15EC52

ifth Semester B.E. Degree Examination, July/August 2022 **Digital Signal Processing**

Max. Marks: 80

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

Module-1

- 1 Consider the signal $x(n) = a^n u(n)$, 0 < a < 1. The spectra of this signal is sampled at frequencies $W_K = \frac{2\pi K}{N}$, $K = 0, 1, \dots, N-1$. Determine the reconstructed spectra for (08 Marks) a = 0.8 when N = 5.
 - b. Compute the 8-point DFT of $x(n) = (-1)^{n+1}$, $0 \le n \le 7$.

(08 Marks)

- Establish the relationship between (i) DFT and DFS (ii) DFT and DTFT 2
 - Define DFT and IDFT. Compute IDFT of the sequence $X(K) = \{2, 1+j, 0, 1-j\}$. (11 Marks)

Module-2

- State and prove the following DFT properties: 3
 - (i) Time reversal of a sequence
- (ii) Circular frequency shift

(08 Marks)

- b. The five samples of 8-point DFT X(K) are given as follows:
 - X(0) = 0.25, X(1) = 0.125 j0.3018, X(6) = X(4) = 0, X(5) = 0.125 j0.0518
 - Determine the remaining samples if sequence x(n) is real valued sequence.

(08 Marks)

- Find the output y(n) of a filter whose impulse response is $h(n) = \{1, 1, 1\}$ and the input signal to the filter is $x(n) = \{3, -1, 0, 1, 3, 2, 0, 1, 2, 1\}$ using overlap save method.

(08 Marks)

What are FFT algorithms? State their advantages over the direct computation of DFT.

Compute the 8-point circular convolution of $x_1(n) = \left(\frac{1}{4}\right)^n$, $0 \le n \le 7$ and $x_2(n) = \cos\frac{3\pi}{8}n$, (04 Marks) $0 \le n \le 7$.

Module-3

- Derive the signal flow graph for N = 8 point Radix 2 DIF-FFT algorithm. (08 Marks)
 - b. Use the 8-point Radix-2 DIT-FFT algorithm to find the DFT of sequence:

 $x(n) = \{0.707, 1, 0.707, 0, -0.707, -1, -0.707, 0\}$

(08 Marks)

OR

- What is Goertzel algorithm? Obtain the Direct form-II realization of it. (08 Marks)
 - b. For $X(K) = \{0, 0, -j4, 2-j2, 0, 2+j2, 0, 2-j2\}$, find sequence x(n) using DIF-FFT (08 Marks) algorithm.

Module-4

- 7 a. Design a Chebyshev filter to meet the following specifications:
 - (i) Passband ripple $\leq 2 \text{ dB}$
 - (ii) Stopband attenuation ≥ 20 dB
 - (iii) Passband edge: 1 rad/sec
 - (iv) Stopband edge: 1.3 rad/sec

(10 Marks)

b. The system function of low pass digital filter is given by $H(z) = 0.5 \left(\frac{1+z^{-1}}{2-z^{-1}}\right)$. From the above equation find y(n).

OR

- 8 a. Derive an expression for order and cutoff frequency of the Butterworth filter. (06 Marks)
 - b. The system function of the analog filter is given as $H_a(s) = \frac{s+0.1}{(s+0.1)^2+16}$. Obtain the system function of the digital filter using Bilinear transformation which is resonant at $\omega_r = \frac{\pi}{2}$.

Module-5

9 a. Determine the filter coefficients $h_d(n)$ for the desired frequency response of the low pass filter given by

$$H_{d}(e^{j\omega}) = \begin{cases} e^{-j2\omega} & \text{for } -\frac{\pi}{4} \le \omega \le \frac{\pi}{4} \\ 0 & \text{for } \frac{\pi}{4} \le |\omega| \le \pi \end{cases}$$

If we define new filter coefficient by $h(n) = h_d(n).\omega(n)$,

where
$$\omega(n) = \begin{cases} 1 & \text{for } 0 \le n \le 4 \\ 0 & \text{elsewhere} \end{cases}$$

Determine h(n) and also the frequency response $H(e^{j\omega})$ and compare with $H_d(e^{j\omega})$. (08 Marks) b. Explain the frequency sampling method of designing linear phase FIR filters. (08 Marks)

OF

- 10 a. The coefficients of three stages FIR lattice structure is $K_1 = 0.1$, $K_2 = 0.2$ and $K_3 = 0.3$. Find the coefficients of direct form I FIR filter and draw its block diagram. (08 Marks)
 - b. Write short notes on:
 - (i) Hamming window
 - (ii) Hanning window
 - (iii) Bartlett window

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(08 Marks)