Pokhara University

Faculty of Science and Technology

Course Code: CMM 320(3 Credit)	Full Marks: 100
Course Title: Signal, System and Processing (3-0-2)	Pass Mark: 45
Nature of the Course: Theory and Practice	Total Lectures: 45 hours
Level: Bachelor/ Year: III/ Semester: V	Program: Bachelor of Engineering in Information Technology

1. Course Description:

"Signal System and Processing" is a fundamental aspect of Information Technology Engineering. This course starts from the basic concepts of continuous and discrete-time signals and proceed to learn how to analyze data via the Fourier transform, how to manipulate data via digital filters and how to convert analog signals into digital. The solid theoretical bases are complemented by applied examples in Matlab. Design and lab exercises are also significant components of the course.

2. General Objectives:

a) To provide fundamental knowledge of digital signal processing techniques and applications.

b) To describe signals mathematically and understand how to perform mathematical operations on signals.

c) To learn basic methods of spectral analysis

d) It will provide knowledge and realization of Digital filter.

e) Understand the principle of Linear System and digital signal processing fundamentals.

3. Methods of Instructions:

- lecture, tutorial, discussion and practical.

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. Introduction to discrete time signals and systems:(7 hrs)** 1.1 Discrete time signal, basic signal types 1.2 Energy signal, power signal 1.3 Periodicity of discrete time signal 1.4 Transformation of independent variable 1.5 Discrete time Fourier series and properties 1.6 Discrete time Fourier transform and properties 1.7 Discrete time system properties 1.8 Linear time invariant (LTI) system, convolution sum, properties of LTI system 1.9 Sampling of continuous time signal, spectral properties of sampled signal
• 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 (7 hrs)** 2.1 Definition of Z-transform and Region of Convergence 2.2 Properties of Z-transform (linearity, time shifting, scaling, differentiation, convolution, Parseval's theorem) 2.3 Inverse Z-transform using long division and partial fraction method 2.4 Relation of ROC with causality and stability
• 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 (4 hrs)** 3.1 Frequency response of LTI system, response to complex exponential. 3.2 Linear constant coefficient difference equation and corresponding system function 3.3 Relationship of frequency response to pole – zero of system
• Learn to represent FIR and IIR filters in direct, cascade, parallel and lattice structure. • Realization of hardware implementation of the filter by studying the filter	**4. Discrete filter structures (7 hrs)** 4.1 FIR filter 4.2 Structures of FIR filter (direct, cascade and lattice) 4.3 Structures of IIR filter (direct form I and II, cascade, lattice and lattice ladder)

elements like adder, delay element, multiplier etc.	4.4 Quantization effect (truncation, rounding, limit cycles and scaling)
• 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 (5 hrs)** 5.1 IIR filter design by transformation (Impulse invariance method, Bilinear transformation method) 5.2 Design of digital low pass Butterworth filter 5.3 Properties of (Chebyshev filter, elliptical filter and Bessel filter) 5.4 Spectral transformation
• 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 FIR filter design using windows (rectangular, Hanning and Hamming windows) 6.2 FIR filter design using Kaiser windows 6.3 FIR filter design using Frequency sampling method 6.4 FIR filter design using Remez Exchange Algorithm
• 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. The Discrete Fourier Transform (8 hrs)** 7.1 Definition of discrete fourier transform (DFT) and Inverse DFT (IDFT) 7.2 Computation of DFT and IDFT using mathematical relation and matrix method 7.3 Properties of DFT (linearity, time shifting, frequency shift, conjugation and conjugation symmetry, duality, convolution and multiplication) 7.4 Circular convolution 7.5 Introduction to Fast Fourier Transform (FFT), Computation of 4-point DFT using Decimation in Time and Decimation in Frequency radix-2 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:

SN	Tutorials
1.	Study of basic analog and digital signals and their transformation
2.	Plot of a Continuous and discrete time signals,numerical on their analysis, system properties
3.	Analysis of a LTI system using pole zero diagram and determine the stability of that system.
4.	Solve the problems related to Discrete Filter
5.	Realization of FIR and IIR filter using filter structures
6.	Design of IIR filter and FIR filter using various methods.
7.	Analysis of a digital signal using Discrete Fourier Transform

6. List of Practical (30 hours for a group of maximum 24 students)

SN	Implementation Description
1	Generate and Investigate Basic Continuous and 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, attendance, term-exams, lab reports and projects etc. The tabular presentation of the internal evaluation is as follows:

Internal Evaluation	**Weight**	**Marks**	**External Evaluation**	**Marks**
Theory		**30**	**Semester End**	**50**
Attendance &Class Participation	10%			
Assignments	20%			
Presentations/Quizzes	10%			
Internal Assessment	60%			
Practical		**20**		
Attendance &Class Participation	10%			
Lab Report/Project Report	20%			
Practical Exam/Project Work	40%			
Viva	30%			
Total Internal		**50**		
Full Marks: 50 + 50 = 100				

Students' Responsibilities

Each student must secure at least 45% marks separately in internal assessment and practical evaluation with 80% attendance in the class in order to appear in the Semester End Examination. Failing to get such score will be given NOT QUALIFIED (NQ) to appear the Semester-End Examinations. Students are advised to attend all the classes, formal exam, test, etc. and complete all the assignments within the specified time period. Students are required to complete all the requirements defined for the completion of the course.

8. Prescribed Books and References
Text Books: 1. Alan V. Oppenheim, Ronald W. Schafer "Discrete-Time Signal Processing", Prentice Hall 2. J.G. Proakis and D.G. Manolakis, "Digital Signal Processing" Prentice Hall of India.
References: **1.** S.K. Mitra, "Digital Signal Processing. A Computer based approach", Mc Graw Hill.