Name : ....

Roll No. : ....

Invigilator's Signature : ....

CS/B.Tech (IT)(N)/SEM-5/IT-504C/2012-13

2012

DIGITAL SIGNAL PROCESSING

Time Allotted : 3 Hours

Full Marks : 70

The figures in the margin indicate full marks.

Candidates are required to give their answers in their own words as far as practicable.

GROUP – A
(Multiple Choice Type Questions)

1. Choose the correct alternatives for the following :   10 × 1 = 10

    i) The linear convolution of x (n) and h (n) results in which of the following numbers of samples of y (n) where L and M are the lengths of x (n) and h (n) respectively ?

    a) $L - M + 1$

    b) $L + M - 1$

    c) $\max (L, M)$

    d)   None of these.

    ii) Zero padding technique is used for converting

    a)   *Z*-transform to Fourier transform

    b)   DFT to *Z*-transform

    c)   linear convolution to circular convolution

    d)   circular convolution to linear convolution.

    iii) The output of a causal system

    a)   does not depend on the inputs

    b)   depends on the present and past inputs

    c)   depends on the present and future inputs

    d)   depends on the past and future inputs.

iv) Which of the following is incorrect ?

   a) ROC contains N number of poles

   b) ROC is a ring in the z-plane centered at the origin

   c) ROC of an LTI stable system contains the unit circle

   d) ROC must be connected region.

v) If $x(n)$ is a finite duration two sided sequence then the ROC is entire z-plane except at

   a) $z = 0$ b) $z = \infty$

   c) $z = 0$ and $z = r$ d) $z = 0$ and $z = \infty$.

vi) The linear convolution of $x(n)$ and $h(n)$ where $x(n) = h(n) = \{1, 2, -1\}$ is

   a) $\{1, 4, 2, -3, 1, 8\}$ b) $\{1, 2, -1\}$

   c) $\{1, 4, 2, -3, 1\}$ d) none of these.

vii) In discrete Fourier transform the range of the summation index n ranges from

   a) 0 to N b) 0 to $N - 1$

   c) 0 to $N + 1$ d) $-\infty$ to $+\infty$.

viii) The impulse response of FIR filters is computed for

   a) finite number of samples

   b) infinite number of samples

   c) depending on the value of $H(z)$

   d) all of these.

ix) What condition on the FIR sequence is to be imposed in order that the filter can be called as a linear phase filter ?

   a) Symmetric condition $h(n) = h(N-1-n)$

   b) Anti-symmetric condition $h(n) = -h(N-1-n)$

   c) Both (a) and (b)

   d) None of these.

x) The Fourier transform of a unit step signal $u(t)$ is

    a) 1   b) $1/j\omega$

    c) $\pi\delta(\omega)+1/j\omega$ d) none of these.

GROUP – B

(Short Answer Type Questions)

Answer any *three* of the following $3 \times 5 = 15$

2. Define FIR and IIR filters with proper examples.

3. Find the *Z*-transform and ROC of $x(n)=r^n \cos(\omega_0 n).u(n)$. State whether the system is stable or not.

4. Find the circular convolution of $x(n)=\{2,0,0,1\}$ and $h(n)=\{4,3,2,1\}$ using DFT and IDFT.

5. Differentiate between any two of the following :

    a) Overlap add method and Overlap save method

    b) FIR and IIR filters

    c) DIT FFT and DIF FFT Algorithm.

6. Draw the Direct Form I and Direct Form II realization for the system described by the difference equation :

$$y(n)-0\cdot75(n-1)+0\cdot11y(n-2)=x(n)-0\cdot5x(n-1)$$

GROUP – C

(Long Answer Type Questions)

Answer any *three* of the following. $3 \times 15 = 45$

7. Find the inverse *Z*-transform of

$$X(z)=\frac{(z^2+z)}{(z-1)(z-3)} \; ; \; \text{ROC} : |z|>|3| \text{ using}$$

    i) PFE method

    ii) Residue method

    iii) Convolution method.   5 + 5 + 5

8. a) Derive the butterfly structure for computing 8 pt. DFT using DIF FFT Algorithm.

   b) For the given analog signal, determine the rate at which the signal must be sampled to avoid aliasing. Also obtain the discrete time signal.

   $$x(t) = 4\cos 50\pi t + 8\sin 300\pi t - \cos 100\pi t \qquad 10 + 5$$

9. Design an ideal High-pass filter with a frequency response

   $$H_d(e^{j\omega}) = 1, \pi/4 \leq |\omega| \leq \pi$$

   $$0, |\omega| < \pi/4$$

   using Rectangular window. Also find the values of $h(n)$ for $N = 11$. Find $H(z)$.

10. a) Determine the digital filter for the given analog filter using Impulse Invariance method, for

    $$H(s) = \frac{(s+a)}{(s+a)^2 + b^2}.$$

    b) What is Warping effect ? What is its effect on magnitude and phase response ? $\qquad$ 10 + 5

11. a) What is the need for employing window technique for FIR Filter design ? What are the desirable characteristics of a Window ? Explain the various types of windows.

    b) Explain Gibbs phenomenon. $\qquad$ 12 + 3

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