Code No: 136BE

R16

JAWAHARLAL NEHRU TECHNOLOGICAL UNIVERSITY HYDERABAD

B. Tech III Year II Semester Examinations, March - 2024

DIGITAL SIGNAL PROCESSING

(Electronics and Communication 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) Determine the fundamental period of the signal $x(n) = cos\left(\frac{60\pi n}{110}\right)$. [2]
b) Outline Parseval's relation in Z-transform. [3]
c) State the property of Circular Shift of a Sequence in DFT. [2]
d) State the difference between DIT and DIF algorithms. [3]
e) How is the order of a Butterworth filter required to achieve a certain stopband attenuation determined? [2]
f) What is meant by Warping effect and how can it be eliminated? [3]
g) What are the desirable characteristics of a Window? [2]
h) Discuss situations where FIR filters are preferred over IIR filters, and vice versa. [3]
i) Define upsampling and interpolation and explain their relationship. [2]
j) What are the strategies for optimizing filter design to achieve the desired balance between round-off and overflow noise? [3]

PART – B

(50 Marks)

2.a) Find the output response of the discrete time system described by the following difference equation
$y(n) - 0.86y(n-1) + 0.186y(n-2) = x(n)$ where $x(n) = \left(\frac{8}{9}\right)^n u(n)$ subjected to the initial conditions $y(-1) = 1$ and $y(-2) = 2$. Also find out the step response.
b) For the following discrete time signals, determine whether or not the system is linear, shift invariant, causal and stable
i) $y(n) = \sinh[nx(n) + x(n+5)]$ ii) $y(n) = x(-n-7)$ [5+5]

OR

3.a) Solve for $x(\infty)$, if $X(Z)$ is given by

$$\frac{z+4}{8(z^2-1)(z+0.7)}$$

b) Determine the system function $H(z)$, impulse response $h(n)$, magnitude response and phase response of the LSI system $y(n) = x(n) + 3x(n-1) + 2y(n-1) - y(n-2)$. [3+7]

4.a) Given a sequence x[n] = {1, 2, 3, 4, 5} and a length N = 3, compute its DFT using the Over-Lap Save method.

b) For a given discrete-time signal x[n] and its Z-transform X(z), express the relationship between DTFT, DFS, DFT, and Z-Transform for this signal. [5+5]

OR

5.a) The 8-point DFT of the sequence $x(n) = \left\{\frac{1}{\sqrt{2}}, 1, \frac{1}{\sqrt{2}}, 0, -\frac{1}{\sqrt{2}}, -1, -\frac{1}{\sqrt{2},}, 0\right\}$. Solve the DFT of the sequence using DIT-FFT algorithm.

b) Evaluate the IDFT of the sequence $X(K) = \{1, 1+j, 2, 1-2j, 0, 1+2j, 0, 1+j\}$ [5+5]

6.a) Design a digital Chebyshev low pass filter satisfying the following specifications.

$$0.707 \le |H(e^{j\omega})| \le 1.0, \; 0 \le \omega \le 0.4\pi$$
$$|H(e^{j\omega})| \le 0.2, \; 0.6 \le \omega \le \pi \text{ with } T = 1 \text{ sec}$$

Using Bilinear transformation method.

b) Discuss the transformations relating low pass filter to HPF, BPF and BRF in Digital domain with necessary equations. [5+5]

OR

7.a) Design a Band pass Butterworth filter with sampling frequency F=7KHz, $\alpha_p = 3dB$ in the passband 800Hz ≤ f ≤ 1000 Hz, $\alpha_s = 40dB$ in the stopband 4000Hz ≤ f ≤ ∞

b) Describe the design of Analog Low pass filter using Chebyshev Type-I filter with necessary equations. [5+5]

8.a) Explain the steps involved in designing an FIR filter using the frequency sampling technique.

b) Design an ideal differentiator with frequency response.

$$H(e^{j\omega}) = j\omega; \; - -\pi \le \omega \le \pi$$

Using hamming window with N=11. [5+5]

OR

9.a) Discuss the Fourier series method for designing FIR filters. What are its advantages and limitations?

b) Design an FIR low-pass filter with a cutoff frequency of 1000 Hz using the Hanning window function. The filter should have a passband ripple of 0.1 dB and a stopband attenuation of at least 60 dB. Use a sampling frequency of 8000 Hz. [5+5]

10.a) With necessary derivations, explain the operation of sampling rate conversion by a factor of I/D in both frequency and time domains.

b) Explain the concept of polyphase decomposition and its role in bandpass signal conversion. [5+5]

OR

11.a) Explain how computational output round-off noise affects the performance of digital filters.

b) Investigate the dead band effects in the implementation of an IIR digital filter with the transfer function $H(z) = \frac{1}{1-0.8z^{-1}}$. Determine the magnitude of dead band errors introduced by quantizing the filter coefficients to 12 bits and inputs ranging from -2048 to 2047. [5+5]

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