Total No. of Questions : 10]

SEAT No. :

P3361

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T.E. (E & TC)

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) A contineous time signal x(t) with fundamental period T = 1/F is sampled at rate $F_s = 1/T_s$ to produce discrete time sinusoid $x(n) = x(nT_s)$. Show that x(n) is periodic if

$$\frac{T_s}{T} = \frac{K}{N}$$

where K and N are integers. **[5]**

b) Define Z transform & ROC. Hence clearly mention & draw ROCs for causal, non-causal, finite and infinite duration sequences. **[5]**

OR

Q2) a) The impulse response of a system is given by $h(n) = 2(0.5)^n u(n)$. Find the system function H(z) and the difference equation for y(n). Compute y(n) when $x(n) = (¼)^n u(n)$ using Z.T. **[5]**

b) Derive the relationship between Fourier transform and Z-transform. **[5]**

P.T.O.

Q3) a) Define DFT and IDFT. Hence using the frequency shift property show that DFT $\{x(n)\cos\frac{2\pi nk}{N}\} = \frac{1}{2}[X(m-k)+X(m+k)]$ **[5]**

b) Find the causal sequence x(n) for

$X(Z)=[6+Z^{-1}]/[(1+0.25Z^{-1})(1+0.5Z^{-1})]$ **[5]**

OR

Q4) a) An analog signal **[5]**

$x_a(t) = \sin(480\pi t) + 3\sin(720\pi t)$ is sampled at the rate 600 samples per sec.

i) Determine Nyquist rate

ii) Determine folding $freq^n$

iii) Find x(n)

iv) What the $freq^n$ in rad in x(n)

b) State & prove periodicity & time shift property of DFT. **[5]**

Q5) a) Explain step by step the Impulse Invariance transformation, its use to design IIRfilter and its drawback. **[8]**

b) Why the transformations are used to convert analog filter into digital filter. Hence convert the analog filter with system function

$$H_a(s)=\frac{2}{(s+1)(s+2)}$$

into digital filter by means of

i) Impulse Invariance Transformation.

ii) Bilinear Transformation with $T=1$. **[10]**

OR

Q6) a) Implement the second order digital filter using Direct form I & Direct form II for the following difference equation. **[8]**

$$y(n) = 2r\cos(w_o n)\,y(n-1) - r^2\,y(n-2) + x(n) - r\cos(w_o n)\,x(n-1)$$

b) Design a digital IIR filter with following specifications. **[10]**

$$0.6 \leq |H(e^{jw})| \leq 1 \qquad 0 \leq w \leq 0.35\pi$$
$$|H(e^{jw})| \leq 0.1 \qquad 0.7\pi \leq w \leq \pi$$

with T = 0.1 sec. and using BLT.

Q7) a) Design a low pass filter with $H_d(w)$ as

$$H_d(w) = \begin{cases} e^{-j2w} & |w| \leq \pi|4 \\ 0 & \frac{\pi}{4} \leq w \leq \pi \end{cases}$$ **[10]**

with Hann window , with N = 5 find H(w) equation.

b) What are the characteristics of FIR filter. Hence prove that FIR filter are inherently stable filter. **[6]**

OR

Q8) a) Draw & explain the characteristics of ideal filters & its requirements. Why the ideal filters are not realizable. Explain the Gibbs phenomenon & why it occurs? **[10]**

b) What is Finite word length effect & how it affects the FIR filter performance. **[6]**

Q9) a) Explain the application of DSP in compact Disc recording system. **[8]**

b) Explain real time application of DSP in medical field. **[8]**

OR

Q10) a) How the DSP is useful in speech processing. Explain any application of speech processing using DSP. **[8]**

b) How the mechanical industry is benefited with DSP algorithm, explain with example. **[8]**