## Sixth Semester B.E. Degree Examination, Feb./Mar. 2022 **Digital Signal Processing**

Time: 3 hrs.

Max. Marks: 100

Note: 1. Answer any FIVE full questions, selecting atleast TWO questions from each part. 2. Assume any missing data suitably.

## PART - A

- Find the N-point DFT of the sequence  $x(n) = a^n$  for 0 < a < a1 (04 Marks)
  - b. Compute the 8-point DFT of the sequence x(n) given below. x(n) = (1, 1, 1, 1, 0, 0, 0, 0)

(10 Marks)

- c. Given x(n) = 1, -1, -2, 3, -1) and  $x_2(n) = (1, 2, 3)$ , find the circular convolution of  $x_1(n)$ and  $x_2(n)$  using matrix method. (06 Marks)
- G(k) and H(k) are 6-point DFTS of sequences g(n) and h(n) respectively. The DFT G(k) = (1 + j, -2.1 + j3.2, -1.2 - j2.4, 0, 0.9 + j3.1, -0.3 + j1.1). The sequences g(n) and h(n) are related by circular time shift as h(n) = g((n-4))6. Determine H(k) without computing the D.F.T.
  - b. A long sequence x(n) is filtered through a filter with impulse response h(n) to yield the output y(n). If x(n) = (1, 1, 1, 1, 1, 3, 1, 1, 4, 2, 1, 1, 3, 1), <math>h(n) = (1, -1). Compute y(n) using overlap save method. Use only 5-point circular convolution in your approach.

(12 Marks)

- Determine 8-point D.F.T for a continuous time signal,  $x(t) = \sin 2\pi$  ft with f = 50Hz. Using D.I.F – FFT algorithm. (12 Marks)
  - b. Find the 4-point circular convolution of the following two sequences, using D.I.T-FFT algorithm  $x_1(n) = (2, 1, 1, 2)$  and  $x_2(n) = (1, -1, -1, 1)$ . (08 Marks)
- a. Find the 8-point DFT of x(n) = (1, 1, 1, 1, 0, 0, 0, 0) using DIT-FFT algorithm. (12 Marks) (08 Marks)
  - Develop a radix -3 DIT FFT algorithm for evaluating the DFT for N = 9.

## PART - B

Design an analog low-pass Butterworth filter that will have -1dB cutoff frequency at 75Hz and have greater than 20dB attenuation for all frequencies greater than 150Hz.

(10 Marks)

- Design a Chebyshev filter to meet the following specifications.
  - i) Pass band ripple :  $\leq 2dB$
  - Pass band edge frequency: 1 rad/sec ii)
  - iii) Stop band attenuation : ≥ 20dB
  - Stop band edge frequency: 1.3 rad/sec

(10 Marks)

6 a. A third order Butterworth filter has the transfer function  $H(s) = \frac{1}{(s+1)(s^2+s+1)}$ , obtain

H(z) using impulse – invariant transformation.

(10 Marks)

b. A digital low pass filter is required to meet the following specifications

 $20\log |H(w)| = 0.2\pi \ge -1.9328dB$ 

 $20\log |H(w)| w = 0.6\pi \le -13.9794dB$ 

The filter must have a maximally flat frequency response. Find H(z) to meet the above specifications using impulse invariant transformation. (10 Marks)

7 a. The frequency response of an FIR filter is given by

H(w) = 
$$e^{-j3w}$$
 (1+1.8cos3w+1.2cos2w+ $\frac{1}{2}$ cosw)

Determine the coefficient of the impulse response h(n) of the FIR filter.

(10 Marks)

b. A filter is to be designed with the following desired frequency response

$$H_{d}(w) = \begin{cases} 0 & -\frac{\pi}{4} < w < \frac{\pi}{4} \\ e^{-j2w} & \frac{\pi}{4} < |w| < \pi \end{cases}$$

Find the frequency response of the FIR filter designed using a rectangular window defined below:

$$W_{R}(n) = \begin{cases} 1, & 0 \le n \le 4 \\ 0 & \text{Otherwise} \end{cases}$$
 (10 Marks)

8 a. Draw the direct form – I and direct form – II realizations of an in linear time invariant system described by the following input output relation

2y(n) - y(n-2) - 4y(n-3) = 3x(n-2).

(06 Marks)

b. The transfer function of a discrete causal system is given by

H(z) = 
$$\frac{1-z^2}{1-0.2z^{-1}-0.15z^{-2}}$$

Draw the cascade and parallel form.

(08 Marks)

c. Realize the linear phase FIR filter having the following impulse response

$$h(n) = \delta(n) - \frac{1}{4}\delta(n-1) + \frac{1}{2}\delta(n-2) + \frac{1}{2}\delta(n-3) - \frac{1}{4}\delta(n-4) + \delta(n-5)$$
 (06 Marks)

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