

Reg. No. :

| | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|
| E | N | G | G | T | R | E | E | . | C | O | M |
|---|---|---|---|---|---|---|---|---|---|---|---|

Question Paper Code : 20932

B.E./B.Tech. DEGREE EXAMINATIONS, NOVEMBER/DECEMBER 2023.

Fourth/Fifth Semester

Computer and Communication Engineering

EC 3492 – DIGITAL SIGNAL PROCESSING

For More Visit our Website
EnggTree.com

(Common to : Electronics and Communication Engineering/ Electronics and Telecommunication Engineering / and Medical Electronics)

(Regulations 2021)

Time : Three hours

Maximum : 100 marks

Answer ALL questions.

PART A — (10 × 2 = 20 marks)

1. What is zero padding? What are its uses?
2. Determine the unit step response of the LTI system with impulse response, $h(n) = \alpha^n u(n), |\alpha| < 1$.
3. Give the expression for location of poles of a Chebyshev type I filter.
4. Obtain the direct form – I realization for the system,
 $y(n) = 0.5y(n-1) - 0.25y(n-2) + x(n) + 0.4x(n-1)$.
5. Define Gibbs phenomenon.
6. Obtain cascade realization with minimum number of multipliers.
$$H(z) = \frac{1}{2} + \frac{1}{4}z^{-1} + \frac{1}{4}z^{-2} + \frac{1}{2}z^{-3}$$
7. Compare fixed point and floating point arithmetic.
8. Draw the quantization noise model for a first order system.
9. Define sampling rate conversion.
10. What is the need for anti-aliasing and anti-imaging filters in down-sampling and up-sampling of a signal respectively?

PART B — (5 × 13 = 65 marks)

11. (a) Determine and sketch the magnitude and phase response of $y(n) + \frac{1}{2}[x(n) + x(n-2)]$

Or

- (b) Determine $X(k)$, for $N = 8$, using DIT-FFT algorithm for the given function below :

$$x(n) = 2^n$$

12. (a) Design an analog Butterworth filter that has $\alpha_p = 0.5 \text{ dB}$, $\alpha_s = 22 \text{ dB}$, $f_p = 10 \text{ kHz}$ and $f_s = 25 \text{ kHz}$.

Or

- (b) Using the bilinear transform design a high-pass filter, monotonic in pass-band with cut-off frequency of 1000 Hz and down 10 dB at 350 Hz. The sampling frequency is 5000 Hz.

13. (a) Design an ideal low-pass filter with a frequency response

$$H_d(e^{j\omega}) = 1 \text{ for } -\frac{\pi}{2} \leq \omega \leq \frac{\pi}{2}$$

$$H_d(e^{j\omega}) = 0 \text{ for } \frac{\pi}{2} \leq |\omega| \leq \pi$$

Find the values of $h(n)$ for $N = 11$.

Or

- (b) Design a high pass filter using hamming window with a cut-off frequency of 1.2 radians/sec and $N = 9$.
14. (a) Find the steady state variance of the noise in the output due to quantization of input for the first order filter.

$$y(n) = ay(n-1) + x(n)$$

Or

- (b) Consider the following second order IIR filter $H(z)$, find the effect on quantization on pole locations of the given system function in direct form and in cascade form. Take $b = 3$ bits.

$$H(z) = \frac{1}{(1 - 0.5z^{-1})(1 - 0.45z^{-1})}$$

15. (a) Discuss the poly-phase structure of interpolator and decimator.

Or

- (b) Describe the features of adaptive filters and any two applications of adaptive filters.

PART C — (1 × 15 = 15 marks)

16. (a) Compute the linear convolution for the following sequence using overlap-save method.

$$x(n) = \{1, 2, -1, 2, 3, -2, -3, -1, 1, 2, -1\}$$

$$h(n) = \{1, 2\}$$

Or

- (b) Using impulse invariance with $T = 1S$, determine $H(z)$ if

$$H(s) = \frac{1}{s^2 + \sqrt{2}s + 1}$$

