Total No. of Questions : 10]

SEAT No. :

P4340

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T.E (E&TC) (End Semester)

DIGITAL SIGNAL PROCESSING

(2015 Pattern)

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) ***Your answers will be valued as a whole.***
4) ***Use of logarithmic tables slide rule, Mollier charts, electronic pocket calculator and steam tables is allowed.***
5) ***Assume suitable data, if necessary.***

Q1) a) An analog signal given as $x_a(t) = 15\cos(1250\pi t) + 17\cos(2170\pi t) + 33\cos(4750\pi t)$ is converted into discrete time signal. Determine Nyquist sampling rate, Folding frequency, resulting discrete time signal $x(n)$ if sampling frequency is 625 Hz. Also write discrete time frequencies in radians. **[5]**

b) An LTI system is defined by difference equation
$y(n) = y(n-1) + y(n-2) + x(n-1)$. Find system function H(z). Draw pole zero diagram. Find out h(n) for causal, non-causal systems, if not why? **[5]**

OR

Q2) a) Find the DFT of the sequence

$$x(n) = \begin{cases} 1 & \text{for } 0 \le n \le 2 \\ 0 & \text{otherwise} \end{cases}$$

for N = 4. Find $|X(K)|$ and $\angle X(K)$ **[5]**

b) Explain the sampling theorem and advantages of Digital over Analog Signal Processing. **[5]**

Q3) a) State any four properties of Z transform. **[4]**

b) Compare circular convolution with linear convolution find the circular convolution of two finite duration sequences. **[6]**

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

$x_2(n) = \{1, 2, 3\}$

OR

Q4) a) What is FFT? Explain Bit-reversal and In place computation concepts in FFT algorithm. Show the 3-bit bit reversed sequence. **[5]**

b) Explain the concept of orthogonality. Check whether the functions given are orthogonal or not over an time interval [0,1], $f(t) = 1$, $x(t) = \sqrt{3}(1-2t)$. **[5]**

Q5) a) Design the second order low pass Digital Butterworth filter with cut off frequency of 1 KHz and sampling frequency 10,000 samples/sec by Bilinear transformation. **[9]**

b) Write the equation, Draw & compare the characteristics of Butterworth filter, Chebyshev filters and elliptic filter. **[9]**

OR

Q6) a) What is Bilinear transformation? Explain the properties of BLT. What is worping effect? How do you take care of it in design. **[9]**

b) State the advantage of direct form II realization over direct form I. Hence implement the following difference equation in direct form I and II. **[9]**

$$y(n) + 0.1\, y(n-1) + 0.72\, y(n-2) = 0.7\, x(n) - 0.95\, x(n-2)$$

Q7) a) Design an FIR filter having desired frequency response as given below using rectangular window

$$\text{Hd(w)}\begin{cases} 1 & |w| \le \pi/4 \\ 0 & \pi/4 \le |w| \le \pi \end{cases} \quad \& \quad w(n) = \begin{cases} 1 & |n| < 2 \\ 0 & \text{otherwise} \end{cases}$$

Find H(w). Does the filter is realizable. Justify your answer. What modification is required in Hd(w) to make it realizable. **[10]**

b) Explain frequency sampling technique of FIR filter designing in detail. **[6]**

OR

Q8) a) Explain windowing technique of FIR filter design in detail. Also explain Gibb's phenomena and how it can be reduced. State different types of windows used with their window function. **[10]**

b) What is the meaning of linear phase. Prove that FIR filters are inherently stable. **[6]**

Q9) a) Speech signal is corrupted by low and high frequency noise. Explain in detail how DSP is used to remove noise with illustration. **[8]**

b) Explain the application of DSP in vibration signature analysis for defective gear teeth. **[8]**

OR

Q10) a) Explain speech coding and compression technique. How signal processing techniques are used in this. **[8]**

b) Explain how DSP is useful in Interference cancellation in ECG. **[8]**

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