

Total No. of Questions : 10]

SEAT No. :

P3361

[Total No. of Pages : 3

[5353] - 552

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 continuous 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 $x(n) = (1/4)^n u(n)$ 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

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

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 freqⁿ

iii) Find $x(n)$

iv) What the freqⁿ 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 IIR filter 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]

$$y(n) = 2r \cos(\omega_0 n) y(n-1) - r^2 y(n-2) + x(n) - r \cos(\omega_0 n) x(n-1)$$

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

$$\begin{aligned} 0.6 \leq |H(e^{jw})| \leq 1 & \quad 0 \leq w \leq 0.35\pi \\ |H(e^{jw})| \leq 0.1 & \quad 0.7\pi \leq w \leq \pi \end{aligned}$$

with $T = 0.1$ sec. and using BLT.

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

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

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]

