Roll No. Total No. of Pages : 02

Total No. of Questions : 09

B.Tech.(EE) (2011 Onwards Elective-II)

B.Tech. (Electrical & Electronics) (2011 & 2012 Batch Elective-II)

(Sem.–7,8)

DIGITAL SIGNAL PROCESSING

Subject Code : BTEE-804C

Paper ID : [A3037]

Time : 3 Hrs. Max. Marks : 60

INSTRUCTION TO CANDIDATES :

1. SECTION-A is COMPULSORY consisting of TEN questions carrying TWO marks each.
2. SECTION-B contains FIVE questions carrying FIVE marks each and students have to attempt any FOUR questions.
3. SECTION-C contains THREE questions carrying TEN marks each and students have to attempt any TWO questions.

SECTION-A

1. Write briefly :

a) Write down the advantages of DSP.

b) Differentiate deterministic and random signals.

c) Differentiate between static and dynamic systems.

d) What do you mean by recursive and non-recursive discrete time systems? Explain.

e) Discuss the time shifting property of z transform.

f) Explain the significance of ROC in z transform.

g) What do you mean by frequency analysis? Explain.

h) Define DFT and List its computational requirements.

i) What do you mean by passband ripple? Explain.

j) Compare IIR and FIR filters.

SECTION-B

2. Determine the Fourier transform and the energy density spectrum of the sequence

$$x(n)=\begin{cases} A, & 0 \le n \le L-1 \\ 0, & \text{otherwise} \end{cases}.$$

3. Find the Z-transform of $x(n) = -a^n u(-n-1)$ $-1 < a < 1$

4. Determine the convolution of the following pair of signals

$$x(n) = |1 \quad 2 \quad 3 \quad 2|$$
$$\uparrow$$

$$h(n) = |3 \quad 2 \quad 1 \quad 5|$$
$$\uparrow$$

5. Compute the 4-point DFT of $x(n) = n + 1;\ 0 \le n \le 3$ by direct method.

6. Discuss the design of linear phase FIR filters using windows.

SECTION-C

7. a) Determine the inverse of the system with impulse response

$$h(n) = \delta(n) - \frac{1}{2}\delta(n-1)$$

b) What is a signal? Differentiate the signal into

a) Multichannel and multidimensional

b) Continuous time and discrete time

8. Convert the analog filter with system function

$$H_a(S) = \frac{s+0.1}{(s+0.1)^2+9}$$

Into a digital IIR filter by means of impulse invariance method.

9. Discuss the following

a) Filtering of long data sequences

b) Concept of frequency in continuous time and discrete time signals.