSP) methodologies stem from human heuristics, which are with limited performance and highly domain specific. For the past decade, deep neural networks (DNNs) have shown the potential to tackle this problem in a pure end-to-end manner, without relying on human priors or feature-engineering but the data itself. Besides, due to this general-purpose computational paradigm, learning a compact representation of acoustic signals can be integrated to various downstream applications such as speech encryption, recognition and natural language understanding towards future multi-modal intelligent systems. However, the rise of DNNs brings in not only potentials but also concerns, among which model complexity is a major challenge especially for acoustic coding systems. Most codecs are deployed on low power devices, such as mobile phones and hearing aids which do not afford a gigantic neural network in spite of the impressive performance. We propose a research methodology to not simply discard conventional DSP methods by embracing the fancy design of advanced neural networks, but revitalize those lightweight yet effective techniques in the modern computational platform. It is becoming increasingly apparent that all forms of communication-including voice-will be transmitted through packet-switched networks based on the Internet Protocol (IP). Therefore, the design of modern devices that rely on speech interfaces, such as cell phones and PDAs, requires a complete and up-to-date understanding of the basics of speech Hardbound. The fields of speech coding and synthesis have developed rapidly over the last decade. Text-to-text speech systems now produce reasonable quality speech, and currently available speech coders can transmit good quality speech at below 10kb/s. This, in combination with the ever-increasing speed of microprocessors and signal processing hardware, has resulted in a large number of practical applications. These applications in turn have stimulated research, and the number of papers published on speech coding and synthesis have proliferated rapidly. Reflecting periodically on such developments have inspired the publication of this book. Topics such as the effect of cross channel errors on coded speech and the determination of a proper pitch contour for synthesized speech are included.Both readers unfamiliar with the fields of speech coding and speech synthesis as well as those already working within the areas, will find the book of interest. The spreading of digital technology has resulted in a dramatic increase in the demand for data compression (DC) methods. At the same time, the appearance of highly integrated elements has made more and more com plicated algorithms feasible. It is in the fields of speech and image trans mission and the transmission and storage of biological signals (e.g., ECG, Body Surface Mapping) where the demand for DC algorithms is greatest. There is, however, a substantial gap between the theory and the practice of DC: an essentially nonconstructive information theoretical attitude and the attractive mathematics of source coding theory are contrasted with a mixture of ad hoc engineering methods. The classical Shannonian infor mation theory is fundamentally different from the world of practical pro cedures. Theory places great emphasis on block-coding while practice is overwhelmingly dominated by theoretically intractable, mostly differential predictive coding (DPC), algorithms. A dialogue between theory and practice has been hindered by two pro foundly different conceptions of a data source: practice, mostly because of speech compression considerations, favors non stationary models, while the theory deals mostly with stationary ones.