Sr. No. 020502

December 2023

B. Tech (RAI) 5th SEMESTER

Digital Signal Processing (PCC-RAI-502-21)

Time: 3 Hours

Max. Marks:75

Instructions:
1. *It is compulsory to answer all the questions (1.5 marks each) of Part -A in short.*
2. *Answer any four questions from Part -B in detail.*
3. *Different sub-parts of a question are to be attempted adjacent to each other.*
4. *Any other specific instructions*

PART -A

Q1 (a) State and prove scaling property of DFT. (1.5)

(b) Write the significance of ROC for Z-Transform. (1.5)

(c) What is meant by stability and causality of signal? (1.5)

(d) Differentiate between linear and circular convolution. (1.5)

(e) State signal flow graph? (1.5)

(f) What is the significance of Fast Fourier Transform in signal processing. (1.5)

(g) Derive the twiddle factor's value for N/2 and N point values of DFT. (1.5)

(h) Differentiate between FIR and IIR filters. (1.5)

(i) What do you mean by windowing technique of digital filters? (1.5)

(j) Draw the frequency response of a Low pass, High pass and Band pass filters. (1.5)

PART -B

Q2 (a) Explain the frequency domain sampling process in detail with an example. (7.5)

(b) Determine the relationship of DFT with other transforms. (7.5)

Q3 (a) Two discrete sequences are given below: (7.5)

$$x(n)=\{2,1,1,0,5\} \text{ and } h(n)=\{1,1,1,1\}$$

Determine the convolution of the sequences using

(i)Tabular method

(ii)Circular method

(b) Determine the inverse z-transform of $X(z)=z/(3z^2-4z+1)$. If the ROC are (7.5)

(i) $|z|>1$

(ii) $|z|<1/3$

(iii) $1/3< |z| <1$

Q4 Derive the bilinear transformation technique used to transform an analog IIR filter to digital IIR filter. (15)

Use bilinear transformation method to obtain H(Z) for T= 1 sec if :

$$H(s) = 1/(s^2+\sqrt{2}\,s+1)$$

Q5 (a) A filter is to be designed with following desired frequency response: (7.5)

$$H_d(e^{jw}) = \begin{cases} e^{-j2w} & ,-\pi/4 \le w \le \pi/4 \\ 0 & ,\pi/4 < w \le \pi \end{cases}$$

Determine the filter coefficients hd(n) if window function is defined as:

$$W(n) = \begin{cases} 1 & ,0 \le n \le 4 \\ 0 & ,\text{otherwise} \end{cases}$$

Also, Determine the frequency response $H(e^{jw})$ of designed filter.

(b) Differentiate between Butterworth and Chebyshev filters. Also write the formula of their impulse response and draw the Gain vs Frequency graph. (7.5)

Q6 How a system of digital filters can be designed with the help of Direct , Cascade and Parallel form. Explain with suitable example. (15)

Q7 Write short note on with example: (15)

(a) Z-Transform

(b) Multirate Digital Signal Processing