Course Name: Statistical Digital Signal Processing

Course Objective

- To equip the student with advanced methods of deterministic as well as stochastic signal modeling techniques;
- Understand how speech and image signals are modeled in practical applications like GSM speech codec and other image coding applications;
- To understand and design advanced FIR and IIR filters;
- To understand apply advanced digital signal processing algorithms to real time applications.

Course Description

- Module 1 : - Signal Modeling
    - The Pade approximation method
    - The Prony approximation method
    - Shank's approximation method
    - All-pole modeling using Prony's method
    - Relationship between all-pole and linear prediction
    - The autocorrelation and covariance methods for all-pole modeling
    - Random process modeling
- Module 2 : - The Levinson Recursion
    - The Levinson-Durbin Recursion
    - The lattice filter structure
    - The Shur-Cohn stability test for digital filters
    - The Cholesky decomposition of a Toeplitz matrix
    - Computing the inverse of a Toeplitz matrix
- Module 3 : - Lattice Filters
    - Derivation of the FIR lattice filter structure
    - The problems of forward and backward linear prediction
    - The split lattice filter
    - All-pole lattice filter, all-pass lattice filters and pole-zero lattice filters
    - The use of lattice filters for all-pole signal modeling
    - The forward covariance method, the backward covariance method and Burg's method for sequential estimation of reflection coefficients
    - Modeling stochastic signals with the lattice method
- Module 4 : - Optimum Filters
    - Design of FIR Wiener filters
    - Design of IIR Wiener filters
    - Recursion approaches to signal estimation – the discrete Kalman filter
- Module 5 : - Spectrum Estimation

  - Classical or nonparametric spectrum estimation techniques – the Periodogram, the modified periodogram, Bartlett's method, Welch's method and Blackman-Tukey method
  - The minimum variance method
  - The maximum entropy method (MEM)
  - Parametric model based power spectrum estimation
  - Frequency estimation algorithms for harmonic processes
  - Principal components frequency estimation
- Module 6 : - Adaptive Filtering
  - FIR adaptive filters
  - Adaptive IIR filters
  - The Recursive Least Squares (RLS) algorithm
  - Application of adaptive filters
  - Linear prediction
  - Echo cancellation
  - Channel equalization
  - Interference cancellation
  - Adaptive notch filtering
  - Adaptive control
  - System identification
  - Array processing