Digital Signal Processing Techniques and its Application

A. Course objective:

At the end of course student will be able to

(a) Compute the frequency response of a given signal and system.
(b) Implement DFT using FFT algorithm (radix 2)
(c) Design the FIR and IIR digital filter both in time domain and frequency domain for a given specification
(d) Apply the appropriate signal processing technique to analysis real world digital signal and system
(e) Apply Multirate signal processing for different sampling rate conversion application
(f) Apply Homomorphic convolution and de-convolution technique for speech signal processing
(g) Apply different signal processing techniques for noise reduction.

B. Course Contents

- Introduction
- Review of Discrete-Time Signals and Systems and their properties
- Z-Transform Application to the Analysis of Linear Time Invariant (LTI) Systems
- Frequency Analysis of Signals and Systems
- Frequency Domain Analysis of LTI Systems
- Sampling and Reconstruction of Signals
- The Discrete Fourier Transform (DFT): its Properties and Applications
- Efficient Computation of the DFT: Fast Fourier Transform Algorithms
- Implementation of Discrete-Time Systems
- Design of Digital Filters
- Homomorphic Signal Processing
- Multirate Digital Signal Processing
- Noise and Different noise reduction methods.

C. Recommended Text Books/Reference Books

1. J. G. Proakis, D. G. Manolakis: "*Digital Signal Processing Principles, Algorithms and Application*", Prentice-Hall, 2007.

2. A.V.Openhem, R.W Schafer and J. R. Buck, "*Discrete time signal processing*" Prentice Hall

3. *Noise reduction in speech application*, Edited by Gillian M. Davis, CRC Press