Multirate and Adaptive Signal Processing

Teaching Scheme:
Lectures: 4 Hrs/ Week

Examination Scheme:
In Semester Assessment:
Phase I : 30
End Semester Examination:
Phase II: 70

Course Objectives:

1. To extend students understanding of DSP concepts further for designing digital systems
2. To understand Multirate concepts to apply for various DSP applications
3. To make student learn the need of adaptive-ness in digital filters
4. To introduce the concept of prediction and associated filters.

Course Outcomes:

The student will use theory of multirate processing for design of basic systems.
The student will show skills for design of adaptive filter for winner filter,
The student will exhibit the knowledge of spectral estimation for random signals.

Unit I: Basics of Random Signals

Review of Correlation: Auto and Cross, Covariance: Auto and Cross, Energy and Power signals, Spectral Density: Energy and Power. Characterization of random signals: review of deterministic signals, random signals, correlation function, power spectra, DT random signals, time averages for DT random process. Lattice structure for FIR filters (Numericals).

Need for Multirate DSP, Decimation by factor D , Interpolation by factor I, Sampling rate conversion by rational factor I/D, Design of practical sampling rate converters, software implementation of sampling rate converters (Decimators and Interpolators), sample rate conversion using poly-phase filter structures, Applications: Efficient D/A conversion in Hi-Fi systems, Multirate narrow band digital filtering. Sub-band coding of speech signals.

Unit III Adaptive filters

Need of adaptive filters, adaptive filters as noise cancellation, configuration of adaptive filters, main components of adaptive filters, Basic Wiener filter theory-Wiener-Hopf Equation, Adaptive Algorithms: LMS basic adaptive algorithm, Implementation of basic LMS algorithm. Recursive least square algorithms (RLS), Adaptive filtering of ocular artifacts from the human EEG, adaptive telephone echo cancellation.

Unit IV Linear prediction and optimum linear filters

Innovation representation of random process, rational power spectra: AR, MA & ARMA process, Forward & backward linear prediction, Solution of the normal equations: The Levinson - Durbin algorithm. Wiener filter for filtering and prediction, FIR Wiener Filter, Orthogonality principle.

Unit V Power Spectrum Estimation

Estimation of Spectra from Finite-Duration observation of signals, Estimation of autocorrelation and power spectrum of random signals: The Periodogram, The use of the DFT in PSE, Non parametric methods for power spectrum estimation- Bartlett window and Welch method.

Unit VI Architecture for Processors and Applications

Overview of Digital Signal Processors, Need of DSP Processor and its advantage over GP processor, Selecting DSPs, Von Neumann architecture, Harward architecture, VLIW Architecture, Architecture of Texas 320Cxxx family, MAC, Pipelining, Addressing modes. Some Applications of DSP: Digital Graphic Equalizer, Digital Frequency Oscillators, Touch tone generation and reception for digital telephones.

Text Books:

1. John G. Proakis, Manolakis, “Digital Signal Processing, Principles, Algorithms and Applications”, Pearson education, Fourth Edition, 2007.
2. E. C. Ifeachor and B. W. Jervis, “Digital Signal Processing- A Practical Approach”, 2nd Edition, Pearson education. 2007.

Reference Books:

1. S. D. Apte, Advanced Digital Signal Processing, Wiley Publications, 2014.
2. Ramesh Babu, Digital Signal Processing, Scitech Publications, Fourth Edition. 2011