

[Total No. of Questions - 9] [Total No. of Printed Pages - 2]

Dec.-22-0270

EC-604 Digital Signal Processing (ECE)

B.Tech. 6th (CBCS)

Time : 3 Hours

Max. Marks : 60

The candidates shall limit their answers precisely within the answer-book (40 pages) issued to them and no supplementary/continuation sheet will be issued.

Note : Attempt five questions in all, selecting one question each from section A, B, C and D. Section E is compulsory.

SECTION - A

1. Show that a system is casual if and only if its impulse response satisfies the condition $h(n) = 0$ for $n \leq 0$. A signal $x(t) = \cos(4\pi t)$ is sampled at 8 Hz. Find The Signal which if sampled at 8 Hz results in same discrete time signals as obtained from $x(t)$. What is The Nyquist rate for signal $x(t) = \sin(2\pi t)\cos(2\pi t)$. (10)
2. What are the basic elements of a DSP system? Describe the role of each. What are the advantages/disadvantages of DSP over analog signal processing? (10)

SECTION - B

3. Plot $3h(2n-1)+h(n)$ for given $h(n)=[1,0,3,5,4]$. Check the following systems for linearity, causality and time invariance properties: (i) $y(n) = nx(n)$ and (ii) $y(n) = x(-n)$ (10)
4. Compute the convolution of $x(n) = \alpha^n u(n)$ for $-3 \leq n \leq 5$, zero otherwise and $h(n) = 1$ for $0 \leq n \leq 4$, zero otherwise. (10)

SECTION - C

5. Discuss Radix 2 Decimation in Time (DIT) FFT algorithm. Draw the basic butterfly computation diagram in DIT-FFT algorithms. Show how the computational complexity is reduced in FFT algorithm as compared to direct computation of DFT? (10)

2

EC-604

6. Compute the 8 point DFT of sequence $x(n) = \{0, 1, 2, 3, 4, 5\}$ using Radix 2 Decimation in frequency FFT algorithm. (10)

SECTION - D

7. Discuss the procedure to design a digital Butterworth low pass filter if the 3 dB cut off cutoff frequency ω_c and filter order N are specified. If you wish to design a digital Butterworth low pass filter with cut off frequency of 40 Hz, filter order 4 and sampling frequency 200 Hz, find the transfer function $H(s)$ of corresponding analog filter. Assume bilinear transformation for mapping for s plane to z plane. (10)
8. Draw the frequency response of Ideal Low Pass, High Pass, Band Pass and Band Stop digital filters. Design a Linear phase FIR filter approximating the ideal response

$$H_d(\omega) = \begin{cases} 1. & \text{for } |\omega| \leq \frac{\pi}{6} \\ 0. & \text{for } \frac{\pi}{6} < |\omega| \leq \pi \end{cases}$$

Determine the coefficients of a 11- tap filter based on window method with rectangular window. Determine and plot the Magnitude and Phase response of the filter. (10)

SECTION - E

9. (i) Show that the operation of folding and shifting is not commutative. If a signal $x(t)$ is scaled by a factor of 2, then shifted to left by 1 and then folded, what will the resultant signal.
 (ii) A signal $x(n)$ is obtained from $x(t)$ with 20 kHz sampling rate. How can you change the sampling rate of $x(n)$ to 15 kHz?
 (iii) Discuss the mapping analog frequencies into digital frequencies using bilinear transformation.
 (iv) Find the circular convolution of $x(n) = \{1, 1, 1, 1\}$ and $h(n) = \{1, 0, 0, 1\}$. (4*5=20)