**Digital Signal Analysis and Processing (3-1-2)**

**Evaluation:**

| | Theory | Practical | Total |
|---|---|---|---|
| Sessional | 30 | 20 | 50 |
| Final | 50 | - | 50 |
| Total | 80 | 20 | 100 |

**Course Objectives:**

1. To provide knowledge on digital signal processing techniques.
2. To design and implement IIR and FIR digital filter.

**Course Contents:**

**1. Signals, Systems and Signal Processing (5 hrs)**

1.1Basic elements of Digital Signal Processing
1.2 Energy signal, Power signal
1.2 Need of Digital Signal Processing over Analog Signal Processing
1.4 Sampling continuous signals and spectral properties of sampled signals.

**2. Discrete-time Signals and System (9 hrs)**

2.1Elementary discrete-time signals
2.2 Discrete time Fourier series and properties
2.3 Discrete time Fourier transform and properties
2.4 Discrete time system properties
2.5 Properties of Linear Time-Invariant systems (LTI)
2.6 LTI convolution sum characterized by constant coefficient difference equations
2.7 Stability of LTI systems, Implementation of LTI system.
2.8 Frequency Response of LTI systems

**3. Review of Z-Transform (4 hrs)**

3.1Definition z-transform
3.2 Convergence of Z-transform, Region of convergence
3.3 Properties of Z-Transform (linearity, time shift, multiplication by exponential sequence, differentiation, time reversal, convolution, multiplication)
3.4 Inverse z-transform-by long division, by partial fraction expansion

**4. Discrete Fourier Transform (8 hrs)**

4.1Definition and application
4.2 Frequency response of LTI system
4.3 Forward and Reverse transform
4.4Properties of the Discrete Fourier Transform: linearity and Symmetry, time shift, frequency shift, duality, convolution, multiplication, conjugation & conjugate symmetry
4.5 Basic concept of Fast Fourier Transform (FFT) algorithm

**5. Discrete Filter Structure** **(6 hrs)**

5.1 FIR filter overview
5.2 Structures for FIR Filter (direct, cascade, frequency sampling, lattice)
5.3 IIR Filter overview
5.4 Structure for IIR filter (direct form I & II, cascade, lattice, lattice ladder)
5.5 Quantization of filter coefficients and effects on location of poles, and zeros

**6. FIR Filter Design** **(7 hrs)**

6.1 Gibbs phenomena in FIR filter design
6.2 Filter Design by Window method (rectangular window, hanning window, hamming window)
6.3 Filter design by Kaiser window
6.4 Filter design by frequency sampling
6.5 Filter design using the Remez exchange algorithm

**7. IIR Filter Design** **(6 hrs)**

7.1 Filter design using low pass approximations Butterworth filter.
7.2 Filter design using impulse invariance method
7.3 Filter design using bilinear transformation
7.2 Properties of Chebyshev & Elliptic filters
7.3High pass, Band pass and Notch filters.

**Laboratory**

1. Overview of DSP tools
2. Scaling, dynamic range and noise behavior of a recursive digital filter
3. Response of a non-recursive digital filter, Implementation in Impulse Invariant and Bilinear Transformation.
4. Bandpass filters implemented using cascade second order systems
5. Design of FIR filter
6. Design of IIR filter

**Reference Books:**


1. A.V. Oppenheim, *Discrete-Time Signal Processing,* Prentice Hall, 1990.
2. J.G. Proakis and D.G. Manolakis, *Digital Signal Processing,* Prentice Hall of India.
3. S.K. Mitra, *Digital Signal Processing, A Computer-based Approach,* McGraw Hill.