Total No. of Questions : 8]

SEAT No. :

PA-1488

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T.E. (Electronics & Telecommunication Engineering)
DIGITAL SIGNAL PROCESSING
(2019 Pattern) (Semester - I) (Eelective - I) (304185(A))

Time : 2½ Hours] *[Max. Marks : 70*

Instructions to the candidates:

1) *Attempt Q.1 or Q.2, Q.3 or Q.4, Q.5 or Q.6, Q.7 or Q.8.*
2) *Neat diagrams must be drawn wherever necessary.*
3) *Figures to the right indicate full marks.*
4) *Assume suitable data if necessarv.*

Q1) a) Find the response of a linear filter with impulse response $h_{(n)} = \{1,2,4\}$ to the input sequence $x_{(n)} = \{1,2\}$ using linear convolution computed through circular convolution. **[8]**

b) Find N = 5 point DFT for $x_{(n)} = \{1,0,1,0,1\}$? **[8]**

c) Explain the linear filtering using overlap save and overlap add methods? **[2]**

OR

Q2) a) Find linear convolution using overlap add method of the following sequence $x(n)$ and $h(n)$ **[8]**

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

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

b) Find the 8-point DFT of sequence **[10]**

x(n) = {1,2,3,4,4,3,2,1} using DIT radix - 2 FFT algorithm.

Q3) a) Convert the analog filter with system function $H_s(s) = \dfrac{s+0.2}{(s+0.2)^2+9}$ into a digital IIR filter by means of impulse invariant technique Assume T = 1 sec. **[10]**

b) Explain the concept of filter design. Elaborate the advantages and disadvantages of digital filters? **[8]**

OR

P.T.O.

Q4) a) The system transfer function of analog filter is given by, $H(s)=\dfrac{S+0.1}{(S+0.1)^2+16}$, obtain the system transfer function of digital filter using bilinear transformation method (BLT) which is resonant at $w_r=\dfrac{\pi}{2}$. **[10]**

b) A digital filter has frequency specification as : **[8]**

Passband frequency = w_p = 0.2π

Stopband frequency = w_s = 0.3π

What are the corresponding specification for passband and stopband frequencies in analog domain if

i) Impulse invariance technique is used for designing

ii) Bilinear transformation is used for desiging assume sampling time Ts = 1 sec.

Q5) a) Elaborate on the ideal filter requirements interms of causality and its implications? **[8]**

b) List out all the windowing techniques? Describe any three with its mathematical formulas characteristics and compare them. **[9]**

OR

Q6) a) Design linear phase FIR Lowpass filter using Hanning Window technique for the frequency characteristics of the filter given by **[9]**

$$Hd(w) = e^{-j3w} \text{ for } \frac{-\pi}{4} \le w \le \frac{\pi}{4}$$

$$= 0 \quad \text{otherwise}$$

b) Obtain the coefficients of FIR lowpass filter to meet the specification given below. Use kaiserwindow **[8]**

Panband edge frequency = 1.5 KHz.

Transsion width = 0.5 KHz.

Stopband alternation ≥ 50dB

Sampling frequency = 8 KHz.

Q7) a) Speech signal is corrupted by low and high frequency noise. Explain indetail how DSP is used to remove noise with illustration. **[8]**

b) Explain how DSP is useful in interference cancellation in ECG. **[9]**

OR

Q8) a) Explain speech coding and compression technique. How signal processing techniques are used in this. **[8]**

b) Explain the application of DSP in vibration signature analysis for defective gear teeth. **[9]**

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