Roll No											

PRESIDENCY UNIVERSITY
BENGALURU

SCHOOL OF ENGINEERING

END TERM EXAMINATION - JULY 2024

Semester :IV	**Date :07.08.2024**
Course Code : ECE3002	**Time :9.30am to 12.30pm**
Course Name : DIGITAL SIGNAL PROCESSING	**Max Marks :100**
Program :B Tech	**Weightage :50%**

Instructions:
(i) Read all questions carefully and answer accordingly.
(ii) Question paper consists of 3 parts.
(iii) Scientific and non-programmable calculator are permitted.
(iv) Do not write any information on the question paper other than Roll Number.

PART A			
	ANSWER ANY 4 QUESTIONS	**4Q X 5M=20M**	
1	FIR filters have a wide range of applications in digital signal processing. They are used in various fields such as digital audio, image processing, data transmission, and biomedical devices. In biomedical and wearable devices, FIR filters are essential for processing and analyzing signals. List any 3 advantages and 2 disadvantages of FIR filter.	(CO 4)	[Knowledge]
2	The DFT can process sequences of any size efficiently but is slower than the FFT and requires more memory, because it saves intermediate results while processing. Draw the basic block diagram of DIT-FFT algorithm.	(CO2)	[Knowledge]
3	Circular convolution, also known as cyclic convolution, is a special case of periodic convolution, which is the convolution of two periodic functions that have the same period. Perform circular convolution of $x1(n) = \{1,2,2,1\}$ and $x2(n) = \{1,2,3,1\}$	CO1)	[Knowledge]
4	Digital Signal Processors (DSP) take real-world signals like voice, audio, video, temperature, pressure, or position that have been digitized and then mathematically manipulate them. Draw the block diagram of a basic digital signal processing system.	(CO1)	[Knowledge]
5	An FIR filter designed using Fourier series method will have Gibbs oscillations. Explain Gibbs phenomenon and the ways to mitigate the same.	(CO 4)	[Knowledge]
6	Based on the impulse response length, there are two types of filters, Infinite Impulse Response(IIR) and Finite Impulse Response(FIR) filters. Differentiate IIR filter from FIR filter.	(CO 4)	[Knowledge]

PART B

	ANSWER ANY 5 QUESTIONS		5Q X 10M=50M
7	The discrete Fourier transform converts a finite sequence of equally-spaced samples of a function into a same-length sequence of equally-spaced samples of the discrete-time Fourier transform, which is a complex-valued function of frequency. Compute the 4-pt DFT of the discrete time sequence x(n) using equations. $x(n) = \{0,2,4,6\}$	(CO1)	[Application]
8	The "Fast Fourier Transform" (FFT) is an important measurement method in the science of audio and acoustics measurement. It converts a signal into individual spectral components and thereby provides frequency information about the signal. Compute the 8-pt DFT of the sequence using DIF-FFT algorithm. $x(n) = \{1,2,3,4,4,3,2,1\}$	(CO2)	[Application]
9	For the given analog transfer function H(s), determine H(z) using Impulse Invariance method. Assume T=1sec. $$\mathrm{H}(s) = \frac{2}{(s+1)(s+2)}$$	(CO3)	[Application]
10	Obtain the direct form-I realization for the system described by the difference equation $y(n) = 0.5y(n-1) - 0.25y(n-2) + x(n) + 0.4x(n-1)$	(CO3)	[Application]
11	Find the DFT of the sequence using DIT-FFT algorithm. $$x(n) = \{1,0,1,0,1,0,1,0\}$$	(CO2)	[Application]
12	Determine the direct form-II realization for the following system described by the difference equation $y(n) = -0.1y(n-1) + 0.72y(n-2) + 0.7x(n) - 0.252x(n-2)$.	(CO3)	[Application]
13	One method of designing digital filter is to design an analog filter of same characteristics and convert the same to digital. The resulting digital filter exhibits some kind of distortion due to warping effect. Explain Warping effect and pre-warping in detail with necessary diagrams.	(CO3)	[Knowledge]

	PART C		
	ANSWER ANY 2 QUESTIONS		2Q X 15M=30M
14	Design a Chebyshev Low Pass filter with maximum passband attenuation of 2.5dB at 20 rad/sec and stopband attenuation of 30dB at 50 rad/sec.	(CO3)	[Application]
15	Design an analog Butterworth Low Pass filter that has -2dB passband attenuation at a frequency of 20 rad/sec and at least -10dB stopband attenuation at 30 rad/sec.	(CO3)	[Application]
16	For the given specifications, passband attenuation is 3dB at 1000 rad/sec, stopband attenuation is 15dB at 500 rad/sec, design a Butterworth High Pass Filter.	(CO3)	[Application]