Total No. of Questions :10]

SEAT No. :

P1727

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[5058]-360

T.E. (Electronics)

DISCRETE TIME SIGNAL PROCESSING

(2012 Course) (End-Sem) (Semester - I) (304210)

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 electronic pocket calculator is allowed.*
4) *Assume suitable data, if necessary.*

Q1) a) A digital communication link carries binary coded words representing samples of input signal $x(t) = 3\cos 600\pi t + 2\cos 1800\pi t$. The link is operated at 10000 bits/s & each input sample is quantized into 1024 different voltage levels **[6]**

i) What is the sample frequency & folding frequency in Hz?

ii) What is Nyquist rate of sampling for x(t) in Hz?

iii) What is resolution of quantization?

b) Find sequence x(n) for which IDFT x(k) is given by X(k) = {3 2+j 1 2-j}. **[4]**

OR

Q2) a) Peform circular convolution of following two sequences **[6]**

$x_1(n) = \{1, 2, 3, 1\}$ $x_2(n) = \{4, 3, 2, 2\}$ using DFT & IDFT method.

b) Define & explain sampling theorem & alising effect. **[4]**

P.T.O.

Q3) a) Prove the following properties of z transform **[6]**

i) Time shifting

ii) Convolution of two sequences in time domain.

b) Derive DIT FFT flow graph for N = 4, hence find DFT of x(n) = {1, 2, 3, 4} **[4]**

OR

Q4) a) Compute the inverse z transform of the following

$$x(z) = \frac{z-1}{1-3z^{-1}} \text{ ---- ROC } |z| < 3.$$ **[6]**

b) Explain advantages of digital signal processing over analog signal processing. **[4]**

Q5) a) Explain in detail frequency sampling method of designing FIR filter. **[7]**

b) Design a low pass digital filter with cut off frequency $\omega_C = \pi/2$ using frequency sampling technique for N = 17. **[10]**

OR

Q6) a) Determine the filter coefficients hd(n) for the desired frequency response of a low pass filter given by **[8]**

$$H_d(e^{jw}) = \begin{cases} e^{-j2w} & -\pi/4 \le w \le \pi/4 \\ 0 & -\pi/4 \le w \le \pi \end{cases}$$

If we define new filter coefficients by $h_d(n)$ w(n)= h(n), where

$$w(n) = \begin{cases} 1 & \text{---- for } 0 \le n \le 4 \\ 0 & \text{---- otherwise} \end{cases}$$

then determine h(n).

b) Explain the need of window functions in design of FIR filter. Also explain advantages & disadvantages of window function. **[5]**

c) Realize a linear phase FIR filter with following impulse response. Give necessary equations $H(z) = \frac{2}{3}z + 1 + \frac{2}{3}z^{-1}$. **[4]**

Q7) a) Using Bilinear transformation, design a butter worth filter which satisfies the following conditions **[9]**

$$0.8 \le \left|H(e^{jw})\right| \le 1 \text{ --- } 0 \le \mathrm{w} \le 0.2\pi$$

$$\left|H(e^{jw})\right| \le 0.2 \text{ --- } 0.6\pi \le \mathrm{w} \le \pi \text{ \& Ts} = 1.$$

b) Explain impulse invariance transformation. What is drawback of this transformation & how BLT overcomes it. Show graphical representation. Explain concept of frequency prewarping. **[8]**

OR

Q8) a) Determine direct form I & II for the filter given by **[6]**

$$y(n) = 2b\cos w_o y(n-1) - b^2 y(n-2) + x(n) - b\cos w_o x(n-1).$$

b) Write short note on chebyshev filter approximation. **[4]**

c) Find out H(z) using impulse invariance method at 5Hz sampling frequency from H(s) as given below $\mathrm{H(s)} = \frac{2}{(s+1)(s+2)}$. **[7]**

Q9) a) Design 2 stage interpolator for following system consider one of the interpolator factor $I_1 = 2$. **[8]**

Baseband 0 - 20 KHz

I/P sampling frequency 44.1 KHz