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**PRESIDENCY UNIVERSITY**

**BENGALURU**

**SCHOOL OF ENGINEERING**

**MAKE-UP EXAMINATION – SEP 2023**

**Date**: 30/09/2023

**Time**: 1.00pm-4.00pm

**Max Marks**: 100

**Weightage**: 50%

**Course Code**: ECE 213

**Course Name**: Digital Signal Processing **Programme** : B.Tech (ECE)

 **Instructions:**

1. ***Read Questions carefully and answer accordingly***
2. ***All Questions are compulsory***

**Part A [Memory Recall Questions]**

**Answer ALL the questions. Each question carries 4 marks. 7Qx4=28M**

* + - 1. Only digital signals can be processed using a DSP processor. Digital to analog convertor is used to convert a real time analog signal into digital signal. Draw the block diagram of a digital to analog convertor and mention the nature of the signal after each block. (CO-1)[Knowledge]
			2. DSP has many advantages over analog signal processing. Mention any two advantages and two limitations of a DSP system. Also mention two applications for a DSP system. (CO-1)[Knowledge]
			3. Analog low pass filters can be designed using two techniques – Butterworth and Chebychev. Write two differences between both. Which is better and why? (CO-1)[Knowledge]
			4. BIT reversal is used in FFT algorithm to compute DFT. What do you understand by BIT reversal? Show by an example of 4 point DFT how the numbers in natural orders are converted to BIT reversed numbers. Clearly show the mapping between the two. (CO-1)[Knowledge]
			5. Filters are designed to remove unwanted components from signal. Discuss at-least two real time example of filters and mention two differences between an analog and digital filter. Comment on four types of filters based on pass-band and stop-band frequencies. Draw the transfer function of each. (CO-1)[Knowledge]
			6. Digital filters may be classified as FIR and IIR. Draw the impulse response of both and write any four differences between them. (CO-1)[Knowledge]
			7. Sampling is used to convert an analog signal into digital signal. State Nyquist sampling theorem and mention the three cases regarding sampling frequency and maximum signal frequency. (CO-1)[Knowledge]

**Part B [Thought Provoking Questions]**

**Answer ALL the questions. Each question carries 8 marks 3Qx8=24M**

* + - 1. Filters are designed to remove unwanted components from signal. There are two different kinds of filters i.e. Analog filters and Digital filters. (CO-2)[Comprehension]
	1. Mention the two techniques through which an analog low pass filters can be converted to digital filter. (2M)
	2. For the analog transfer function H(s) = $\frac{2}{(s+1)(s+2)},$ determine H(z) using Impulse Invariance Method. Assume sampling time (T) = 1second. (6M)

9. IIR filters are recursive in nature as the output is dependent on both present input and feedback. IIR filters may be realized by different techniques. (CO-2)[Comprehension]

i) Mention any two techniques through which IIR filters may be realized. (2M)

ii) Obtain the Direct Form-I realization for the system described by the difference equation and also find the impulse response H(z). (6M)

y(n)-0.4x(n-1)= 0.5y(n-1)-0.25y(n-2)+x(n)

10. FIR filters shows the property of linear phase when the impulse response h(n)=h(N-1-n), where N is the length of the filter. Realize the system function using linear phase realization (CO-2)[Comprehension]

H(z) = 0.5 + (1/3)z-1 + z-2 + (1/4)z-3 + z-4+ (1/3)z-5+ (1/2)z-6

**Part C [Problem Solving Questions]**

**Answer ALL the Questions. Each question carries 12 marks 4Qx12=48M**

11. It is desirable to break a long sequence into smaller sequence to find the convolution. Mention the two techniques of signal segmentation for long sequences. Given the sequence x(n)={2,1,-1,-2,-3,5,6,-1,2,0,2,1} and impulse response of h(n) = {3,2,1}, compute the output y(n) using overlap-add method. (CO-3)[Application]

12. Bob wants to design an analog low pass Chebychev filter to filter noise from music signal. The filter should have a pass-band attenuation of 3dB at frequency (fp) of 1 KHz and at-least 16dB stop-band attenuation at frequency (fs) of 2KHz. Design the filter for the given specification and determine H(z) (CO-3)[Application]

13. Rohan wants to design an analog low pass Butterworth filter for separating speech from music signal for his application. The filter should have a pass-band attenuation of 2dB at frequency of 20 rad/sec and at-least 10dB stop-band attenuation at 30 rad/sec. Design the filter for the given specification. (CO-2)[Comprehension]

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14. A low pass filter to be designed with the following desired frequency response.

$$H\_{d}\left(e^{jw}\right)=H\_{d}\left(w\right)= \begin{matrix}e^{-j2w},&when \left|w\right|<pi/4\\0,&when, \frac{pi}{4}<\left|w\right|<pi\end{matrix}$$

Determine the filter coefficients hd(n) and h(n) if w(n) is a rectangular window defined as follows,

$$w\_{R}(n)=\begin{matrix}1,&when 0\leq n\leq 4\\0,&otherwise\end{matrix}$$

Also find the frequency response, H(w) of the resulting FIR filter. (CO-3)[Application]