5th cmp

Pokhara University
Faculty of Science and Technology

Course No.: CMM 344 (3 Credits)	Full marks: 100
Course title: Digital Signal Analysis and Processing (3-1-2)	Pass marks: 45
Nature of the course: Theory and Practical	Total Lectures: 45 hrs
Level: Bachelor	Program: BE (CE, SE and IT)

1. Course Description

This course covers theory and methods for digital signal analysis processing including basic principles governing the analysis and design of discrete-time systems as signal processing devices. Major parts of the course will concentrate on signal analysis using Fourier transforms, linear system analysis and filter design. The discrete Fourier transform and its properties as well as the relationship between continuous and discrete time transforms will be studied. The course presents an analysis on how discrete time, linear shift invariant systems can be characterized using linear difference equations and the impulse response and show how tools such as the z-transform and discrete Fourier transform can be used in the design and analysis of such systems. An introduction to Fast Fourier Transform and the design and implementation of digital filters is presented towards the end of the course.

2. General Objectives

- To provide fundamental knowledge of digital signal processing techniques and applications.
- To introduce basic techniques in designing and implementing digital signal processing systems.
- To learn basic methods of spectral analysis.
- To teach students to design digital filters

3. Methods of Instruction

Lecture, Discussion, Readings, Practical works and Project works.

4. Contents in Detail

Specific Objectives	Contents

• Learn about the basics of discrete time signals and its classification • Understand the concept of different types of discrete time systems • Comprehend the concept of discrete linear convolution and its properties	**1. Discrete Signals and systems: (8 hrs)** 1.1. Introduction of signals. Basic Elements of DSP system. application of DSP. Sampling continuous signals and spectral properties of sampled signals. Basic Elementary signals - unit impulse signal. unit step .signal. ramp signal . Sigmoid signal. exponential signal. sinusoidal signal. 1.2. Classification of signals – analog and digital signals. deterministic and random signals. periodic and aperiodic signals, even and odd signals and energy and power signals. 1.3. Transformation of independent variable – Time shifting. Amplitude scaling, Time scaling, Time inversion and Combined Operations 1.4. Discrete time system. Classification- static and dynamic systems, time variant and time invariant systems, linear and nonlinear systems, causal and noncausal systems, stable and unstable systems 1.5. Linear Time Invariant (LTI) System, Convolution summation of discrete systems, response to discrete inputs, Properties of Convolution – Commutative property, Associative property, Distributive property, Causality and Stability of LTI system
• Comprehend the concept of Z-Transform, ROC and properties of Z-transform • Learn to perform forward and inverse Z-transform • Understand the difference between Unilateral and Bilateral Z-transform • Grasp the concept of causality and stability	**2. Z-transform (6 Hrs)** 2.1. Definition, Region of Convergence of Z-transform . Relationship to causality and stability 2.2. Properties of Z-transform (linearity. time shift. multiplication by exponential sequence. differentiation. time reversal, convolution. multiplication. Parseval's theorem) 2.3. Inverse z-transform by long division and partial fraction expansion. by power series expansion.
• Grasp the concept of plotting Magnitude and Phase Response of LTI system • Understand the concept of poles and zeros and their relationship to causality	**3. Analysis of LTI system in frequency domain (5 hrs)** 3.1. Frequency response of LTI system. response to complex exponential 3.2. Linear constant co-efficient difference equation and corresponding system function 3.3. Relationship of frequency response to pole-zero of system 3.4. Linear phase of LTI system and its relationship to causality.

