Total No. of Questions : 10]

SEAT No. :

P2382

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T.E. (E & TC)

DIGITAL SIGNAL PROCESSING

(2012 Course) (Semester - I) (304182) (End Semester)

Time : 2½ Hours] *[Max. Marks :70*

Instructions to the candidates:

1) *Neat diagrams must be drawn wherever necessary.*
2) *Figures to the right indicate full marks.*
3) *Use of logarithmic tables slide rule, Mollier charts, electronic pocket calculator and steam tables is allowed.*
4) *Assume suitable data, if necessary.*

Q1) a) Consider the analog signal $x_a(t)$ as $x_a(t) = 6 \cos 50 \pi t + 3 \sin 200 \pi t - 3 \cos 100 \pi t$ **[5]**

i) Determine the minimum sampling frequency.

ii) Determine $x(n)$ at minimum sampling frequency.

iii) Sketch the waveform and show the sampling points.

b) Determine the transfer function and impulse response of the LTI system given by the difference equation. **[5]**

$$y(n)+\frac{3}{4}y(n-1)+\frac{1}{8}y(n-2)=x(n)+x(n-1)$$

OR

Q2) a) State and prove convolution property of Z transform. **[5]**

b) Compute 4- points DFT of the sequence given by $x(n) = (-1)^n$ using DIT FFT algorithm. **[5]**

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Q3) a) State four important advantages of digital signal processing over analog signal processing. **[4]**

b) For the following sequences, **[6]**

$$x_1(n) = \begin{cases} 1 & 0 \le n \le 2 \\ 0 & \text{otherwise} \end{cases}$$

$$x_2(n) = \begin{cases} 1 & 0 \le n \le 2 \\ 0 & \text{otherwise} \end{cases}$$

Compute linear convolution using circular convolution.

OR

Q4) a) Using partial fraction expansion, find inverse Z-Transform of following system function and verify it using long division method, **[5]**

$$H(Z) = \frac{1 + 2\ Z^{-1}}{1 - 0.4Z^{-1} - 0.12\ Z^{-2}} \quad \text{if } h(n) \text{ is causal.}$$

b) State and prove circular time shift property of DFT **[5]**

Q5) a) Design a butterworth digital IIR lowpass filter using bilinear transformation to satisfy following specifications: **[10]**

$$\begin{aligned} 0.6 \le |H(e^{jw})| \le 1.0 \quad & 0 \le w \le 0.35\pi \\ |H(e^{jw})| \le 0.1 \quad & 0.7\pi \le w < \pi \end{aligned} . \quad \text{Use T = 0.1 seconds.}$$

b) Compare between Bilinear transformation method and impulse invariant method. **[3]**

c) Draw direct form I & direct form II realisations for the second order system given by: **[4]**

$$y(n) = 2b \cos w_0\ y(n\text{-}1) - b^2\ y(n\text{-}2) + x(n) - b \cos w_0 x(n\text{-}1)$$

OR

Q6) a) The system function of an analog filter is given by **[4]**

$$H(s) = \frac{s+0.2}{(s+0.2)^2+9}$$

Convert it to digital filter using Impulse Invariant technique. Assume T = 1 second.

b) Given $H(s) = \frac{1}{s+1}$. Apply impulse invariant method to obtain digital filter transfer function and difference equation. Assume T = 1 second. **[4]**

c) For the system given by following equation **[9]**

$$H(z) = \frac{1-z^{-1}}{1-0.2z^{-1}-0.15z^{-2}}.$$

Draw cascade and parallel realisation.

Q7) a) Design a linear phase FIR band pass filter using hamming window with cut off frequencies 0.2 rad/sec & 0.3 rad/sec. M = 7. **[9]**

b) Explain the characteristics of window function. **[4]**

c) Distinguish between FIR and IIR filter. **[4]**

OR

Q8) a) Design a linear phase FIR lowpass filter with a cutoff frequency of 0.5 rad/sample by taking 11 samples of ideal frequency response. **[9]**

b) What is Gibb's phenomenon? How it is reduced? **[4]**

c) Show that the filter with symmetric impulse response has linear phase response. **[4]**

Q9) a) With the help of neat diagram, and waveform explain sampling rate conversion by non-integer factor. **[4]**

b) Sampling rate is to be reduced from 96kHz to 1 kHz. Highest frequency of interest is 450 Hz $\delta_P = 0.01$, $\delta_S = 0.001$. Design a decimator with decimating factors of 32 and 3. **[6]**

c) Write short notes on **[6]**

i) MAC unit

ii) Barrel shifter

OR

Q10) a) What is the role of anti aliasing filter & anti imaging filter in decimator & interpolator, respectively. **[4]**

b) Describe four important features of a digital signal processor. **[6]**

c) Explain the architecture of TMS 320C67XX digital signal processor. **[6]**

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