Total No. of Questions : 10]

SEAT No. :

P3155

[Total No. of Pages : 3

[4858] - 1044

T.E. (E & TC)

Digital Signal Processing

(2012 Pattern) (End Sem.) (Semester - I)

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

Instructions to the candidates:

1) ***Solve 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) ***Right side figures indicate marks.***

Q1) a) Explain Advantages of Digital Signal Processing over Analog Signal Processing. **[5]**

b) Explain the concept of orthogonality. **[2]**

c) Check whether the functions given are orthogonal or not over an time interval [0, 1] $f(t)=1; x(t)=\sqrt{3}(1-2t)$. **[3]**

OR

Q2) a) Compute 4-point DFT of sequence $x(n)=\{1 2 3 1\}$ using DIT-FFT radix-2 algorithm. What is Computational Complexity of Radix-2 FFT algorithm? **[6]**

b) Compute circular convolution of $x_1(n)=\{1,2,3,4\}$ & $x_2(n)=\{2,1,2,1\}$. **[4]**

Q3) a) Plot the magnitude and phase spectrum of the sampled data sequence = {2, 0, 0, 1} which was obtained using a sampling frequency of 20Hz. **[5]**

b) Explain linear filtering effect for long duration sequences. **[5]**

OR

Q4) a) State any four properties of Z-Transform. **[4]**

b) Compute Z-Transform and ROC of the following sequence

$$x(n)=\left[\frac{-1}{3}\right]^n u[n]-\left[\frac{1}{3}\right]^n u[-n-1].$$ **[6]**

P.T.O.

Q5) a) Design a Butterworth filter using impulse invariant method transformation to satisfy the following specifications. **[8]**

$$0.707 \le |H(e^{jw})| \le 1 \quad \text{for } 0 \le w \le 0.2\pi$$
$$|H(e^{jw})| \le 0.2 \quad \text{for } 0.6\pi \le w \le \pi$$

b) Explain impulse invariant method for S-plane to Z-plane mapping. Explain its limitations. **[8]**

OR

Q6) a) What is frequency warping effect? How the mapping is done in bilinear transformation method? **[7]**

b) Draw the direct form-I and II structures for the following systems. **[9]**

i) $3y(n) - 2y(n-1) + y(n-2) = 4x(n) - 3x(n-1) + 2x(n-2)$.

ii) $y(n) = 0.5[x(n) + x(n-1)]$.

Q7) a) Explain the characteristics of the FIR filters. **[8]**

b) Determine the impulse response $h(n)$ of a filter having desired frequency response.

$$H_d(e^{jw}) = \begin{cases} e^{-j(N-1)w} & \text{for } 0 \le |w| \le \dfrac{\pi}{2} \\ 0 & \text{for } \dfrac{\pi}{2} \le |w| \le \pi \end{cases}$$

N = 7, use windowing technique approach. Use hamming window. **[8]**

OR

Q8) a) Explain Gibbs phenomenon. Compare between windows available. **[8]**

b) Determine the impulse response $h(n)$ of a filter having desired frequency response,

$$H_d(e^{jw}) = \begin{cases} e^{-j(N-1)w} & \text{for } 0 \le |w| \le \dfrac{\pi}{2} \\ 0 & \text{for } \dfrac{\pi}{2} \le |w| \le \pi \end{cases}$$

N = 7, Use frequency sampling approach. **[8]**

Q9) a) What is sampling rate conversion? What is multirate DSP? Why it is required? **[6]**

b) A signal $x(n)$, at a sampling frequency of 2.048 KHz is to be decimated by a factor of 32 to yield a signal at sampling frequency 64Hz. The signal band of interest extends from 0-30 Hz. The anti-aliasing filter should satisfy the following specifications:

Pass Band deviation : 0.01dB

Stop Band deviation : 80dB

Pass Band : 0 – 30Hz

Stop Band : 32 – 64Hz

The signal components in the range from 30 to 32 Hz should be protected from aliasing. Design a suitable one-stage decimator. **[8]**

c) How the DSP processors are selected? (any four points) **[4]**

OR

Q10) a) State four important features of DSP processors. **[4]**

b) Draw the architecture of typical DSP processor TMS320C67XX and explain it in short. **[6]**

c) 5 Design a two stage decimator that down samples an audio signal by a factor of 30, satisfying the following constraints: **[8]**

Input sampling frequency : 240 KHz

Highest frequency of interest : 3.4 KHz

Passband ripple : 0.05

Stopband attenuation : 0.01