AP5152 ADVANCED DIGITAL SIGNAL PROCESSING

DETAILED SYLLABUS

OBJECTIVES:

- The student comprehends mathematical description and modelling of discrete time random signals.
- The student is conversant with important theorems and algorithms.
- The student learns relevant figures of merit such as power, energy, bias and consistency.
- The student is familiar with estimation, prediction and filtering concepts and techniques.

UNIT I DISCRETE RANDOM SIGNAL PROCESSING

Wide sense stationary process – Ergodic process – Mean – Variance - Auto-correlation and Autocorrelation matrix - Properties - Weiner Khitchine relation - Power spectral density – filtering random process, Spectral Factorization Theorem–Finite Data records, Simulation of uniformly distributed/Gaussian distributed white noise – Simulation of Sine wave mixed with Additive White Gaussian Noise.

UNIT II SPECTRUM ESTIMATION

Bias and Consistency of estimators - Non-Parametric methods - Correlation method - Co-variance estimator - Performance analysis of estimators – Unbiased consistent estimators – Periodogram estimator - Barlett spectrum estimation - Welch estimation.

UNIT III LINEAR ESTIMATION AND PREDICTION

Model based approach - AR, MA, ARMA Signal modeling - Parameter estimation using Yule-Walker method - Maximum likelihood criterion - Efficiency of estimator - Least mean squared error criterion – Wiener filter - Discrete Wiener Hoff equations – Mean square error.

UNIT IV ADAPTIVE FILTERS

Recursive estimators - Kalman filter - Linear prediction – Forward prediction and Backward prediction, Prediction error - Whitening filter, Inverse filter - Levinson recursion, Lattice realization, Levinson recursion algorithm for solving Toeplitz system of equations.

UNIT V MULTIRATE DIGITAL SIGNAL PROCESSING

FIR Adaptive filters - Newton's steepest descent method - Adaptive filters based on steepest descent method - Widrow Hoff LMS Adaptive algorithm - Adaptive channel equalization - Adaptive echo canceller - Adaptive noise cancellation - RLS Adaptive filters - Exponentially weighted RLS – Sliding window RLS - Simplified IIR LMS Adaptive filter.

REFERENCES:

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3. P. P. Vaidyanathan, “Multirate Systems and Filter Banks”, Prentice Hall, 1992. a.

4. Simon Haykin, “Adaptive Filter Theory”, Prentice Hall, Englehood Cliffs, NJ1986. a.

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