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Code No: 117CK

JAWAHARLAL NEHRU TECHNOLOGICAL UNIVERSITY HYDERABAD

B. Tech IV Year I Semester Examinations, November/December - 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) List out different realization forms of digital filters. [2]

b) Define the frequency response of a discrete-time system. [3]

c) Give the relation between z-transform and DTFT. [2]

d) Draw the basic butterfly diagram for DIF FFT algorithm. [3]

e) What is mean by bilinear transformation method of designing IIR Filter? [2]

f) What are the parameters that can be obtained from the Chebyshev filter specification? [3]

g) Under what condition an FIR filter will exhibit linear phase response. [2]

h) Write the features of Hamming window. [3]

i) What is overflow and truncation error? [2]

j) Why the limit cycle problem does not exist when FIR digital filter is realized in direct form? [3]

Part-B

(50 Marks)

2.a) Define an LTI System and show that the output of an LTI system is given by the convolution of Input sequence and impulse response.

b) Realize the following system in direct form I

$$H(Z) = \frac{1}{1 + a_1 z^{-1} + a_2 z^{-2}}$$

[5+5]

OR

3.a) Obtain the parallel realization of the system described by the difference equation

$$y[n] - \frac{13}{12}y[n-1] + \frac{9}{24}y[n-2] - \frac{1}{24}y[n-3] = x[n] + 2x[n-1]$$

b) Find the frequency response $H(e^{jw})$ of the linear time-invariant system whose input and output satisfy the difference equation $y[n] - \frac{1}{2}y[n-1] = x[n] + 2x[n-1] + x[n-2]$.

[5+5]

4. Define DFT and then state and prove properties of DFT. [10]

OR

5. Find the DFT of a sequence $x[n] = \{1, 2, 3, 4, 4, 3, 2, 1\}$ using:
   a) DIT algorithm b) DIF algorithm [10]

6. For the given specifications design an analog Butterworthy filter.
   $0.9 \leq |H(j\Omega)| \leq 1 \; for \; 0 \leq \Omega \leq 0.2\pi$
   $|H(j\Omega)| \leq 0.2 \; for \; 0.4\pi \leq \Omega \leq \pi$ [10]

OR

7. Using the bilinear transform, design a high pass filter, monotonic in password with cutoff frequency of 1000 Hz and down 10 dB at 350 Hz. The sampling frequency is 5000 Hz. [10]

8. Design an FIR low pass filter satisfying the following specifications.
   $\alpha_p \leq 1 \; dB; \; \alpha_s \geq 44 \; dB, \; w_p = 20 \; rad/sec, \quad w_s = 30 \; rad/sec, \; w_{sf} = 100 \; rad/sec$
   [10]

OR

9. Design a low pass filter using Hanning window with a cutoff frequency of 0.9 radians/sec and N=6. Draw the filter structure and plot its spectrum. [10]

10.a) Define Multirate systems and Sampling rate conversion.
   b) Discuss the sampling rate conversion by a factor I with the help of a neat block diagram. [3+7]

OR

11.a) What is overflow? When it occurs? What are the methods to prevent overflow? Explain.
   b) What is meant by "dead band" of the filter? Explain dead band of first order filter. [5+5]

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