



Signal Processing and Systems Theory

Technical Challenges and Design Issues in Bangla Language Processing

Digital Signal Processing with Examples in MATLAB

Subband adaptive filtering is rapidly becoming one of the most effective techniques for reducing computational complexity and improving the convergence rate of algorithms in adaptive signal processing applications. This book provides an introductory, yet extensive guide on the theory of various subband adaptive filtering techniques. For beginners, the authors discuss the basic principles that underlie the design and implementation of subband adaptive filters. For advanced readers, a comprehensive coverage of recent developments, such as multiband tap-weight adaptation, delayless architectures, and filter-bank design methods for reducing band-edge effects are included. Several analysis techniques and complexity evaluation are also introduced in this book to provide better understanding of subband adaptive filtering. This book bridges the gaps between the mixed-domain natures of subband adaptive filtering techniques and provides enough depth to the material augmented by many MATLAB® functions and examples. Key Features: Acts as a timely introduction for researchers, graduate students and engineers who want to design and deploy subband adaptive filters in their research and applications. Bridges the gaps between two distinct domains: adaptive filter theory and multirate signal processing. Uses a practical approach through MATLAB®-based source programs on the accompanying CD. Includes more than 100 M-files, allowing readers to modify the code for different algorithms and applications and to gain more insight into the theory and concepts of subband adaptive filters. Subband Adaptive Filtering is aimed primarily at practicing engineers, as well as senior undergraduate and graduate students. It will also be of interest to researchers, technical managers, and computer scientists.

Signal processing plays an increasingly central role in the development of modern telecommunication and information processing systems, with a wide range of applications in areas such as multimedia technology, audio-visual signal processing, cellular mobile communication, radar systems and financial data forecasting. The theory and application of signal processing deals with the identification, modelling and utilisation of patterns and structures in a signal process. The observation signals are often distorted, incomplete and noisy and hence, noise reduction and the removal of channel distortion is an important part of a signal processing system. Advanced Digital Signal Processing and Noise Reduction, Third Edition, provides a fully updated and structured presentation of the theory and applications of statistical signal processing and noise reduction methods. Noise is the eternal bane of communications engineers, who are always striving to find new ways to improve the signal-to-noise ratio in communications systems and this resource will help them with this task. \* Features two new chapters on Noise, Distortion and Diversity in Mobile Environments and Noise Reduction Methods for Speech Enhancement over Noisy Mobile Devices. \* Topics discussed include: probability theory, Bayesian estimation and classification, hidden Markov models, adaptive filters, multi-band linear prediction, spectral estimation, and impulsive and transient noise removal. \* Explores practical solutions to interpolation of missing signals, echo cancellation, impulsive and transient noise removal, channel equalisation, HMM-based signal and noise decomposition. This is an invaluable text for senior undergraduates, postgraduates and researchers in the fields of digital signal processing, telecommunications and statistical data analysis. It will also appeal to engineers in telecommunications and audio and signal processing industries.

This textbook offers a fresh approach to digital signal processing (DSP) that combines heuristic reasoning and physical appreciation with sound mathematical methods to illuminate DSP concepts and practices. It uses metaphors, analogies and creative explanations, along with examples and exercises to provide deep and intuitive insights into DSP concepts. Practical DSP requires hybrid systems including both discrete- and continuous-time components. This book follows a holistic approach and presents discrete-time processing as a seamless continuation of continuous-time signals and systems, beginning with a review of continuous-time signals and systems, frequency response, and filtering. The synergistic combination of continuous-time and discrete-time perspectives leads to a deeper appreciation and understanding of DSP concepts and practices. • For upper-level undergraduates • Illustrates concepts with 500 high-quality figures, more than 170 fully worked examples, and hundreds of end-of-chapter problems, more than 150 drill exercises, including complete and detailed solutions • Seamlessly integrates MATLAB throughout the text to enhance learning

Masters Theses in the Pure and Applied Sciences

Model-Based Actions and Sparse Representation

Adaptive Filtering