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PRESIDENCY UNIVERSITY BENGALURU

 **SET-A**

SCHOOL OF ENGINEERING

**END TERM EXAMINATION – MAY/JUNE 2024**

**Semester:** Semester IV-2022

**Course Code:** ECE3002

**Course Name:**  - Digital Signal Processing

**Program:** B. Tech.

**Date :** June 12, 2024

**Time :** 9:30 AM - 12:30 PM

# Max Marks : 100

**Weightage :** 50%

# Instructions:

1. *Read all questions carefully and answer accordingly.*
2. *Question paper consists of 3 parts.*
3. *Scientific and non-programmable calculator are permitted.*
4. *Do not write any information on the question paper other than Roll Number.*

**PART A**

**ANSWER ANY THREE QUESTIONS (3 Q X 5 M = 15 M)**

1. Digital filters are a fundamental component of DSP, which involves processing digital signals to perform tasks like filtering, noise reduction, modulation, demodulation, and more. List the essential components or building blocks involved in the practical implementation of digital filters?

(CO1) [Knowledge]

1. Filters are extensively used in a variety of applications such as audio processing, image processing, communications systems, and control systems. Identify the type of filter which is described by the following difference equation y(n)+2y(n-1)+3y(n-2)=x(n)+5x(n-1) and also write the Transfer function equation H(z) of the filter.

(CO1) [Knowledge]

1. Nyquist-Shannon Sampling Theorem, is a fundamental concept in signal processing that establishes the conditions for accurately representing a continuous-time signal in its discrete-time form. State and explain the three cases of sampling theorem.

(CO2) [Knowledge]

1. IIR filters are more computationally efficient than FIR filters for achieving similar frequency responses, as they typically require fewer parameters. Explain the design procedure of IIR Butterworth filter with relevant mathematical equations.

(CO3) [Knowledge]

1. Most of the time, the final goal of using a filter is to achieve a kind of frequency selectivity on the spectrum of the input signal. Explain the procedure for designing a Finite Impulse Response (FIR) filter, incorporating relevant mathematical expressions/Formulae.

(CO4) [Knowledge]

**PART B**

**ANSWER ANY TWO QUESTIONS (2 Q X 20 M = 40 M)**

1. A) In the context of designing a filter for an audio processing system, the task is to determine the appropriate order of the filter using Chebyshev approximation based on given specifications. With defined parameters such as passband ripple =3dB at a frequency of 1kHz and stopband ripple

=16dB at a frequency of 2kHz. The objective is to calculate the order of the filter, ensuring optimal filtering performance for the audio application. Additionally, once the order is determined, derive the transfer function H(s) of the filter to facilitate further design and implementation.

* 1. Describe the hardware implementation of the given system H(z)= with minimum number of multipliers.

(CO3) [Comprehension]

1. Consider a scenario where you're designing a digital signal processing system requiring a low-pass filter with specific frequency response characteristics. The filter must have a length of 7 taps and utilize a Hamming window for implementation efficiency. Your objective is to optimize the filter design to minimize multiplier usage, taking into account hardware limitations and operational efficiency.



(CO4) [Comprehension]

1. A) Imagine you're leading the development of a digital filter to enhance sound quality for a high-end audio system. Given an analog transfer function  defining the desired frequency response, your task is to utilize the impulse invariant technique to derive the discrete-time transfer function H(z).
	1. Consider a scenario where implementing a digital filter with linear phase characteristics. Compare the number of multipliers needed between two options: one utilizing a direct form structure and the other linear phase structures; Given the system .(Draw both the structures)

(CO4,CO3) [Comprehension]

**PART C**

**ANSWER ANY THREE QUESTIONS (3 Q X 15 M = 45 M)**

1. The Direct Form structure a straightforward way to represent the difference equation of a filter in a block diagram form, allowing for easy understanding and implementation. Obtain Direct form I and Direct Form II structures for the given difference equation. y(n)+2y(n-1)+3y(n-2)=x(n)+5x(n-1).

(CO3) [Application]

1. To achieve the minimum number of multipliers while implementing a filter, we typically aim to exploit symmetries or redundancies in the filter coefficients. How can we design an efficient implementation for the given filter with transfer function  while minimizing the number of multipliers used?

(CO4) [Application]

1. In Digital Signal Processing (DSP), filters play a crucial role in modifying or extracting specific features from signals.
2. Discuss the differences between analog filter and digital filter.
3. Obtain the Direct Form-I and Direct Form-II realization of the filter with difference equation



(CO3,CO4) [Application]

1. The Fast Fourier Transform (FFT) is an algorithmic approach to compute the Discrete Fourier Transform (DFT) of a sequence of complex or real numbers efficiently. Find the 8-point DFT of the sequence x(n)={1,1,1,1,1,0,0,0} using DIF-FFT algorithm

(CO1,CO2) [Application]