• Learn to represent FIR and IIR filters in direct, cascade and lattice structure. • Understand the effect of Quantization and limit cycles	**4. Discrete filter structures (6 hrs)** 4.1. Structures for FIR filter (direct form, cascade, lattice) 4.2. Structures for IIR filter (direct form I, direct form II, cascade, lattice, lattice ladder) 4.3. Quantization effect (truncation, rounding)
• Grasp the concept of designing analog Butterworth and Chebyshev filters. • Comprehend how analog IIR filter is converted to digital IIR filter • Understand the idea of converting LPF prototype filter to LP, HP, BP and BS filters • Learn the property of Butterworth, Chebyshev and Elliptical filters and their differences.	**5. IIR Filter Design (7 hrs)** 5.1. Introduction to Classical filter design using polynomial approximations - Butterworth, Chebyshev 5.2. Filter design using impulse invariance method and bi-linear transformation 5.3. Properties of Chebyshev & Elliptic filters
• Understand Gibbs phenomena • Grasp the concept of designing FIR filters using Windowing and frequency sampling methods. • Understand the concept of designing optimum FIR filters	**6. FIR Filter Design (7 hrs)** 6.1. Gibbs phenomena in FIR filter design approximations 6.2. Applications of window functions - Rectangular window, Triangular window, Hanning Window, Hamming Window and Kaiser windows 6.3. FIR filter design by the frequency sampling method
• Understand the concept of Discrete Fourier Transform and its properties • Grasp the concept of circular convolution • Learn about FFT, DIT, DIF algorithms. • Comprehend the computational complexity of FFT algorithm	**7. Discrete Fourier Transform (6 hrs)** 7.1. Discrete Fourier transform (DFT) definition and representation, 7.2. Properties of DFT (linearity, time shift, frequency shift, conjugation and conjugate symmetry, duality, convolution, multiplication), circular convolution 7.3. Fast Fourier Transform (FFT) algorithm (decimation in time algorithm, decimation in frequency algorithm) 7.4. Computational complexity of FFT algorithm.

5.List of Tutorials

The following tutorial activities of 15 hours per group of maximum 24 students should be conducted to cover all the required contents of this course.

S.N.	Tutorials
1	Solve the problems related to Section 1.2, 1.4and 1.5
2	Solve the problems related to Z-Transform (Section 2)
3	Solve the problems related to Frequency Response of LTI system (Section 3)
4	Solve the problems related to Discrete Filter Structures (Section 4)
5	Solve the problems related to the design of digital IIR filters using Impulse Invariance and Bilinear Transformation methods
6	Solve the problems related to the design of digital FIR filters using Windowing methods.
7	Solve the problems related to circular convolution, DIT and DIF

6.Practical Works

Laboratory work of 30 hours per group of maximum 24 students should cover implementation of the following topics using simulation software.

SN	List of Practicals
1	Generate and Investigate Basic Discrete Time Signals i. Unit impulse signal, Unit step signal, Ramp signal, Sinusoidal Signal. Exponential signal and square signal ii. Compute even and odd parts of signal iii. Convolution Sum of two sequences
2	Frequency response and pole zero plot of differential equation
3	Compute 4-point and 8-point DFT using FFT and investigate their frequency responses
4	Design an IIR lowpass filter using Impulse Invariance and Bilinear Transformation Method
5	Design a FIR filter using different windows and compare the result.
6	Real-World Digital Signal Analysis and Processing examples and demonstrations

7. Evaluation System and Students' Responsibilities

Evaluation System

The internal evaluation of a student may consist of assignments, attendances, internal assessment, lab reports, project works etc. The internal evaluation scheme for this course is as follows:

Internal Evaluation	Weight	Marks	External Evaluation	Marks
Theory		30		
Attendances & Class Participations	10%			

Assignments	20%			
Presentations/Quizzes	10%			
Internal Assessment	60%			
Practical		20	Semester-End examination	50
Attendances & Class Participations	10%			
Lab Report/Project Report	20%			
Practical Exam/Project Work	40%			
Viva	30%			
Total Internal		50		
Full Marks: 50 + 50 = 100				

Student Responsibilities

Each student must secure at least 45% marks in the internal evaluation with 80% attendance in the class to appear in the Semester End Examination. Failing to obtain such score will be given NOT QUALIFIED (NQ) and the student will not be eligible to appear in the End-Term examinations. Students are advised to attend all the classes and complete all the assignments within the specified time period. If a student does not attend the class(es), it is his/her sole responsibility to cover the topic(s) taught during the period. If a student fails to attend a formal exam, quiz, test, etc. there won't be any provision for a re-exam.

8. Prescribed Books and References

Text Books

1. Alan V. Oppenheim, Ronald W. Schafer, John R. Buck, "Discrete-Time Signal Processing", Pearson Education.

References

1. J.G. Proakis and D.G. Manolakis,"Digital signal Processing", Prentice Hall.
2. S. K. Mitra, "Digital signal Processing, A Computer-based Approach", McGraw Hill