Total No of Questions: [12] **SEAT NO. :**

[Total No. of Pages : 3]

B.E. 2008 (Digital Signal Processing)

[403149]

(Elective - III) (Semester - II)

Time: 3 Hours **Max. Marks : 100**

Instructions to the candidates:

1) Answers to the two sections should be written in separate answer books.
2) Answer any three questions from each section.
3) Neat diagrams must be drawn wherever necessary.
4) Figures to the right side indicate full marks.
5) Use of Calculator is allowed.
6) Assume Suitable data if necessary

SECTION I

Q1) a) State sampling theorem and explain aliasing effect. [6]

b) A discrete time signal is given by [6]

$$x(n)=\{1,\underset{\uparrow}{1},1,1,2\}$$

Sketch the following signals i) $x(n-2)$ ii) $x(3-n)$ iii) $x(n-1)\delta(n-1)$

c) Find linear convolution using multiplication method [6]

$x(n) = 1$ for $n=-3$ & $h(n)= 1$ for $n=2$ & 3

$= 2$ for $n= -2$ $= 0$ elsewhere

0 elsewhere

OR

Q2) a) State advantages of digital signals over analog signals [6]

b) Determine whether following systems are static/dynamic, linear/nonlinear, causal/noncausal, and stable/unstable. [6]

i) $y(n) = x(n)+ nx(n+1)$ ii) $y(n)= e^{-x(n)}$ iii) $y(n)= \cos(x(n))$

c) Explain A/D conversion with quantization and encoding. [6]

Q3) a) State and prove any four properties of z transform. [8]

b) Find z transform and ROC of [8]

$$x(n)=\left[\frac{1}{2}^{n}+\frac{3}{4}^{n}\right]u(n-1)$$

OR

Q4) a) State and prove any four properties of Discrete time fourier transform . [8]

b) Find inverse z transform using partial fraction method [8]

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

Q5) a) Explain different Ideal selective filters and derive their impulse response. [8]

b) Explain generalized linear phase system and four types of GLPS [8]

OR

Q6) a) Explain causality and stability with respect to z transform. [8]

b) Plot the frequency response (magnitude and phase response) of [8]

$$y(n) - 0.5\, y(n-1) = x(n)$$

SECTION II

Q7) a) State the Circular Shift property of Discrete Fourier Transform (DFT) [4]

b) Obtain DFT of the following sequence: [6]

$x(n) = \{ 0.5, 0.5, 0.5, 0.5\}$ using Decimation-in-Time (DIT) Radix-2 FFT algorithm.

c) Determine the circular convolution of the following sequences using 4-point DFT and IDFT [6]

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

$$x_2(n) = \{4,3,2,2\}$$

OR

Q8) a) Compare DFT with DTFT [4]

b) Compute the IDFT of the following using Decimation-in- Frequency (DIF) Radix-2 FFT algorithm. [6]

$$X(k) = \{ 8, -2-j2, 4, -2+j2\}$$

c) Find Linear convolution using circular convolution of the following Sequences: [6]

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

Q9) a) State the specifications required in any filter design. [4]

b) Differentiate between Bilinear Transformation method & Impulse invariance method. [6]

c) Design a low pass digital IIR filter of Butterworth type using bilinear transformation method (BLT) for the specifications given below: [8]

i) Amount of gain required in pass band=0.8

ii) Pass band frequency=0.2π

iii) Amount of attenuation required= 0.2

iv) Stop band frequency=$0.6\,\pi$. Assume Sampling time ,T= 1.

OR

Q10) a) Differentiate between FIR filters and IIR filters. [4]

b) State & define various kinds of window functions used for FIR filter design. [6]

c) Design a symmetric low pass digital FIR filter for the following specifications using Fourier series method:- [8]

Cut-off frequency = 500 Hz

Sampling frequency = 2000 Hz

Order of filter = 10

Filter length required= 11; Use Hamming Window.

Take $h(n) = \dfrac{\sin(n\pi/2)}{2(n\pi/2)}$

Q11) a) Realize the system described by – [8]

y (n) - 0.8 y(n-1) + 0.12 y(n-2)=5x(n) + 2x(n-1)

in Direct form-II & cascade form for IIR filters.

b) Write short notes on (**Any two**) [8]

i) Spectrum Analysis ii) DSP based vibration analysis system

iii) Finite Register Length effect.

OR

Q12) a) Determine FIR linear phase and cascade realization of the system function expressed as [8]

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

b) Write short notes on (**Any two**) [8]

i) Power factor correction using DSP ii) Applications of DSP in Machine control iii) Basic structures for IIR Filters.