R13

Code No: 126EK

JAWAHARLAL NEHRU TECHNOLOGICAL UNIVERSITY HYDERABAD

B.Tech III Year II Semester Examinations, May - 2016

DIGITAL SIGNAL PROCESSING

(Common to ECE, EIE)

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) Write four advantages of Digital Signal Processing over Analog Signal Processing. [2]
b) Show that the frequency response of a discrete system is a periodic function of frequency. [3]
c) Give the relation between DTFT and Z-Transform. [2]
d) Distinguish between Linear convolution and circular convolution. [3]
e) What are the advantages of Butterworth filter? [2]
f) What are the advantages and disadvantages of Chebysher filter. [3]
g) Define Impulse Response. [2]
h) Define sampling and Nyquist Rate. [3]
i) Define Decimation. [2]
j) What is the need for Multirate Digital Signal Processing? [3]

PART – B

(50 Marks)

2.a) Test the following systems for linearity, time invariance, causality and stability.
$y(n) = \sin(2nf\pi/F)x(n)$
b) A digital system is characterized by the following difference equation:
$Y(n) = x(n)+ay(n-1)$ Assuming that the system is relaxed initially, determine its impulse response. [5+5]

OR

3. By taking an example compute DFT by using Over-Lap save method. [10]

4.a) Compute the circular convolution of the sequences
$x_1(n) = \{1, 2, 0, 1\}$ and
$x_2(n) = \{2, 2, 1, 1\}$ Using DFT approach.
b) What is FFT? Calculate the number of multiplications needed in the calculation of DFT using FFT algorithm with 32 point sequence. [5+5]

OR

5.a) Prove the following properties.
i) $x^*(n) \rightarrow X^*((-K))_N R_N(K)$
ii) $x^*((-n))_N R_N(n) \rightarrow X_{ep}(k) = \frac{1}{2}\left[X((K))_N + X^*((-K))_N\right] R_N(K)$
b) Compare FFT for the sequence:
$x[n] = \{1, 0, 1, 1, 0, 1, 1, 1\}$ [5+5]

6.a) Discuss in detail about spectral transformations.
b) Explain how IIR digital filters are designed from analog filters. [5+5]

OR

7.a) Compare the impulse invariance and bilinear transformation methods.
b) Find the order and poles of a low pass Butterworth filter that has a -3db bandwidth of 400 Hz and an attenuation of 20db at 1KHz. [4+6]

8.a) Draw and explain frequency response of FIR digital filter.
b) Design a high pass filter using hamming window with a cut-off frequency of 1.2 radians/second and N=9. [5+5]

OR

9.a) List the designing steps of FIR filters using fourier method.
b) Design a low pass digital FIR filter using Kaiser Window satisfying the specifications given below.
Pass band cut-off frequency = 100 Hz.
Stop band cut-off frequency = 200 Hz.
Pass band ripple = 0.1dB
Stop band attenuation = 20 dB
Sampling frequency = 1000 Hz. [3+7]

10.a) What are the Dead band Effects? Discuss.
b) What is mean by sampling rate conversion? Explain. [5+5]

OR

11.a) What are Limit Cycles and discuss various types of Limit Cycles in brief.
b) Discuss the process of performing sampling rate conversion by an rational factor I/D. [6+4]

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