Total No. of Questions : 10]

SEAT No. :

P2429

[Total No. of Pages : 3

[5253]-152

T.E. (E & TC)

DIGITAL SIGNAL PROCESSING

(2012 Pattern) (Semester - I)

Time : 2½ Hours] ***[Max. Marks : 70***

Instructions to the candidates:

1) ***Solve Q1 or Q2, Q3 or Q4, Q5 or Q6, Q7 or Q8 and Q9 or Q10.***
2) ***Neat diagrams must be drawn wherever necessary.***
3) ***Figures to the right indicate full marks.***
4) ***Assume suitable data if necessary.***

Q1) a) List the advantages of digital signal processing over analog signal processing. **[3]**

b) Explain the concept of orthogonality. **[3]**

c) What should be the sampling frequency to avoid aliasing for an analog signal represented as, $x(t) = \cos(150\pi t)+2\sin(300\pi t)- 4\cos(600\pi t)$. Obtain the discrete time signal if this sequence is sampled at $Fs = 400$Hz Does aliasing occurs? If yes, calculate the aliased frequencies from the original frequencies. **[4]**

OR

Q2) a) Perform the circular convolution of two sequences given below. **[4]**

$$X1(n) = \{\underset{\uparrow}{1}, 2, 3, 4\}$$

$$X2(n) = \{\underset{\uparrow}{1}, 1, 2, 2\}$$

b) State and prove any 3 properties of DFT. **[3]**

c) Write short note on ,"Overlap and save method" **[3]**

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Q3) a) What is the need of transform? What is the relationship between Z transform and DFT. **[3]**

b) Compute the Discrete Cosine Transform of following sequence. **[4]**

$x(n) = \{\underset{\uparrow}{4}, 3, 5, 1\}$

c) Compute the Z transform & define its ROC for the following signals. **[3]**

i) $x(n) = n^2\, u(n)$ for $n \geq 0$

ii) $x(n) = 2^{(n)}\, u(n-2)$

OR

Q4) a) By using partial fraction method find the Inverse Z transform of

$$X(z) = \frac{z^2}{(z+1)(z+2)}$$ **[3]**

b) Comment on, "Causality and stability", of LTI system in Z domain. **[3]**

c) Compute the DFT of the following sequence. **[4]**

$x(n) = \{\underset{\uparrow}{1}, 2, 3, 4\}$

Q5) a) Explain in detail impulse invariance technique for IIR filter design. What are it's drawbacks? How Bilinear Transformation method overcomes the limitations of impulse invariance technique. **[8]**

b) A digital filter has frequency specification pass band frequency is 0.2π and stop band frequency 0.3π. What are corresponding specification for pass band and stop band frequency in analog domain if, **[8]**

i) Impulse invariance technique is used for designing

ii) Bilinear Transformation method is used for designing.

OR

Q6) a) Obtain direct form I and II realization of a system described by **[8]**

$y(n) - 3/4\ y(n-1) - 1/2\ y(n-2) + 1/8\ y(n-3) = x(n) + 5/4\ x(n-1)$

b) Explain the steps used for designing an IIR filter using bilinear transformation method (BLT). What is warping effect in BLT. **[8]**

Q7) a) What is FIR system? Compare FIR system with an IIR system. **[8]**

b) Design FIR digital filter to approximate an ideal low pass filter with pass band gain of unity ,cut off frequency 850Hz and sampling frequency 5000Hz.The length of impulse response should be 5. Use a Rectangular Window. **[8]**

OR

Q8) a) Draw & explain the characteristics of ideal filters & its requirements. Why the ideal filters are used? Explain the Gibbs phenomenon, why it occurs? **[8]**

b) The desired response of FIR low pass filter is **[8]**

$$H_d(e^{jw}) = e^{-j3\omega} \qquad -\frac{3\pi}{4} \le w \le \frac{3\pi}{4}$$

$$= 0 \qquad \frac{3\pi}{4} \le w \le \pi$$

Determine H (e^{jw}) for M = 7 using Hamming window.

Q9) a) Explain applications of DSP in voice processing and image processing. **[8]**

b) Design two stage decimator for following specifications, **[10]**

Sampling rate of input signal = 20 KHz

Down sampler = 60,

Pass band = 0 to 40 Hz,

Transition band 40 to 50 Hz,

Pass band ripple = 0.01 dB, Stop band ripple = 0.002dB.

OR

Q10) a) Explain the importance features of TMS 320C67XX series DSP processor. Draw its Functional block diagram and explain each block in detail. **[10]**

b) Explain necessity of : **[8]**

i) MAC unit

ii) Barrel shifter

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