Total No. of Questions : 10]

SEAT No. :

P 3274

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TE (Electronics) Endsem

DISCRETE TIME SIGNAL PROCESSING

(2012 Pattern)

Time : 2½ Hours] *[Max. Marks :70*

Instructions to the candidates:

1) *Answer the Q.No.1 or Q.No.2, Q.3 or Q.4 and Q.5 or Q.6, and Q.7 or Q.8 and Q.9 or Q.10*
2) *Neat diagram must be drawn wherever necessary.*
3) *Assume suitable data, if necessary.*

Q1) a) State the mathematical model used to represent a DT. System Define these model with mathematical form. **[5]**

b) Perform circular convolution of following sequences using DFT-IDFT method. **[5]**

i) $X_1(n) = \{1,2,3,4\}$

ii) $X_2(n) = \{2,1,1,2\}$

OR

Q2) a) Obtain 4 pint DFT for a sequence x(n) = {1,-2,2,1} and plot the magnitude spectrum. **[5]**

b) Obtain ZT of a DT signal using ZT properties where,

x(n) = 2n.u(n-1) **[5]**

Q3) a) What is the significance of 'N' in N point DFT? Define N point DFT by means of twiddle factor W and Compute twiddle factor for N = 4 **[5]**

b) Why the problem of aliasing is observed during the sampling process? Derive the relationship between analog frequency F and DT frequency f. **[5]**

OR

Q4) a) Compute inverse Z transform of the following.

$X(Z) = \{Z^2 / (Z-1)(Z-0.2)\}$ **[5]**

b) Draw a Poles Zero plot for a system described as-

y (n) = x (n) -x (n-1) + 0.2y (n-1) + 0.15 y (n-2) **[5]**

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Q5) a) Explain impulse invariance method of IIR filter design. **[6]**

b) The system function of analog filter is given by, $H_a(S) = \dfrac{s+0.1}{(s+0.1)\hat{}2+9}$

Design IIR Filter using impulse invariance method. **[6]**

c) What is wrapping effect? What is its effect on magnitude and phase response? **[4]**

OR

Q6) a) Compare Impulse variance & BLT. **[4]**

b) Discuss Design steps of IIR filter using bilinear transform method. How frequency response is obtained? **[6]**

c) An IIR low pass filter is required to meet the following specification : **[6]**

Passband peak to peak ripple :	$\leq$ 1dB
Passband edge :	1.2 kHz
Stopband attenuation :	$\geq$ 40 dB
Stopband edge :	2.5kHz
Sample rate :	8 kHz

The filter is to be designed by performing BLT on an analog system function of required order butterworth filter so as to meet the specifications in the implementation.

Q7) a) Explain important feature of window function. **[4]**

b) Using frequency sampling method, design a band pass filter with the following specifications. **[8]**

Sampling frequency = 8000Hz

Cutoff frequencies Fcl = 1000 Hz

F_{c_2} = 3000 Hz

Determine the filter coefficient for N = 7.

c) The impulse response of a system h(n) = a^n u(n), a $\neq$ 0. Determine a & sketch pole zero plot for this system to act as : **[5]**

i) Stable Low pass filter ii) Stable High pass filter

OR

Q8) a) Explain Gibb's Phenomenon. **[4]**

b) If transfer function of FIR LPF is $H(z) = 1 + 2Z^{-1} + 3Z^{-2} + 2Z^{-3} + Z^{-4}$ Show That

i) Impulse Response h(n) is Symmetry. **[5]**

ii) Phase Delay τ_p and Group delay τ_g are constant.

c) Design a FIR digital filter to approximate an ideal LPF with passband gain of unity, Cut off frequency of 850 Hz. And working at sampling frequency of 5000Hz. The Length of impulse response should be 5. Use Rectangular & hamming window. **[8]**

Q9) a) Explain sampling rate conversion by rational factor 1/D. **[3]**

b) Name two methods of sampling rate conversion with advantages & limitations. **[4]**

c) Draw the block schematic for decimator and explain the need for a filter.

Derive the expression for decimated output signal i.e. y(m) and draw the spectrum of the signal after filtering and after decimation process. **[10]**

OR

Q10) a) Explain pipelining concept. Also explain MAC, ALU and Barrel shifter unit of DSP Processor. **[12]**

b) Explain the application of DSP processor in Speech processing. **[5]**

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