Code No: 127CK

JAWAHARLAL NEHRU TECHNOLOGICAL UNIVERSITY HYDERABAD B.Tech IV Year I Semester Examinations, December - 2019 DIGITAL SIGNAL PROCESSING (Electrical and Electronics Engineering)

Time: 3 Hours

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 is a shift invariant system? Give example. [2] Find the transfer function of first order recursive filter. b) [3] c) Determine the basis matrix for 8-point DFT. [2] Which properties of the twiddle factors are used for development of fast algorithm for d) DFT computation and how? [3] List out the characteristics of Chebyshev Type I filter. [2] e) f) What are the different applications of IIR filter? [3] Why do we truncate the impulse response of the ideal filter? **g**) [2] h) What is the use of windowing in case of Hilbert transformer design? [3] i) How will you make the interpolator efficient? [2] i) What are limit cycle oscillations? [3]

PART-B

(50 Marks)

- 2.a) Find the impulse response and step response of a discrete-time linear time invariant system whose difference equation is given by y(n)=y(n-1)+0.5y(n-2)+x(n)+x(n-1)
 - b) A discrete time system has a unit impulse response h(n) given by $h \ n = \frac{1}{2}\delta(n+\delta(n-1)+\frac{1}{2}\delta(n-2)$. Find the system frequency response H($e^{j\omega}$); plot magnitude and phase response. [5+5]

OR

- 3.a) Determine the impulse response of the system described by difference equation y n = y n 1 0.5y n 2 + x n + x n 1. Plot the pole zero response and discuss on the stability.
 - b) Obtain the direct form II and parallel form for $y \ n \ -\frac{5}{6}y \ n-1 \ +\frac{1}{6}y \ n-2 \ =x \ n \ +2x(n-1)$ [5+5]
- 4.a) State and prove the properties of Discrete Fourier Transform.
 - b) The input sequence is $x(n) = \{1,2,3,4,5,1,2,3,4,5,1,2,3,4,5\}$. Let the impulse response of the filter be $h(n) = \{3,2,1,1\}$. Use overlap and add method to calculate the convolution. [5+5]

OR

- 5.a) Compute eight point FFT for $x(n)=\{1,2,2,1,1,2,1,1\}$ using decimation in time FFT Algorithm.
 - b) With necessary equations obtain the relation between DTFT and DFT. [5+5]

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Max.Marks:75

6.a) Using a impulse invariant transformation with T=1sec, design a digital Butterworth filter which satisfies the following conditions:

$$\begin{array}{l} 0.75 \leq H(e^{j\omega}) \leq 1 \ ; \ 0 \leq \omega \leq \frac{\pi}{2} \\ H(e^{j\omega}) \leq 0.2 \ ; \ \frac{3\pi}{4} \leq \omega \leq \pi \end{array}$$

b) What is the meaning of bilinear transformation? Explain the mapping between S-domain and Z-domain using this method. [5+5]

OR

- Design a digital Chebyshev filter for the following specifications using BLT method. The 7.a) filter is required to have 2dB ripple in the pass band edge frequency of 2000Hz and an attenuation of 40dB at 6000 Hz. The sampling frequency is 10000Hz.
 - What is the principle of impulse invariant method? Explain the mapping between b) S domain and Z domain for impulse invariant method. Can you convert stable analog filter in to a stable digital filter using this method. [5+5]
- 8.a) Design FIR digital band-pass filter with the magnitude response is specified as

$$H(f) = \begin{array}{c} 1 & for \ 100 \le f \le 200 Hz \\ 0 & otherwise \end{array}$$

Use Hanning window.

- What is the use of introducing delays in the impulse response of the filter? b) [5+5] OR
- 9.a) Design a 9-coefficient FIR HPF using frequency sampling method with cut-off frequency $\frac{3F_s}{9}$ where F_s is the sampling frequency. Plot the magnitude response of the resulting filter. Explain Fourier series expansion method for FIR filter design. [6+4]
 - b)
- 10.a) Draw the block schematic for a decimator and explain the need for a filter. How is aliasing Avoided? Draw we spectrum of the signal after filtering and after decimation process.
 - Design a two stage interpolator to increase the sampling frequency from 128KHz to b) 512KHz. [5+5]
 - OR

11.a) Consider the transfer function

H z =
$$\frac{1}{(1 - 0.8z^{-1})(1 - 0.72z^{-1})}$$

Find the steady-state noise power due to product round-off.

b) What are the methods to avoid overflow of adders?

[5+5]

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