2. Any revealing of identification, appeal to evaluator and /or equations written eg, 42+8=50, will be treated as malpractice.

Important Note: 1. On completing your answers, compulsorily draw diagonal cross lines on the remaining blank pages.

GBCS SCHEME



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

Time: 3 hrs. Max. Marks: 100

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

Module-1

1 a. Compute 4-point DFT of casual three sample sequence given by

$$x(n) = \frac{1}{3}$$
; $0 \le n \le 2$
= 0 : else.

(06 Marks)

b. State and prove linearity property of DFT.

(06 Marks)

c. Find the circular convolution of two finite duration sequences $x_1(n)$ and $x_2(n)$ using concentric circle method. Where $x_1(n)$ and $x_2(n)$ are given by

$$x_1(n) = \{1, -1, -2, 3, -1\}$$

$$x_2(n) = \{1, 2, 3\}.$$

Consider T = 1sec.

(08 Marks)

(08 Marks)

OR

2 a. Compute circular convolution using Stockham's method for following sequences: $x_1(n) = \{2, 3, 1, 1\}$ and $x_2(n) = \{1, 3, 5, 3\}$.

b. Find the output y(n) of a filter whose impulse response h(n) = (1, 2) and input signal $x(n) = \{1, 2, -1, 2, 3, -2, -3, -1, 1, 1, 2, -1\}$ using overlap save method. Use block length of N = 4.

Module-2

- 3 a. Develop decimation in time algorithm for finding FFT. Draw signal flow graph for N = 8 for DIT algorithm. (10 Marks)
 - b. Find the 8 point DFT of sequence $x(n) = \{1, 1, 0, 0, -1, -1, 0, 0\}$ using DIT FFT algorithm. Draw signal flow graph. (10 Marks)

OR

- 4 a. Develop a decimation in frequency FFT algorithm for N = 8. Draw signal flow graph.

 (10 Marks)
 - b. The DFT X(k) of sequence is given as, $X(k) = \{0, 2\sqrt{2}(1-j), 0, 0, 0, 0, 0, 0, 2\sqrt{2}(1+j)\}$. Determine the corresponding time sequence x(n) using DIF-FFT algorithm. Write its signal flow graph. (10 Marks)

Module-3

5 a. A system function of the normalized lowpass filter is given below:

$$H(s) = \frac{1}{s^2 + \sqrt{2s+1}}$$
. Determine $H(z)$ using impulse invariant transformation.

b. Design an analog filter with maximally flat response in the passband and an acceptable attenuation of -2dB at 20radians/second. The attenuation in the stop band should be more than 10dB beyond 30 radian/second. (12 Marks)

a. Transform the analog filter $H(s) = \frac{s+0.1}{(s+0.1)^2+9}$ into a digital filter using bilinear

transformation. The digital filter should have resonant frequency $w_r = \pi/4$. (05 Marks)

Design an analog Chebyshev filter with the following specifications:

Passband ripple: 1dB for $0 \le \Omega \le \text{rad/sec}$.

Stopband attenuation : -60dB for $\Omega \ge 50 \text{ rad/sec}$.

(10 Marks)

c. Let $H(s) = \frac{1}{s^2 + \sqrt{2s + 1}}$ represent the transfer function of a lowpass filter with a passband of

1 rad/sec. Use frequency transformation to find the transfer functions of the following analog filters.

- A lowpass filter with pass band of 10rad/sec i)
- A high pass filter with cut-off frequency of rad/sec. ii)

(05 Marks)

Module-4

Compare Butterworth and Chebyshev filter approximations. 7

(05 Marks)

Design a digital low pass filter to satisfy the following pass band ripple $1 \le H(j\Omega) \le 0$, for $0 \le \Omega \le 1404\pi$ rad/sec and stop band attenuation $|H(\Omega)| > 60\,dB$ for $\Omega \ge 8268$ π rad/sec

sampling interval
$$T_s = \frac{1}{10^4} \text{sec}$$
. Use BLT for designing.

(15 Marks)

A discrete time system H(z) is expressed as 8

$$H(z) = \frac{10\left(1 - \frac{1}{2}z^{-1}\right)\left(1 - \frac{2}{3}z^{-1}\right)\left(1 + 2z^{-1}\right)}{\left(1 - \frac{3}{4}z^{-1}\right)\left(1 - \frac{1}{8}z^{-1}\right)\left[1 - \left(\frac{1}{2} + j\frac{1}{2}\right)z^{-1}\right]\left[1 - \left(\frac{1}{2} - j\frac{1}{2}\right)z^{-1}\right]}.$$

For the discrete time system defined by H(z), find the difference equation of the system.

(02 Marks)

- b. For the discrete time system, H(z) realize the system in direct form-I and II. (08 Marks)
- For the discrete time system H(z), realize parallel and cascade forms using second order (10 Marks) sections.

The desired frequency response of the low pass filter is given by
$$H_{d}(e^{jw}) = H_{d}(w) = \begin{cases} e^{-j3w}; & |w| < 3\pi/4 \\ 0; & 3\pi/4 < |w| < \pi \end{cases}$$

Determine the frequency response of FIR filter if the hamming window is used, with N = 7.

b. Design an ideal band pass filter with frequency response.

 $H_d(e^{jw}) = 1$, for $\frac{\pi}{4} \le |w| \le \frac{3\pi}{4}$. Use rectangular window with N = 11 in the design.(12 Marks)

Determine the impulse response h(n) of a filter having desired frequency response.

 $H_{d}(e^{jw}) = \begin{cases} e^{-j(N-1)w/2} & \text{for } 0 \le |w| \le \pi/2 \\ 0 & \text{for } \frac{\pi}{2} \le |w| \le \pi \end{cases}$ N = 7, use frequency sampling approach.(10 Marks)

- Realize the following system function $H(z) = 1 + \frac{3}{4}z^{-1} + \frac{17}{8}z^{-2} + \frac{3}{4}z^{-3} + z^{-4}$ in
 - ii) Cascaded form. i) Direct form (10 Marks)