Code No: 126EK

R13

JAWAHARLAL NEHRU TECHNOLOGICAL UNIVERSITY HYDERABAD

B. Tech III Year II Semester Examinations, December - 2017

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) Define stability. [2]
b) List the applications of Z- transform. [3]
c) List the properties of DFS. [2]
d) What is the value of x(n)*h(n), $0 \le n \le 11$ for the sequences x(n)={1,2,0,-3,4,2,-1,1,-2,3,2,1,-3} and h(n)={1,1,1} if we perform using overlap save fast convolution technique? [3]
e) Why do we go for analog approximation to design a digital filter? [2]
f) Discuss about the pole locations for the digital Chebyshev filters. [3]
g) Compared different window Techniques. [2]
h) What conditions are to be satisfied by the impulse response of an FIR system in order to have a linear phase? [3]
i) Define up sampling and Down sampling. [2]
j) What are the issues in quantization during analog to digital conversion? [3]

PART - B

(50 Marks)

2.a) Check whether the following systems are stable, causal.
(i) $h(t) = te^{at} u(t)$ (ii) $h(n) = e^{n/2} u(n-4)$
b) Determine the impulse response of the system described by the difference equation y(n)-3y(n-1)-4y(n-2)=x(n)+2x(n-1) using Z transform. [4+6]

OR

3.a) A system is described by the difference equation y(n)-y(n-1)-y(n-2) = x(n-1). Assuming that the system is initially relaxed, determine its unit sample response h(n).
b) Show that an LSI system can be described by its unit step response. [6+4]

4. Implement the Decimation in frequency FFT algorithm of N-point DFT where N-8. Also explain the steps involved in this algorithm. [10]

OR

5.a) If x(n) is a periodic sequence with a period N, also periodic with period 2N. $X_1(K)$ denotes the discrete Fourier series coefficient of x(n) with period N and $X_2(k)$ denote the discrete Fourier series coefficient of x(n) with period 2N. Determine $X_2(K)$ in terms of $X_1(K)$.

b) What is FFT? Calculate the number of multiplications needed in the calculation of DFT using FFT algorithm with 32 point sequence. [5+5]

6.a) Find the order and poles of a low pass Butterworth filter that has a -3db bandwidth of 500 Hz and an attenuation of 40db at 1KHz.

b) Compare the impulse invariance and bilinear transformation methods. [6+4]

OR

7. Explain design of IIR digital filter using Impulse Invariant Techniques. [10]

8. Design a low pass digital FIR filter using Kaiser window satisfying the specifications given below.

Pass band cut-off frequency = 150 Hz.
Stop band cut-off frequency = 250 Hz.
Pass band ripple = 0.1dB
Stop band attenuation = 40 dB
Sampling frequency = 1000 Hz. [10]

OR

9. Design a high pass filter using hamming window with a cut-off frequency of 1.2 radians/second and N=9. [10]

10.a) Give the frequency domain analysis of Decimator.

b) Briefly discuss the dead –band effects. [5+5]

OR

11.a) Explain the necessity of multirate signal processing and hence define decimation and interpolation

b) Discuss the role of finite length representation and the associate errors. [5+5]

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