Total No. of Questions : 10]

SEAT No. :

P3156

[Total No. of Pages : 3

[4858] - 1060

T.E. (Electronics) (Semester - II)

DISCRETE TIME SIGNAL PROCESSING

(2012 Pattern) (End Sem.)

Time : $2\frac{1}{2}$ Hours] ***[Max. Marks : 70***

Instructions to the candidates:

1) *Answer Q.1 or Q.2, Q.3 or Q.4, Q.5 or Q.6, Q.7 or Q.8, Q.9 or Q.10.*
2) *Neat diagrams must be drawn wherever necessary.*
3) *Figures to the right indicate full marks.*
4) *Use of electronic pocket calculator is allowed.*
5) *Assume suitable data, if necessary.*

Q1) a) An analog signal $x(t)=\sin(480\pi t)+3\sin(720\pi t)$ is sampled at 600 times per second.

i) What are the frequencies in radians in the resulting DT Signal $x(n)$.

ii) If $x(n)$ is passed through an ideal DAC, what is reconstructed signal $y(t)$ = ? **[6]**

b) Perform circular convolution of following sequences using matrix multiplication method.

$x_1(n)=\{1,2,3,4\}$

$x_2(n)=\{2,1,1,2\}$. **[4]**

OR

Q2) a) Draw butterfly structures of 8 point DIT FFT & 8 point DIF FFT. **[6]**

b) Give advantages of Digital Signal Processing over analog signal processing. **[4]**

Q3) a) A system has unit sample response $h(n)$ given by **[6]**

$$h(n)=-\frac{1}{4}\delta(n+1)+\frac{1}{2}\delta(n)-\frac{1}{4}\delta(n-1)$$

P.T.O.

i) Is the system BIBO stable.

ii) Is filter causal.

iii) Find frequency response.

b) State following properties of DFT **[4]**

i) Convolution in time domain (circular convolution).

ii) Time shifting (circular time shift).

OR

Q4) a) Compute inverse Z transform of the following: **[6]**

$$X(Z)=\frac{Z^2}{(Z-1)(Z-0.2)}.$$

b) State & explain sampling theorem. **[4]**

Q5) a) Design a FIR digital filter to approximate an ideal LPF with passband gain of unity, cut off frequency of 850 Hz & working at sampling frequency of 5000 Hz. The length of impulse response should be 5. Use rectangular & hamming window. **[9]**

b) Deduce cascade realization of **[4]**

$$H(z)=\left(1+\frac{1}{4}z^{-1}+z^{-2}\right)\left(1+\frac{1}{8}z^{-1}+z^{-2}\right).$$

c) Explain frequency sampling method for FIR filter design. **[4]**

OR

Q6) a) Determine impulse response *h* (*n*) of a filter having desired frequency response

$$\begin{aligned} H_d(e^{j\omega}) &= e^{-j}(M-1)\omega/2 & 0 \le \omega \le \pi/2 \\ &= 0 & \pi/2 \le \omega \le \pi \end{aligned}$$

M = 7. Use frequency sampling approach. **[10]**

b) Show that symmetric FIR filter has linear phase response. **[7]**

Q7) a) Explain Impulse invariance transformation. What is drawback of this transformation & how BLT overcomes it. Show graphical representation. Explain concept of frequency pre-warping. **[8]**

b) The system transfer function of analog filter is given by

$$H(s)=\frac{s+0.1}{(s+0.1)^2+16}$$

Obtain the system transfer function of digital filter using BLT which is resonant at $\omega_r = \pi/2$. **[9]**

OR

Q8) a) Consider LTI system, initially at rest described by difference equation

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

i) Determine impulse response $h(n)$ of the system.

ii) Determine direct form II & parallel form realization of this system.

[8]

b) Write short note on Butterworth filter approximation. **[5]**

c) Give comparison between IIR & FIR filters. **[4]**

Q9) a) Design two stage decimator with sampling rate to be reduced from 10 KHz to 500 Hz. Passband edge 150 Hz, stopband edge 180 Hz, passband ripple 0.002 & stopband ripple 0.001. Consider decimation factors 10 & 2. **[8]**

b) With the help of block diagram explain architecture of TMS320 C28XX processor. **[8]**

OR

Q10) a) Explain methods of sample rate reduction & increase. **[8]**

b) Explain implementation of triggering for converter with DSP processor. **[8]**