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**Presidency University**

**Bengaluru**

**SCHOOL OF ENGINEERING**

**End -Term Examinations, July 2024**

**Date**: 04/07/2024

**Time**: 09:30am – 12:30pm

**Max Marks**: 100

**Weightage**: 50%

**Semester**: III

**Course Code**: ECE3002

**Course Name**: Digital Signal Processing

**Department:** ECE

**Instructions:**

1. *Read the all questions carefully and answer accordingly.*
2. *Do not write any matter on the question paper other than roll number.*

**PART A**

**Answer any SIX Questions. Each question carries 10 marks. (6Qx 10M= 60M)**

1. The Discrete Fourier Transform (DFT) possesses several key properties that make it a powerful tool in signal processing and analysis. Derive the property of the DFT which states the energy in the time domain is equal to the energy in the frequency domain. (CO: 01 BL: 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. (CO: 01 BL: Knowledge)
2. 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. (CO:02 BL: Knowledge)
3. The Fast Fourier Transform (FFT) is an algorithmic implementation of the Discrete Fourier Transform (DFT) that significantly reduces the computational complexity of computing the DFT. Compute the 4-point DFT using DIT-FFT algorithm of the function x(n)=sin(\frac{2\pi n}{N}) (CO:02 BL: Knowledge)
4. Infinite Impulse Response (IIR) filters are a type of digital filter commonly used in signal processing.  Explain the design procedure of IIR Chebyshev filter with relevant mathematical equations. (CO: 03 BL: Knowledge)
5. 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. (CO: 03 BL: Knowledge)
6. Filters are fundamental components in signal processing used to modify the frequency content of signals. Outline the differences between IIR and FIR filters, considering their respective characteristics and implementations. (CO:04 BL: Knowledge)
7. 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. (CO:04 BL: Knowledge)

**PART B**

**Answer any TWO Questions. Each question carries 20 marks. (2Qx 20M= 40M)**

1. Using linear convolution find y(n)=x(n) \* h(n) for the sequences x(n)=(1,2,-1,2,3,-2,-3,-1,1,1,2,-1) and h(n)=(1,2). Compare the result by solving the problem using (a) overlap-save method and (b) overlap-add method. (CO:01 BL: Application)
2. 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.  
   B) Describe the hardware implementation of the given system

H(z)=\frac{1}{8}+\frac{1}{5}z^{-1}+\frac{1}{5}z^{-2}+\frac{1}{8}z^{-3} with minimum number of multipliers.

(CO: 03 BL: Comprehension)

1. High pass filters are commonly used in various signal processing applications to remove or reduce low-frequency components from a signal while preserving the higher-frequency components. A high pass filter is to be designed with the following desired frequency response.  
   H_{d}(\omega )=\begin{Bmatrix} 0, &\frac{-\pi }{4}< \omega < \frac{\pi }{4} \\ e^{-j2\omega}, & \frac{\pi }{4}< \left | \omega \right | < \pi \end{Bmatrix}  
   Find the the frequency response of the FIR filter designed using a rectangular window defined below.  
   w_{R}(n)=\begin{Bmatrix} 1, &0\leq n\leq 4 \\ 0, & Otherwise \end{Bmatrix} (CO: 04 BL: Application)