**UEIXXX: DIGITAL SIGNAL PROCESSING AND APPLICATIONS**

| L | T | P | Cr |
|---|---|---|---|
| **3** | **1** | 2 | **4.5** |

**Course Objective:**To understand the basic concepts and techniques for digital signal processing, familiarization with DSP concepts by studying the design of different digital filters and transform-domain processing.

**Introduction:** Review of Discrete Time Signals and Systems and z-Transforms, Solution of Difference Equations Using One-sided z-Transform, Frequency domain Characteristics of LTI Systems, LTI Systems as Frequency-Selective Filters.

**Discrete Fourier Transform (DFT) and Fast Fourier Transform (FFT):** Discrete Fourier Transform and its Properties, Divide and Conquer Approach, Decimation in Time and Decimation in Frequency FFT Algorithms.

**Digital Filter Structure:** Describing Equation of digital filter, Structures for FIR Systems: Direct Form Structure, Cascade Form Structure, Structure for IIR Systems: Direct Form Structures, Cascade Form Structure, Parallel Form Structure and Lattice Structure.

**Design of Digital Filters:** Causality and its Implications, Difference between analog filters and digital filters, FIR filter design using windows, Design of IIR filters from analog filters using: Approximation of Derivatives, Impulse Invariance and Bilinear Transformation, Frequency transformations.

**Analysis of Finite Word length Effects:** Introduction, The quantization process and errors, Analysis of coefficient quantization effects in FIR filters, A/D noise analysis, Analysis of arithmetic round off errors, Limit cycles in IIR filters,

**Laboratory work:** Convolution and correlation, Solution of difference equations using z- Transform and Fourier tools, FFT and spectrum analysis, design of high pass, low pass, band pass and band stop FIR filter using window method, design of IIR filter using Matched Z Transform (MZT), Bilinear Z Transform (BZT), Pole Zero Placement and Impulse Invariant methods.

**Course Learning Outcomes (CLO):**
After the successful completion of the course the students will be able to:
1. Analyze the signals in time and frequency domain
2. Apply the transformation tools on signals and systems and analyze their significance and applications.
3. design the structures of different types of digital filters
4. design various digital filters and analyze their frequency response
5. Analyse finite word length effects.

***Text Books***

1. *Proakis, J.G. and Manolakis, D.G., Digital Signal Processing, Prentice Hall of India Private Limited (2006).*
2. *Rabiner, C.R. and Gold, B., Theory and Applications of Digital Signal Processing,Prentice Hall of India Private Limited (2000).*


***Reference Books:***

1. *Antonion, A., Digital Filters: Analysis Design and Application, Prentice Hall of India Private Limited (1999).*
2. *Oppenhein, A.V. and Schafer, R.W., Digital Signal Processing, Prentice Hall of India Private Limited (1998).*


**Evaluation Scheme:**

| **S.NO.** | **Evaluation Elements** | **Weightage (%)** |
|---|---|---|
| 1 | MST | 25 |
| 2 | EST | 35 |
| 3 | Sessional (May include Assignments//Quizzes/Lab Evaluations) | 40 |