

Total No. of Questions : 10]

SEAT No. :

P3592

[5560]-546

[Total No. of Pages : 2

T.E.(Electronics Engineering)
DSP & APPLICATIONS
(2015 Course) (Semester - II) (304206)

Time : 2½ Hours]

[Max. Marks : 70

Instructions to the candidates:

- 1) *Attempt Q1 or Q2, Q3 or Q4, Q5 or Q6, Q7 or Q8, Q9 or Q10.*
- 2) *Neat diagrams must be drawn wherever necessary.*
- 3) *Figures to right indicate full marks.*
- 4) *Assume Suitable data, if necessary.*
- 5) *Use of electronic pocket calculator is allowed.*

- Q1) a)** The sequence $x(n)$ is given as $x(n) = [1, 2, 3, -1, 2, 3]$. Draw & obtain the sequence if decimator factor is 2 & interpolator factor is 2. [6]
- b)** Obtain IDFT of the following sequence $X(k) = [2 \ 1-j \ 0 \ 1+j]$. [4]

OR

- Q2) a)** Determine all possible sequences for the system given by z transform

$$X(z) = \frac{z(z^2 - 2z + 3)}{(z - 2)(z - 3)(z - 4)} \quad [6]$$

- b)** Describe in brief aliasing effect. What is use of antialiasing filter in sampling. [4]

- Q3) a)** Obtain linear convolution of two sequences determined as $x(n) = \begin{bmatrix} 4 & 5 & 3 \end{bmatrix}$ & $h(n) = \begin{bmatrix} 2 & 3 \end{bmatrix}$ using circular convolution. Also verify your answer with traditional linear convolution method. [6]

- b)** Derive relation between fourier transform & z transform. [4]

OR

- Q4) a)** An LTI system is represented by difference equation $y(n) = 3y(n-1) - 2y(n-2) + x(n) - x(n-1)$. [6]

- i) Obtain system function $H(z)$.
- ii) Draw & obtain pole zero plot
- iii) Comment on stability of the system from pole zero plot

- b)** Illustrate with neat sketch radix 2 DIT & DIF FFT butterfly structures to evaluate 4 point DFT of a signal. [4]

P.T.O.

- Q5) a)** Derive an expression for transformation of analog system function $H_a(s)$ to digital system function $H(z)$ using Impulse Invariance Method. Also illustrate with neat sketch relationship of s plane to z plane. Comment on drawback of Impulse Invariance Method. [9]
- b)** Realize the system described by following system function using direct form I & direct form II structure. [8]

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

OR

- Q6) a)** Design a digital low pass butterworth IIR filter using BLT for following specifications. [12]

$$F_{PB} = 1 \text{ kHz}, F_{SB} = 3 \text{ kHz}, F_s = 8 \text{ kHz}, A_p = 2\text{dB} \text{ \& } A_s = 15\text{dB}.$$

- b)** Describe in brief finite word length effect in IIR filter design. [5]
- Q7) a)** Describe in brief FIR filter characteristics. Also show that FIR filters for symmetric & antisymmetric impulse response gives linear phase. [8]
- b)** Using frequency sampling method, design a low pass FIR filter to meet following specifications. [9]

Passband - 0 - 6 kHz

Sampling frequency - 18 kHz

Filter length - 11

OR

- Q8) a)** Explain FIR filter designing using windowing technique. Compare different window functions wrt transition band, main lobe, peak side lobe etc. Also comment on role of kaiser window in FIR filter design. [9]
- b)** Draw & obtain cascade realization of FIR filter defined by system function [8]

$$H(z) = \left(1 + \frac{1}{2}z^{-1} + z^{-2}\right) \left(1 + \frac{1}{4}z^{-1} + z^{-2}\right)$$

- Q9) a)** Describe in brief application of DSP in DC motor control. [8]
- b)** Illustrate your answer with neat sketch & also describe in brief architecture of general DSP processor. [8]

OR

- Q10) a)** Explain how DSP is useful in interference cancellation in ECG. [8]
- b)** State & explain four important features of DSP processor. [8]

