Code No: 126VK **R15**

JAWAHARLAL NEHRU TECHNOLOGICAL UNIVERSITY HYDERABAD
B. Tech III Year II Semester Examinations, February - 2023
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) Obtain the system function for the LTI system $y(n) + y(n-2) = x(n) + x(n-3)$. [2]
b) Find the causality and stability of the systems $y(n) = x(-n) + x(n-4) + x(3n-2)$. [3]
c) Compute the N-point DFT of $x(n) = \delta(n - n_0)$ [2]
d) Obtain the relation between DFT and Z-transform [3]
e) How is the drawback in bilinear transformation method eliminated? [2]
f) Find the equivalent digital transfer function H (z) by using impulse invariant method for the analog transfer function H(s) = 1/(S+2). Assume T=0.5sec [3]
g) What is the necessary and sufficient condition for the linear phase characteristic of an FIR filter? [2]
h) What are the advantages and disadvantages of the window technique for designing FIR filter? [3]
i) What are the applications of Multirate DSP? [2]
j) Elaborate the meaning of fractional sampling rate conversion. [3]

PART – B

(50 Marks)

2.a) Determine the frequency response for the system given by
$y(n) - \frac{4}{7}y(n-1) + \frac{1}{8}y(n-2) = x(n) + 5x(n-1)$
b) Check whether the following systems are Stable, Causal, Linear, Time Invariant and Memory less
i) $y(n) = \sin x(n)$ ii) $y(n) = \sum_{k=-\infty}^{\infty} x^2(k)$. [4+6]

OR

3.a) For the system function $H(z) = \frac{1+2z^{-1}+z^{-2}}{1-\frac{5}{7}z^{-1}+\frac{3}{7}z^{-2}}$, draw a signal flow graph that implements this system as a cascade form & parallel form realization.
b) A system is given by the difference equation $y(n) - \frac{3}{7}y(n-1) = x(n)$. Determine the solution when the input is $x(n) = \left(\frac{1}{7}\right)^n u(n)$ and the initial condition is given by $y(-1) = 1$ using Z-transform. [5+5]

4.a) Compute the DFT for the sequence $x(n) = \{0.5,0.5,0.5,0.5,0,0,0,0\}$. Also plot the magnitude and phase response.

b) Find the circular convolution using DFT and IDFT of the sequence $x_1(n) = \{4,3,1,2\}$ and $x_2(n) = \{1,3,5,3\}$. [5+5]

OR

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

b) Find the IDFT of the sequence $X(K) = \{1,1+j,2,1-2j,0,1+2j,0,1+j\}$. [6+4]

6.a) Design a digital Chebyshev filter using Bilinear transformation method satisfying the constraint

$$0.75 \leq |H(e^{j\omega})| \leq 1.0,\ 0 \leq \omega \leq \pi/2$$

$$|H(e^{j\omega})| \leq 0.2,\ 0.75\pi \leq \omega \leq \pi \text{ with T= 1 sec .}$$

b) What is a Filter? Mention the different types of filters based on frequency response. Also explain how an Analog filter is converted in to a Digital filter. [5+5]

OR

7.a) Design a Band pass Butterworth filter with sampling frequency f = 7 KHz, $\alpha_p = 3\,dB$ in the passband 800Hz ≤ f ≤ 1000 Hz, $\alpha_s = 30dB$ in the stopband 2000Hz ≤ f ≤ ∞.

b) Discuss the design procedure of Impulse Invariance method. Also explain why impulse invariant method is not preferred in the design of high pass IIR filter? [5+5]

8.a) Determine the filter coefficients of an FIR filter for the desired frequency response

$$H(e^{j\omega}) = \begin{cases} e^{-j2\omega} & |\omega| \leq \pi/4 \\ 0 & \pi/4 \leq |\omega| \leq \pi \end{cases}$$

Find the frequency response $H(\omega)$ of the filter using rectangular window.

b) Describe the frequency sampling realization of an FIR filter. [6+4]

OR

9.a) Determine the impulse response of an FIR filter with Pass band edge frequency of 1.5kHz, Stop Band Edge frequency of 2kHz, Sampling frequency of 8kHz using Hamming Window.

b) Distinguish between the different windows used in the design of FIR filters. [6+4]

10.a) With necessary examples, explain the difference between single-rate and multi-rate systems.

b) Explain the decimation and interpolation process with an example. Also find the spectrum. [5+5]

OR

11.a) Describe in detail about the Roundoff Noise in IIR digital Filters.

b) Discuss about the zero-input limit cycle oscillations due to finite word length of registers. [5+5]

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