DEPARTMENT OF ELECTRICAL ENGINEERING

Marwadi University

Subject Code: 01PEED0108

Subject Name: Digital Signal Processing

MTech. Year – 1 (Semester – 1)

Objective: In the age of information technology most of this technology is based on the theory of digital signal processing (DSP) and implementation of the theory by devices embedded in what are known as *digital signal processors* (DSPs). Hence, the purpose of this course is to provide an understanding of Digital Signal Processing by introducing sampling, quantization, Discrete Fourier Transform and Filter design.

Credits Earned: 4 Credits

Course Outcomes: After completion of this course, student will be able to

- Discriminate discrete signals and systems from continuous ones and understand the process of discretization.
- Formulate engineering problems in terms of DSP tasks.
- Analyze discrete time signals in frequency domain.
- Design digital filters with the concept of z-transform.
- Conceptualize the need of adaptive filters in power electronics applications.
- Interpret key architectural features of Digital Signal Processor.

Pre-requisite of course: Engineering Mathematics (Differential Equation, Complex Analysis, Fourier analysis, etc.)

Teaching and Examination Scheme

Teaching Scheme (Hours)			Credits	Theory Marks			Tutorial/ Practical Marks		Total Marks
Theory	Tutorial	Practical		ESE (E)	Mid Sem (M)	Internal (I)	Viva (V)	Term work (TW)	
3	0	2	4	50	30	20	25	25	150

Contents

Unit	Topics	Contact Hours
1	**Introduction:** Definition of Signals and systems, classification of signals, elements of digital signal processing system, concept of frequency in continuous and discrete time signals, Periodic Sampling, Interpolation, Frequency domain representation of sampling, Reconstructions of band limited signals from its samples.	4
2	**Discrete-Time Signals and Systems (Time and Transformed Domain analysis):** Linear convolution and its properties, Linear Constant Coefficient Difference equations, Frequency domain representation of Discrete-Time Signals & Systems, Representation of sequences by discrete time Fourier Transform, (DTFT), Properties of discrete time Fourier Transform, and correlation of signals, Fourier Transform Theorems, Z-transform & Inverse.	6
3	**Analysis of Linear Time Invariant system:** Analysis of LTI systems in time domain and stability considerations. Frequency response of LTI system, System functions for systems with linear constant-coefficient Difference equations, Freq. response of rational system functions relationship between magnitude & phase, All pass systems, inverse systems, Minimum/Maximum phase systems, systems with linear phase.	4
4	**Structures for Discrete Time Systems:** Block Diagram and signal flow diagram representations of Linear Constant-Coefficient Difference equations, Basic Structures of IIR Systems, lattice and lattice-ladder structures, Transposed forms, Direct and cascade form Structures for FIR Systems, Linear Phase FIR structure, Effects of Co-efficient quantization.	8
5	**Discrete-Fourier Transform & Fast Fourier Transform:** Representation of Periodic sequences: The discrete Fourier Series and its Properties Fourier Transform of Periodic Signals, Sampling the Fourier Transform, The Discrete-Fourier Transform, Properties of DFT, Linear Convolution using DFT. FFT-Efficient Computation of DFT, Goertzel Algorithm, radix2 Decimation-in-Time and Decimation-in-Frequency FFT Algorithms.	6
6	**Filter Design Techniques:** Design of Discrete-Time IIR filters from Continuous-Time filters Approximation by derivatives, Impulse invariance and Bilinear Transformation methods; Design of FIR filters by windowing techniques.	6
7	**Advance DSP Techniques:** Introduction**,** Basic principles of Forward Linear Predictive filter and applications such as system identification, echo cancellation, equalization of channels.	4

8	**Architecture of DSP Processors & applications:** Harward architecture, pipelining, Multiplier-accumulator (MAC) hardware, architectures of fixed and floating point (TMSC6000) DSP processors. Applications.	4
	Total Hours	42

References:

1. Digital Signal Processing: Principles, Algorithm & Application", 4th edition, Proakis, Manolakis, Pearson.
2. Discrete Time Signal Processing: Oppenheim, Schafer, Buck Pearson education publication, 2nd Edition, 2003.
3. Digital Signal Processing fundamentals and Applications, Li Tan, Jean Jiang, Academic Press, 2nd edition,2013.
4. Digital Signal Processing – A computer based Approach, S.K.Mitra, Tata McGraw Hill,3rd edition,2006.
5. Fundamentals of digital Signal Processing –Lonnie C. Ludeman.
6. Digital Signal processing-A Practical Approach, 2nd edition, Emmanuel I. Feacher, and BarrieW..Jervis, Pearson Education.
7. Digital Signal Processing, S. Salivahanan, A. Vallavaraj, C. Gnapriya TMH.
8. Digital Signal Processors, Architecture, programming and applications by B. Venkatramani, M Bhaskar, Mc-Graw Hill.
9. Adaptive Signal Processing, Bernard Widrow, Samuel D. Sterns, Pearson; 1st edition

Suggested Theory distribution:

The suggested theory distribution as per Bloom's taxonomy is as per follows. This distribution serves as guidelines for teachers and students to achieve effective teaching-learning process.

Distribution of Theory for course delivery and evaluation					
Remember	Understand	Apply	Analyze	Evaluate	Create
5%	10%	15%	30%	20%	30%

Suggested List of Experiments:

MATLAB or Scilab scientific package can be used for the below list of experiments.

1. Write a program for Direct form – I, II form realization of the given IIR system function.
2. Write a program to plot pole-zero of a given FIR filter.
3. Write a program to demonstrate the time shifting and frequency shifting property of DTFT.
4. Write a program to perform circular convolution of two sequences using DFT.
5. Write a program to up sample the sinusoidal sequence by an integer factor.
6. Write a program to down sample the sinusoidal sequence by an integer factor
7. Write a program to convert the sampling by non integer factor of a sinusoidal sequence.
8. Create Blackman Harris, Hamming and Gaussian window and plot them in the filter design toolbox. Design an FIR filter with side lobe attenuation of 40 dB using Kaiser Window of 200 points

9. Design low pass Butterworth, high pass elliptical filter, and band pass chebychev-2 filter digital filter with given specification using impulse invariance method.
10. Design a second-order digital bandpass Butterworth filter with the following specifications f_u= 2.6 kHz, f_L = 2.4 kHz , fs = 8000 Hz. Plot the magnitude and phase response.

Recommendation:

- Students are recommended to attended MOOC courses offered from NPTEL, Coursera, and Udmey.
- Research papers from IEEE Transactions on Signal Processing, Speech Signals, Industrial Electronics.