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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:

$$0.8 \le \left|H\left(e^{j\omega}\right)\right| \le 1 \ , \qquad 0 \le \omega \le 0.2\pi$$
$$\left|H\left(e^{j\omega}\right)\right| \le 0.2 \qquad , \qquad 0.6\pi \le \omega \le \pi$$

(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-

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

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.