

B. TECH.
(SEM V) THEORY EXAMINATION 2022-23
DIGITAL SIGNAL PROCESSING

Time: 3 Hours**Total Marks: 100****Note:** Attempt all Sections. If require any missing data; then choose suitably.**SECTION A****1. Attempt all questions in brief. 2 x 10 = 20**

- (a) Explain the basic elements required for realization of digital system.
- (b) Differentiate between recursive and non-recursive systems.
- (c) Calculate the DFT of the sequence $x(n) = \{1, 2, 1, 3\}$.
- (d) What is Twiddle factor? Write its properties.
- (e) What is the difference between circular convolution and linear convolution?
- (f) What is Frequency Warping?
- (g) Demonstrate the term Gibb's Phenomenon with schematic diagram.
- (h) Write the expression for Hanning window.
- (i) Explain the term Decimation with suitable example.
- (j) Find the output of the sequence $[1 \ 2 \ 3]$ after up sampling by a factor $N=3$.

SECTION B**2. Attempt any three of the following: 10 x 3 = 30**

- (a) Determine DF – I & DF – II realization for a following IIR transfer function
 $H(z) = (0.28z^2 + 0.319z + 0.04)/(0.5z^3 + 0.3z^2 + 0.17z - 0.2)$
- (b) Explain Impulse response invariance method of IIR digital filter design. Also explain mapping of poles from analog domain to digital domain.
- (c) Explain finite word length effect in digital filters. Also explain (i) Coefficient quantization error (ii) Quantization noise – truncation and rounding.
- (d) Derive and draw the flow graph for DIT FFT algorithm for $N=8$.
- (e) Discuss QMF and sub-band coding of speech signals in detail.

SECTION C**3. Attempt any one part of the following: 10 x 1 = 10**

- (a) Obtain direct form and cascade form realization for the transfer function of a FIR system given by-

$$H(z) = \left(1 - \frac{1}{4}z^{-1} + \frac{3}{8}z^{-2}\right) \left(1 - \frac{1}{8}z^{-1} - \frac{1}{2}z^{-2}\right)$$

- (b) (i) Explain the technologies used for DSP in detail.
(ii) Compare IIR and FIR digital filters.

4. Attempt any **one** part of the following: 10 x 1 = 10

- (a) Using bilinear transformation, design a Butterworth filter which satisfies the following conditions:

$$0.8 \leq |H(e^{j\omega})| \leq 1, \quad 0 \leq \omega \leq 0.2\pi$$
$$|H(e^{j\omega})| \leq 0.2, \quad 0.6\pi \leq \omega \leq \pi$$

- (b) Obtain system function of digital filter which is resonant at $\omega_r = \frac{\pi}{2}$, using Bilinear Transformation from the system function of analog filter given as-

$$H(s) = \frac{s + 0.1}{(s + 0.1)^2 + 16}$$

5. Attempt any **one** part of the following: 10 x 1 = 10

- (a) Design a symmetric FIR low pass digital filter whose desired frequency response is as-

$$H_d(\omega) = \begin{cases} e^{-j\omega\tau}, & \text{for } -1 \leq \omega \leq 1 \\ 0, & \text{otherwise} \end{cases}$$

The length of the filter is 7 and $\omega_c = 1$ radian/sample. Use rectangular window function.

- (b) Explain the concept of the Limit Cycle Oscillations & dead band effect with suitable example.

6. Attempt any **one** part of the following: 10 x 1 = 10

- (a) Determine the DFT of the sequence $x(n) = \{1, 1, 2, 2, 3, 3\}$ and determine the corresponding amplitude and phase spectrum.
- (b) Find the DFT of the following discrete time sequence using DIF FFT algorithm $x(n) = \{1, -1, -1, -1, 1, 1, 1, -1\}$

7. Attempt any **one** part of the following: 10 x 1 = 10

- (a) Calculate the circular convolution using graphical method for $x(n) = [1, 2, 3, 4]$ and $h(n) = [4, 3, 2, 1]$.
- (b) Explain the process of multirate signal processing in detail. Also enlist the advantages of multirate signal processing.