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| **Institute:** | Institute of Technology |
| **Name of Programme:** | B.Tech. Electronics & Communication Engineering |
| **Course Code:** | 2EC402 |
| **Course Title:** | Digital Signal Processing |
| **Course Type:** | (■ Core/ □ Value Added Course/□ Departmental Elective/ □ Institute Elective/ □ University Elective/ ( □ Open Elective Any other) |
| **Year of Introduction:** | 2023-24 |

**Credit Scheme**

| L | T | Practical component | | | | C |
|---|---|---|---|---|---|---|
| | | LPW | PW | W | S | |
| 2 | - | 2 | - | - | - | 3 |

### Course Learning Outcomes (CLOs):

At the end of the course, students will be able to

1. characterise an LTI system using z-transform through pole-zero analysis BL 4
2. compute DFT using FFT to interpret spectral characteristics of one-dimensional signals. BL 2
3. design IIR and FIR digital filters for the given practical specifications and analyse the finite word-length effect on a designed filter. BL 6
4. realise digital filter structures for IIR and FIR systems BL 3

| Unit No. | Syllabus | Teaching hours |
|---|---|---|
| I | **Discrete Time Fourier Transform (DTFT) and Discrete Fourier Transform (DFT):** Definition and properties/theorems of DTFT and DFT, frequency response, DFT symmetry relations, Circular shifting and circular convolution in DFT, Linear convolution using DFT, Applications of DFT, FFT algorithm | 08 |
| II | **Z Transform and LTI Discrete-time Systems in the Transform Domain:** Definition and properties, ROC, The inverse-z transform, the transfer function, Effect of the pole and zero of rational z-transform on the overall response, simple digital filters (LPF, HPF, BPF, Notch), Linear Phase digital filters, Complementary transfer functions, Inverse Systems, System identification, Comb filter | 08 |
| III | **IIR and FIR Digital Filter Design:** IIR filter design methods, FIR filter design methods, computationally efficient FIR filter design | 05 |
| IV | **Digital Filter Structures:** Basic FIR and IIR filter structures, Polyphase structure, Analysis of finite word-length effect in filter structure design | 03 |
| V | **Signal Processing Applications:** Spectrum Analysis and Estimation, Echo cancellation, Signal Compression | 03 |
| VI | **DSP Processors:** DSP Processor Architecture: modified Harvard, VLIW, multiply & accumulate,, data address generator circular buffering, bit reverse addressing mode, zero nested loop, barrel shifter, dual port memory, special purpose high performance interfaces | 03 |

**Self Study:**
The self-study contents will be declared at the commencement of semester. Around 10% of the questions v be asked from self-study contents.

**Laboratory Work:**
Laboratory work will be based on above syllabus with minimum 10 experiments to be incorporated.

**List of Experiments:**
MATLAB / C / C++/ Python/ Scilab/Octave/DSP Kit Programming based Experiments to realise following DSP Concepts

| Sr. No. | Title of the experiment | Teaching Hours |
|---|---|---|
| 1 | Apply the linear convolution, autocorrelation, and cross-correlation for given signals for different applications. | 02 |
| 2 | Compute Discrete-Time Fourier Transform (DTFT) of given time domain sequence and interpret frequency response. | 02 |
| 3 | Compute Discrete-Time Fourier Transform (DTFT) of given time domain sequence and interpret frequency response. | 02 |
| 4 | Compute Discrete Fourier Transform (DFT) of given time domain sequence and illustrate the concept of frequency resolution. | 02 |
| 5 | Realize Discrete Fourier Transform (DFT) using Fast Fourier Transform (FFT) algorithm and highlight the saving in computations. | 04 |
| 6 | Demonstrate the importance of pole-zero plots in analysis of digital systems. Further, identify magnitude response through pole-zero locations. | 04 |
| 7 | Design simple digital IIR filters for given specifications. | 02 |
| 8 | Design digital FIR filters using window methods. | 04 |
| 9 | Design digital IIR filters and compare Butterworth, Chebyshev-1, Chebyshev-2, and elliptical responses. | 02 |
| 10 | Realise Least-Mean-Square filter for adaptively remove noise in the given signal. | 04 |
| 11 | Illustrate the process to remove echo and reverberation effect from audio signals. | 04 |
| 12 | Realise linear predictive coding algorithm for speech processing. | 04 |
| 13 | Generate Dual Tone Multi-Frequency (DTMF) signal and detect using Goertzel algorithm. | 04 |
| 14 | Show that Discrete Cosine Transform (DCT) can be effectively used for signal compression algorithms. | 04 |
| 15 | Digital Filter Implementation using Digital Signal Processor (DSP) Kit. | 04 |

**Suggested Readings:**

1. Sanjit K. Mitra, Digital Signal Processing, Tata McGraw Hill
2. Oppenheim, Schafer, Discrete-Time Signal Processing, Buck Pearson Education Publication
3. Emmanuel Ifeachor and Barrie Jervis, Digital Signal Processing: A Practical Approach, Pearson Education, India
4. Proakis, Manolakis, Digital Signal Processing: Principles, Algorithm & Application, PHI
5. Vinay K. Ingle, John G. Proakis, Brooks Cole, Digital Signal Processing Using MATLAB, Thomson Learning