Total No. of Questions : 10]

SEAT No. :

P2953

[Total No. of Pages : 4

[5669]-542

T.E. (E & TC)

DIGITAL SIGNAL PROCESSING

(2015 Pattern)

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, & Q.9 or Q.10.*
2) *Draw suitable diagrams where necessary.*
3) *Figures to the right indicate full marks.*

Q1) a) Show relationship between analog frequencies and digital frequencies. **[4]**

b) An analog signal contains frequencies upto 10kHz **[6]**

i) What is range of sampling frequencies must be selected for faithful reconstruction of signal?

ii) If signal is sampled with sampling frequency 8kHz what is folding frequency?

iii) Does aliasing occours if i/p is 5kHz? if yes calculate alised frequencies from original frequencies

(Consider sampling freq to be 8k)

Does aliasing occours at i/p frequency if i/p is 9kHz? if yes calculate aliased.

OR

Q2) a) Consider signal $x(t)$ = 5 sin (500 mt) if signal is sampled at Fs = 1500 Hz **[4]**

i) What is DT signal obtained after sampling?

ii) Find frequency of DT signal.

iii) Find DT signal for sampling frequency Fs = 300

b) State & Prove following properties in Z transform **[6]**

i) Scaling property.

ii) Time shift property.

P.T.O.

Q3) a) Determine IZT of following functions. **[8]**

i) $X(z) = \dfrac{(8z-19)}{(z-2)(z-3)}$

$x(n)$ is causal

ii) $\dfrac{z^3 + z^2}{(z-1)(z-3)} = x(z)$ & ROC $|z| > 3$

b) Obtain z transform of following

$x(n) = (0.5)^n\, u(n) + (-0.2)^n\, u(n-3)$ **[2]**

OR

Q4) a) Compute IDFT by matrix method **[4]**

$X(k) = \{10, -2 + 2j, -2, -2 -2j\}$

b) Compute 8 point DFT of sequence **[6]**

$x(n) = \{0,1\ 2, 3\}$ & draw magnitude & phase plot

Q5) a) Compare IIR & Fir filters on following points. **[6]**

i) Filter governing mathematical equation

ii) Memory requirment

iii) Stability

iv) Recursiveness

v) Phase response

vi) Processing time

b) What is warping effect in bilinear transform? What is it's effect on magnitude & phase response? **[4]**

c) Convert analog filter with system function H(s) into digital IIR filter using impulse invariance method **[6]**

$H(s) = \dfrac{10}{S^2 + 75 + 10}$

OR

Q6) a) Compare butterworth filter & chebyshev filter on following points. **[4]**

i) Frequency response

ii) Order for given specification

iii) Transition band

iv) Phase response & pole location

b) Design Butterworth filter for following specification. **[8]**

$$0.8 \leq |H_a(S)| \leq 1 \quad 0 \leq F \leq 1000Hz$$
$$|H_a(s)| \leq 0.2 \quad F \geq 5000Hz$$

c) Draw direct form 2 realization for the following **[4]**

$$H(z) = \frac{1+3z^{-1}}{2+z^{-1}-4z^{-3}}$$

Q7) a) Explain following window functions **[6]**

i) Rectangular window

ii) Hamming window

iii) Hanning window

b) Design linear phase FIR filter using hamming window with cutoff freq.

0.2 rad/sec and 0.3 rad/sec

use M = 7

Calculate 1st two filter coefficients only (n=0 & n=1) **[10]**

OR

Q8) a) What is gibbs phenomenon? How the effect of gibbs phenomenon is reduced? **[4]**

b) Using frequency sampling method design law pass FIR filter to meet following specification.

Pass band - 0-5kHz Filter length = 9

Sampling frequency → 18kHz

Obtain h(n) for n = 0.1 only **[12]**

Q9) a) Write a note on digital cross over audio system. **[6]**

b) With the help of block diagram explain enhancement of ECG signal for heart rate detection. **[6]**

c) Explain speech compression & decompression with block diagram. **[6]**

OR

Q10) a) Explain compact disc recording system. **[6]**

b) Compare digital signal processing & analog s/g processing on following points.

→ Accuracy & component tolerance

→ Cost

→ Upgradation adaptation

→ Implementation

→ Repeatability

→ Versatility **[6]**

c) Explain how the defective gear tooth can be identified using vibration analysis? **[6]**

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