Code No.: 6329

FACULTY OF ENGINEERING

B.E. III/IV Year (EE/INST) II Semester (Main) Examination, June 2010

DIGITAL SIGNAL PROCESSING

Time : 3 Hours] [Max. Marks : 75

*Answer **all** questions from Part A.*
*Answer any **five** questions from Part B.*

Part A – (Marks :25)

1. Give the names of the manufacturers of Digital Signal Processors? (2)
2. The potential applications of DSP in Instrumentation /control are : (2)
3. Define time–shifting and time reversal of a Discrete time sequence. (2)
4. A discrete time system can be (i) static or dynamic (ii) linear or non–linear. Examine the following relations with respect to the above properties : (3)
   (i) $y(n) = cos(x(n))$
   (ii) $y(n) = x(-n + 2)$.
5. What is periodic convolution? (3)
6. Obtain the time–sequences from its DFT coefficients $\{2.0,\ 1+j,\ 0, 1-j\}$ with $N = 4$. (3)
7. Find the z–transform of $x(k) = k^2$. (3)
8. What are the advantages of active filters over the passive filters? (2)
9. List the advantages of digital filter (at least four). (2)
10. Write the procedure steps for computing the IIR filter coefficients. (3)

Part B – (Marks : 5 × 10 = 50)

11. (a) Let $e(n)$ be an exponential sequence, $e(n) = \alpha^n$ for all n and let $x(n)$ and $y(n)$ denote two arbitrary sequences. Show that.
$$[e(n).x(n)] * [e(n).\ y(n)] = e(n).[x(n) * y(n)]$$ (5)

(b) Determine whether or not the signals below periodic and, for each signal that is periodic, determine the fundamental period. (5)

(i) $x(n) = cos(0.125\pi n)$ (ii) $x(n) = sin(\pi + 0.2n)$ (iii) $x(n) = e^{\frac{jn\pi}{16}} cos(n\pi/17)$.

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12. (a) $H(z) = \frac{1-z^{-1}}{1+2z^{-1}-3z^{-2}}$, Realize the filter $H(z)$ in direct form – II realization. (5)

(b) Explain the methods to find the inverse z–transform. (5)

13. (a) State and explain at least five important properties of DFT. (4)

(b) Compute the *DFT* of each of the following finite–length sequences considered to be of length N. (6)

(i) $x(n) = \delta(n)$ (ii) $x(n) = \delta(n - n_0)$, Where $0 < n_0 < N$ (iii) $x(n) = a^n, 0 \le n \le N-1$.

14. (a) Develop an 8-point *FFT* algorithm using decimation in time and draw its complete flow graph. (5)

(b) Calculate the time sequence $x(n)$ for given *DFT* components $\{2, 1+j, 0, 1-j\}$.(5)

15. (a) What are popular window functions used for computing the coefficients of FIR filters. Explain them in detail. (5)

(b) It is required to design a digital filter to approximate the following normalized analog transfer function:

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

Using the impulse invariant method obtain the transfer function, $H(z)$, of the digital filter, assuming a 3 dB cutoff frequency of 150Hz and a sampling frequency of 1.28 kHz. Also realize it with suitable structure. (5)

16. Consider the following specifications for a low pass filter.

$$\left|H\left(e^{j\omega}\right)\right| \le 0.01 \qquad 0 \le |\omega| \le 0.2\pi$$

$$0.95 \le \left|H\left(e^{j\omega}\right)\right| \le 1.05 \qquad 0.3\pi \le |\omega| \le 0.7\pi$$

$$\left|H\left(e^{j\omega}\right)\right| \le 0.02 \qquad 0.8\pi \le |\omega| \le \pi$$

Design a linear phase FIR filter to meet these specifications using a Blackman the window. (10)

17. (a) Explain the main characteristics of IIR filters and their requirement specifications. (5)

(b) A notch filter has the following transfer function :

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

Determine the transfer function of an equivalent discrete–time filter using the Bilinear transformation method. Assume a notch frequency of 50Hz and a sampling frequency of 500Hz. Also realize with suitable structure. (5)