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Roll No. ....

603(O) & EI–603(N)

B. E. (Sixth Semester) EXAMINATION, June, 2010

(Old & New Scheme)

(Common for EC, EI & IT Engg.)

DIGITAL SIGNAL PROCESSING

Time : Three Hours

Maximum Marks : 100

Minimum Pass Marks : 35

Note : Attempt any *five* questions. Each question carries equal marks. Notations have standard meaning.

1. (a) Discuss the properties of Discrete Fourier transforms. Also discuss DTFT sampling.

   (b) Find the N-point DFT of the sequence :

   $$x(n) = \cos(n\,\omega_0)\ \ 0 \le n \le N-1$$

   Compare the values of the DFT coefficients X (k) when $\omega_0 = 2\pi k_0/N$ to those when $\omega_0 \neq 2\pi k_0/N$. Explain the difference.

2. (a) Consider the finite length sequence :

   $$x(n) = \delta(n) + 2\,\delta(n-5)$$

   Find the 10-point discrete Fourier transform.

   (b) Determine the DFT of the sequence :

   $$x(n) = \begin{cases} \frac{1}{4}, & \text{for } 0 \le n \le 2 \\ 0, & \text{otherwise} \end{cases}$$

P. T. O.

3. (a) Discuss the signal flow graph implementation of the structure of the FIR systems in direct form.

   (b) Discuss the transposed structures of the filters in Direct I and Direct II form.

4. With suitable signal flow graph describe the algorithm of Decimation in frequency algorithm of fast Fourier transform.

5. (a) Discuss the window method of design of FIR filters.

   (b) A low pass filter is to be designed with the following desired frequency response :

$$H d\left(e^{j\omega}\right) = \begin{cases} e^{-j2\omega}, & -\dfrac{\pi}{4} \leq \omega \leq \dfrac{\pi}{4} \\ 0 \quad , & \dfrac{\pi}{4} < |\omega| \leq \pi \end{cases}$$

   Determine filter coefficients if the window function is defined as follows :

$$\omega(n) = \begin{cases} 1, & 0 \leq n \leq 4 \\ 0, & \text{otherwise} \end{cases}$$

6. (a) Discuss basic steps of design of IIR filters.

   (b) By impulse invariant technique convert the following analog filter into a digital filter whose system function :

$$H(s) = \frac{s + 0{\cdot}2}{(s + 0{\cdot}2)^2 + 9}$$

   Assume T = 1 sec.

7. (a) Using bilinear transformation obtain H (z) if :

$$H(s) = \frac{1}{(s+1)^2} \qquad T = 0{\cdot}1 \text{ sec.}$$

   (b) Use the backward difference for derivative to convert the analog low pass filter with system function :

$$H(s) = \frac{1}{s+2}$$

8. Write short notes on any *two* of the following :

   (i) Energy density spectrum

   (ii) Estimation of power spectrum of random signals

   (iii) Role of DFT in spectral estimation

   (iv) Basic AR, MA, ARMA models