RM-7071/600

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B. E. III (Sem. VI) (Computer) Examination
May / June – 2010
Digital Signal Processing

Time : **3** Hours] [Total Marks : **100**

Instruction :

(1)

નીચે દર્શાવેલ ☛ નિશાનીવાળી વિગતો ઉત્તરવહી પર અવશ્ય લખવી.
Fillup strictly the details of ☛ signs on your answer book.

Name of the Examination :
☛ **B. E. 3 (Sem. 6) (Computer)**

Name of the Subject :
☛ **Digital Signal Processing**

☛ Subject Code No. : **7 0 7 1** ☛ Section No. (1, 2,.....) : **1&2**

Seat No. :

Student's Signature

(2) Answer the **two** sections in **separate** answer books.
(3) Figures to the **right** indicate full marks.
(4) Assume suitale data whenever **necessary**.
(5) Acronyms carry their usual meaning.
(6) Draw the figures wherever needed.
(7) Use of scientific calculator CASIO fx - 82/83, Fx - 100 or equivalent of other companies is allowed.

SECTION-I

Q-1(a). Do as directed (10)

1. Define Deterministic signal and Random signal
2. Define multichannel signal
3. Determine linear convolution of $x_1(n) = u(n)$ and $x_2(n) = \delta(n)$ by means of z-transform.
4. For the system given as $y(n) = y(n-1) + x(n)$, is this system is recursive? What is the impulse response of the system.
5. Define zero input and zero state response.

Q.1(b)

(i) From the following system classify the system according to its linearity, casility, stability (5)

1. $y(n)=|x(n)|$
2. $y(n) = x(n)u(n)$
3. $y(n) = ax(n) + b$

(ii) Find the autocorrelation of signal $x(n) = a^n u(n)$, $|a| < 1$ using z-transform (5)

Q-2(a) Explain the properties of LTI system (8)

(b) By differentiating X(z) and then using appropriate properties of the z-transform, determine $x(n)$ for the following transforms (7)

1. $X(z) = \log(1 - 2z), |z| < 0.5$
2. $X(z) = \log(1 - 0.5z^{-1}), |z| > 0.5$

OR

Q-2(a) Explain any three properties of z-transfom (7)

(b) Consider the analog signal $x_a(t) = 5\ cos\ 50\pi t + 10\ sin300\pi t - cos\ 100\pi t$ (8)

(1) What is the range of sampling frequencies allows exact reconstruction of this from its samples?

(2) Suppose that the signal is sampled at the rate F_S = 350Hz, What is the discrete time signal obtained after sampling? Find out the period of discrete time signal.

(3) Suppose that the signal is sampled at the rate F_S = 150Hz, What is the discrete time signal obtained after sampling? Find out the period of discrete time signal.

(4) Suppose that the signal is sampled at the rate F_S = 75Hz, What is the discrete time signal obtained after sampling? Find out the period of discrete time signal.

(5) Suppose that the signal is sampled at the rate F_S = 35Hz, What is the discrete time signal obtained after sampling? Find out the period of discrete time signal.

Q-3 Attempt any three (15)

(a) What is power spectral density? Obtain Parseval's theorem for periodic signal.

(b) Explain residue method to evaluate inverse z- transform.

(c) Determine the energy spectral density of signal $x(n) = \left(\frac{1}{2}\right)^n u(n)$. Find the total energy constrained by $x(n)$

(d) State the advantages and limitation of Digital signal processing over analog signal processing

(e) Define causal and non causal system. Determine causality condition in LTI system.

SECTION-II

Q-4(a). Do as directed (10)

1. For antisymmetric FIR filter is not suitable for lowpass filter (True/False). Give reason.
2. Find 4-point DFT of a sequence $x(n) = u(n) - u(n-3)$
3. Write circular frequency shift property
4. Why we need FFT algorithms to find DFT?
5. In which way we can find linear convolution using circular convolution.

Q.4(b)

(i) Explain why causal and stable IIR filter cannot be linear phase. (3)

(ii) Consider the sequence $x(n)$ of length 6, given by $x(n) = \{\underset{\uparrow}{0},1,2,3,4,5\}$ (7)

Sample its DTFT at four equally spaced points $\omega_k = \frac{2\pi k}{4}$, $k = 0,1,2,3$. Apply four point IDFT to obtain sequence back. Comment on the result

Q-5(a) Develop 8-point radix-2 FFT algorithm using DIT approach. (8)

(b) Consider the analog resonator having system function (7)

$$H(s) = \frac{s+0.1}{s^2+0.2s+9.01}$$

Convert into digital IIR filter using bilinear transformation. The digital filter is to have a resonant frequency of $\omega_r = \pi/2$.

OR

Q-5(a) Explain divide and conquer approach to find DFT. (7)

(b) Determine circular convolution of $x_1(n) = \{\underset{\uparrow}{3},2,1,4\}$ and $x_2(n)=\{\underset{\uparrow}{1},2,1,2\}$ (8)

Using DFT-IDFT method.

Q-6 Attempt any three : (15)

(a) Explain overlap-save method to find linear filtering of long data sequence.

(b) Explain Paley-Wiener theorem, and obtain the relationship between the real part and imaginary part of practically realizable filter having transfer function $H(\omega)$.

(c) Design the digital FIR linear phase lowpass filter having cut-off frequency is $\pi/4$ using rectangular window for M=7 (filter length = 7).

(d) Explain impulse invariance method for designing digital IIR filter.

(e) Consider the finite length sequence $x(n) = \delta(n) + \delta(n-5)$. Find its 10 point DFT. Find the sequence $y(n)$ that has DFT given by $e^{\frac{-j2k2\pi}{10}}X(k)$.