R18

Code No: 156AR

JAWAHARLAL NEHRU TECHNOLOGICAL UNIVERSITY HYDERABAD

B. Tech III Year II Semester Examinations, July - 2023

DIGITAL SIGNAL PROCESSING

(Common to ECE, EIE)

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)	What are the conditions a system to be a linear?	[2]
b)	How to convert continuous signal to discrete signal?	[3]
c)	State Dirichlet conditions.	[2]
d)	What is the relation between ZT and DFT?	[3]
e)	What are the different conversion methods are used to convert analog filters to digital filters?	[2]
f)	What are the advantages of Chebyshev filters? What are the different types of it?	[3]
g)	What is effect of window length in the design of FIR filters?	[2]
h)	Compare FIR and IIR filters.	[3]
i)	What are the conditions that a system to be stable?	[2]
j)	What are the three important factors are presented for choosing among the various IIR system realizations?	[3]

PART – B

(50 Marks)

2.a) The discrete time system y(n)=ny(n-1) +x(n), n>0 is at rest. Check if the system is linear time invariant and BIBO stable.

b) Determine the impulse response h(n) for the system described by the second-order difference equation v(n) - 3v(n - 1) - 4v(n - 2) = x(n) + 2x(n - 1). [5+5]

OR

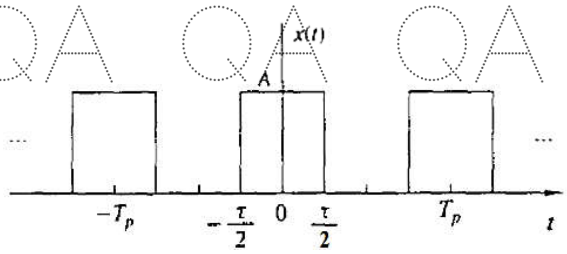
3.a) Determine if the system having the system function:

$$H(z) = \frac{1}{1 - \frac{2}{4}z^{-1} - \frac{1}{2}z^{-2}}$$

Is stable.

b) Design an interpolator that increases the input sampling rate by a factor of I = 5. [5+5]

4.a) Determine the Fourier series of the rectangular pulse train signal show in Figure below:



b) Find eight point DFT for the following sequence: $x(n) = \begin{cases} 1 & n = 0 \\ 0 & 1 \le n < 4 \\ 1 & 5 \le n < 7 \end{cases}$ [5+5]

OR

5.a) Compute the eight point DFT of the sequence $x(n) = \begin{cases} 1, & 0 \le n < 7 \\ 0, & otherwise \end{cases}$. By using the decimation-in-frequency FFT algorithm.

b) Draw the flow graph of the radix-2 DIT FFT algorithm for N = 16 and eliminate all signal paths that originate from zero inputs assuming that only x(0) and x(1) are nonzero. [6+4]

6.a) Determine the order and the poles of a lowpass Butterworth filter that has a -3-dB bandwidth of 500 Hz and an attenuation of 40 dB at 1000 Hz.

b) Convert the analog filter with system function:

$$H_a(s) = \frac{s + 01}{(s + 0.1)^2 + 16}$$

Into a digital IIR filter by means of the impulse invariance method. The digital filter is to have a resonant frequency of $w_r = \pi/2$. [4+6]

OR

7. Find the system function H(z) of the lowest order Chebyshev digital filter that meets the following specifications:
a) 1 dB ripple in the passband $0 \le |w| \le 0.3\pi$
b) At least 60dB attenuation in the stopband $0.35\pi \le |w| \le \pi$. Use the bilinear transformation. [5+5]

8. Design an FIR linear phase, digital filter approximating the ideal frequency response

$$H_d(w) = \begin{cases} 1 & for\ |w| \le \pi \\ 0 & for\ \frac{\pi}{2} \le |w| \le \pi \end{cases}$$

Determine and plot the magnitude and phase response of the filter using Hanning window. [10]

OR

9. Determine the coefficients {h(n)} of a linear-phase FIR filter of length M = 15 which has a symmetric unit sample response and a frequency response that satisfies the condition. [10]

$$H_r\left(\frac{2\pi k}{15}\right) = \begin{cases} 1 & k = 0,1,2,3 \\ 0 & k = 4,5,6,7 \end{cases}$$

10. Given a filter $y(n) = \frac{1}{3}[x(n) + x(n-1) + x(n-2)]$. Obtain and plot the magnitude, dB magnitude, phase, frequency response for the filter. [10]

OR

11. Consider an FIR filter with system function
$H(z) = 1 + 2.88\ z^{-1} + 3.40z^{-2} + 1.74z^{-3} + 0.4z^{-4}$.
Sketch the direct form I, parallel form realizations of the filter and determine in detail the corresponding input and output equations. Is the system minimum phase? [10]

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