Code No: 117CK **R13**

JAWAHARLAL NEHRU TECHNOLOGICAL UNIVERSITY HYDERABAD
B. Tech IV Year I Semester Examinations, March - 2017
DIGITAL SIGNAL PROCESSING
(Electrical and Electronics Engineering)

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

Note: This question paper contains two parts A and B.

Part A is compulsory which carries 25 marks. Answer all questions in Part A. Part B consists of 5 Units. Answer any one full question from each unit. Each question carries 10 marks and may have a, b, c as sub questions.

Part- A (25 Marks)

1.a)	What are the applications of digital signal processing?	[2]
b)	Explain region of convergence and its properties.	[3]
c)	What is zero padding? What are its uses?	[2]
d)	State and prove time shifting property of DFT.	[3]
e)	Give any two properties of Butterworth low pass filter.	[2]
f)	What is warping effect? What is its effect on magnitude and phase response?	[3]
g)	What is Gibbs phenomenon?	[2]
h)	Explain the procedure for designing FIR filters using windows.	[3]
i)	What is overflow oscillations?	[2]
j)	What is the need for anti-aliasing filter prior to down sampling?	[3]

Part-B (50 Marks)

2.a) Determine the impulse response h(n) for the system described by the second order difference equation

$$y(n) - 4y(n-1) + 4y(n-2) = x(n-1)$$

b) Find the magnitude and phase response for the system characterized by the difference equation

$$y(n) = \frac{1}{2}x(n) + x(n-1) + \frac{1}{2}x(n-2)$$ [5+5]

OR

3.a) Check the following filter for time invariant, causal and linear

(i) $y(n) = (n-1)x^2(n+1)$

(ii) $y(n) = n^2 x(n-2)$

b) Draw the structures of cascade and parallel realizations of

$$H(z) = \frac{(1-z^{-1})^3}{\left(1-\frac{1}{2}z^{-1}\right)\left(1-\frac{1}{8}z^{-1}\right)}$$ [5+5]

4.a) Determine the 8 point DFT of the sequence

$$X(n) = \begin{cases} 1 & ; -4 \le n \le 4 \\ 0 & ; otherwise \end{cases}$$

b) Compare overlap-save method and overlap-add method. [5+5]

OR

5.a) Compute 4-point DFT of a sequence x (n) = { 0,1,2,3} using DIT algorithm.
b) Find the IDFT of the sequence using DIF algorithm [5+5]
X (k) = {10, -2-j2, -2, -2+j2}

6.a) What are the steps to design an analog Chebyshev low pass filter.
b) Apply bilinear transformation to

$$H(s) = \frac{2}{(s+1)(s+2)}$$ with T=1 Sec and find H (z). [5+5]

OR

7. Consider an analog filter with transfer function

$$H(s) = \frac{1}{(s+1)(s^2+s+1)}$$

Is this a Butterworth or Chebyshev filter? Obtain the transfer function of an IIR digital filter using impulse invariant transformation. Assume T = 1 Sec. [10]

8.a) Explain the type -1 FIR filter design procedure using frequency sampling method.
b) List the features of Blackman window spectrum. [10]

OR

9. Explain the design procedure of linear phase FIR filter using Fourier series method.[10]

10.a) What are the effects of finite word length in digital filters?
b) Explain limit cycles in recursive structures. [5+5]

OR

11.a) Explain interpolation process with an example.
b) Explain with block diagrams how can sampling rate be converted by a rational factor M/L both in time domain and frequency domain. [5+5]

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