USN

18EC52

Fifth Semester B.E./B.Tech. Degree Examination, Dec.2025/Jan.2026

Digital Signal Processing

Time: 3 hrs. Max. Marks: 100

Note: Answer any FIVE full questions, choosing ONE full question from each module.

Important Note : 1. On completing your answers, compulsorily draw diagonal cross lines on the remaining blank pages.
2. Any revealing of identification, appeal to evaluator and /or equations written eg, 42+8 = 50, will be treated as malpractice.

Module-1

1 a. If x(n) = [2, 4, 6, 8], find DFT X(k) and also sketch its magnitude and phase response. **(08 Marks)**

b. Let X(k) be a DFT of a real sequence x(n). For DFT pair shown, compute the values of the boxed quantities using appropriate properties.

$$\{\boxed{x(0)}, 1, 2, \boxed{x(3)}\ 3, 3\} \xleftrightarrow{DFT} \{12, \boxed{X(1)}, -1.5+j0.866, 0, \boxed{X(4)}, -1.5-j2.598\}$$

(06 Marks)

c. State and prove the Parseval's theorem and circular frequency shift property of DFT. **(06 Marks)**

OR

2 a. Given sequence $x_1[n] = [1, 2, 3, 1]$ and $x_2[n] = [4, 3, 2, 2]$, find $x_3(n)$ such that $X_3(k) = X_1(k)\cdot X_2(k)$ using DFT and IDFT method. **(10 Marks)**

b. Prove that the sampling of DTFT of a sequence x(n) result in N point DFT. **(06 Marks)**

c. The DFT of the sequence x(n) is given by $X(k) = [20, -4+4j, -4, -4-4j]$. Compute the sequence $x((n-2))_N$ by applying the appropriate property of DFT. **(04 Marks)**

Module-2

3 a. Compute DFT of the sequence $x_1(n) = \{1, 2, -1, 2, 4, 2, -1, 2\}$ using radix-2 decimation in frequency FFT algorithm. Keep the track of all intermediate results and show them on butterfly diagram. **(10 Marks)**

b. Compute the response of a digital audio system which has long input data sequence x(n) = [1, 2, –1, 2, 3, –2, –3, –1, 1, 1, 2, –1] and h(n) = [1, 2] using overlap add method. Use 5-point circular convolution. **(10 Marks)**

OR

4 a. Develop radix – 2, DIT-FFT algorithm and write signal flow graph for N = 8. **(12 Marks)**

b. Given X(k) = [5, 0, 1 – j, 0, 1, 0, 1 +j, 0]. Compute x(n) using decimation in frequency FFT algorithm. Keep track of all intermediate results and show them on butterfly diagram. **(08 Marks)**

Module-3

5 a. Realize the FIR filter described by the system function $H(z) = 1+\frac{13}{24}z^{-1}+\frac{5}{8}z^{-2}+\frac{1}{3}z^{-3}$ in direct form. **(04 Marks)**

b. Consider a three-stage FIR lattice structure having the coefficients $k_1 = 0.65$, $k_2 = -0.34$ and $k_3 = 0.8$. Realize this filter in direct form. **(08 Marks)**

c. A low pass FIR filter is to be designed with the following desired frequency response:

$$H_d(w)\begin{cases} e^{-j2w} & \text{for } |w| \le \pi/4 \\ 0 & \text{for } \pi/4 < |w| \le \pi \end{cases}$$

Design the filter using rectangular window. **(08 Marks)**

OR

6 a. Consider an FIR filter with system function

$$H(z) = 1 + 2.88z^{-1} + 3.408z^{-2} + 1.74z^{-3} + 0.4z^{-4}$$

Sketch the direct form and lattice realization of the filter. **(10 Marks)**

b. Design a High pass FIR filter having the desired frequency response using Hanning window.

$$H_d(w) = \begin{cases} e^{-j3w} & \text{for } \frac{3\pi}{4} \le |w| \le \pi \\ 0 & \text{for } |w| \le \frac{3\pi}{4} \end{cases}$$

(10 Marks)

Module-4

7 a. Given an analog filter whose transfer function is

$$H(s) = \frac{10}{s+10}$$

Convert it to the digital filter transfer function and difference equation when the sampling period is given as T = 0.01 second. **(06 Marks)**

b. Consider the normalized low pass filter with a cut off frequency of 1 rad/sec.

$$H_p(s) = \frac{1}{s+1}$$

Use $H_p(s)$ and the BLT to obtain (i) IIR digital high pass filter with a cutoff frequency of 30 Hz, assuming a sampling rate of 200 Hz. (ii) IIR digital band stop filter with a lower cutoff frequency of 20 Hz, an upper cutoff frequency of 40 Hz and a sampling rate of 120 Hz. **(06 Marks)**

c. Using BLT design a second order digital lowpass Butterworth filter with a cutoff frequency of 3.4 KHz at a sampling frequency of 8000 Hz. **(08 Marks)**

OR

8 a. Given a filter transfer function

$$H(z) = \frac{0.3430z^2 + 0.6859z + 0.3430}{z^2 + 0.7075z + 0.7313}$$

(i) Realize the digital filter using direct-form I and direct-form II.
(ii) Determine the difference equations for each implementation. **(10 Marks)**

b. Design a second-order digital bandbass Butterworth filter with the following specifications:

(i) Upper cutoff frequency of 2.6 KHz

(ii) Lower cutoff frequency of 2.4 KHz

(iii) Sampling frequency of 8000 Hz. **(10 Marks)**

Module-5

9 a. With a neat diagram explain the special hardware units of digital signal processor. **(10 Marks)**

b. Add the following floating-point number whose formats are defined in Fig.Q9(b) and determine the sum in decimal format.

0001000000010011 + 0100001000000101

-2^3	2^2	2^1	2^0	-2^0	2^{-1}	2^{-2}	2^{-3}	2^{-4}	2^{-5}	2^{-6}	2^{-7}	2^{-8}	2^{-9}	2^{-10}	2^{-11}
⟵ 4 bit exponent ⟶				⟵ 12 bit mantissa ⟶											

Fig.Q9(b) **(10 Marks)**

OR

10 a. With a neat diagram explain the basic architecture of TMS320C54X family processor. **(10 Marks)**

b. Convert the following number in IEEE single precision format to the decimal format.

010100100.101…..0000 **(05 Marks)**

c. Given the FIR filter

$$y(n) = -0.36x(n) + 1.6x(n-1) + 0.36x(n-2)$$

With a passband gain of 2 and the input being half of the range, develop the DSP implementation equations in the Q-15 fixed point system. **(05 Marks)**

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