Code No: 157BG

R18

JAWAHARLAL NEHRU TECHNOLOGICAL UNIVERSITY HYDERABAD
B. Tech IV Year I Semester Examinations, February - 2025
DIGITAL SIGNAL PROCESSING
(Electrical and Electronics Engineering)

Time: 3 Hours **Max. Marks: 75**

Note: i) Question paper consists of Part A, Part B.
ii) Part A is compulsory, which carries 25 marks. In Part A, Answer all questions.
iii) In Part B, Answer any one question from each unit. Each question carries 10 marks and may have a, b as sub questions.

PART – A

(25 Marks)

1.a) A discrete-time signal given by $x[n]$ ={1,3,2} is interpolated by a factor of 2. Find the new signal. [2]
b) Define Time Invariant and stable systems. Also give examples. [3]
c) Define the DTFT for a discrete-time signal. How does it differ from the DFT when applied to the same signal? [2]
d) Compute the Inverse FFT of $X[k] = \{10, -4, -2, 0\}$. [3]
e) What is the key difference between the impulse invariance method and the bilinear transformation method? [2]
f) What is the effect of frequency warping in the bilinear transformation method, and how is it corrected? [3]
g) List the characteristics of FIR Digital Filters. [2]
h) Explain why IIR filters can become unstable and how this can be avoided during design [3]
i) What causes dead band effects in quantized systems? [2]
j) Derive the expression for the frequency response of a stable system given H(z). [3]

PART – B

(50 Marks)

2.a) An LSI system is described by the impulse response $h[n] = \{1,1,1\}$ for n=0,1,2. Calculate the output $y[n]$ for the input $x[n] = \{1,2,3,4\}$.
b) A discrete-time system has the difference equation $y[n] - 0.5y[n-1] = x[n]$. Find the frequency response $H(e^{j\omega})$ of the system. [5+5]

OR

3.a) Determine whether the difference equation $y[n]-y[n-1]=x[n+2]+5x[n]-6x[n-3]$ is linear, time invariant and causal.
b) Given a continuous signal sampled at 100Hz that is represented in discrete form as $x[n] = \{0,1,2,3,4\}$. Perform down sampling by 5 and up sampling by 2 and obtain the final output? [5+5]

4.a) Find the Fourier Series representation of the periodic function defined as

$$x(t) = \begin{cases} t & if -\pi < t < \pi \\ 0 & otherwise \end{cases}$$

b) Use the Over-Lap Add method to compute the DFT of a long signal $x[n]$ = {1,2,3,4,5,6,7,8} with an FIR filter $h[n]$ = {0.5, 0.5}. [5+5]

OR

5.a) Calculate the 8-point FFT of the signal $x[n]$ ={1,−1,0,2,0,0,0,0} using Decimation-in-Time algorithm.

b) State and prove the properties of Circular Convolution and Parseval's Theorem of DFT.[5+5]

6.a) Design a digital low-pass filter using the Impulse Invariance Technique based on a Butterworth filter to meet the specifications of Sampling frequency f_s = 1 kHz, Passband edge frequency f_p = 200 Hz, Stopband edge frequency f_s = 400 Hz, Passband ripple A_p=3 dB, Stopband attenuation A_S = 20 dB.

b) A prototype low-pass filter is defined by the transfer function $H_{LP}(z) = \frac{1+0.4z^{-1}}{1-0.7z^{-1}+0.12z^{-2}}$. Transform this low-pass filter into a high-pass filter with the same cutoff frequency using the spectral transformation method. [5+5]

OR

7.a) Design a digital low-pass filter using a Chebyshev Type-I analog filter and the bilinear transformation method to meet the specifications of Sampling frequency f_s = 2 kHz, pass band edge frequency f_p = 400 Hz, stop band edge frequency f_s = 800 Hz, pass band ripple A_p=1 dB, stop band attenuation A_S = 30 dB.

b) A low-pass filter is defined as $H_{LP}(z) = \frac{1+0.5z^{-1}+0.25z^{-2}}{1-0.9z^{-1}+0.81z^{-2}}$. Design another low-pass filter with a scaled cutoff frequency of $0.6f_s$ using spectral transformation assuming f_s = 3 kHz.[5+5]

8.a) Design a band-pass FIR filter using the Fourier method with the specifications of pass band edge frequency range $f_{p1} = 500\ Hz, f_{p2} = 1000\ Hz$, stop band edge frequency range $f_{s1} = 200\ Hz, f_{s2} = 1800\ Hz$, sampling frequency f_s = 6 kHz, pass band ripple A_p=1 dB, stop band attenuation A_S = 40 dB. Use the Fourier method to determine the filter coefficients and order of the filter.

b) Explain how the Windowing Technique is used to design a low-pass FIR filter. [6+4]

OR

9.a) Design a high-pass FIR filter using the Hamming window with the following specifications: sampling frequency f_s = 4 kHz, pass band edge frequency f_p=1000 Hz, stop band edge frequency f_s = 500 Hz, pass band ripple A_p=1 dB, stop band attenuation A_S = 20 dB. Find the filter order and the coefficients using the Hamming window.

b) Design a band-stop FIR filter using the frequency sampling technique with the following specifications: Pass band edge frequency range $f_{p1} = 500\ Hz, f_{p2} = 2000\ Hz$, stop band edge frequency range $f_{s1} = 1000\ Hz, f_{s2} = 2000\ Hz$, sampling frequency f_s = 6000 Hz, pass band ripple A_p=1 dB, stop band attenuation A_S = 30 dB. Determine the filter order and calculate the filter coefficients using the frequency sampling technique. [5+5]

10.a) Find the system function $H(z)$ for the difference equation
$y[n] - 0.7y[n-1] + 0.12y[n-2] = x[n] + 0.5x[n-1]$

b) Draw the structure of the filter $H(z) = \frac{z^2+1.5z+0.5}{z^3-0.6z^2+0.25z-0.05}$ using Direct Form II realization.

[5+5]

OR

11.a) Determine whether the system $H(z) = \frac{z+0.5}{z^2-0.7z+0.12}$ is stable or not.

b) Given the recursive difference equation $y[n] = 0.5y[n-1] + x[n]$. Assume x[n] = 0 for $n \geq 1$ and the initial condition is $y[0] = 0.25$. Check if the system exhibits a limit cycle with a quantization step of $\Delta = 0.1$.

[5+5]

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