Roll No. Total No. of Pages : 02

Total No. of Questions : 09

B.Tech.(Electronics & Electrical Engg.) (2012 to 2017)

(Sem.–6)

DIGITAL SIGNAL PROCESSING

Subject Code : BTEEE-601

M.Code : 71130

Time : 3 Hrs. Max. Marks : 60

INSTRUCTIONS 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) Discuss the characteristics of digital signal processor.

b) Explain the applications of DSP.

c) Discuss the time shifting property of z transform.

d) What do you mean by LTI system? Explain.

e) Compare the computational requirements FFT algorithms.

f) Draw and explain cascade FIR structure.

g) Discuss the advantages and disadvantages of Goertzel algorithm.

h) List the advantages of DSP processors.

i) Draw and explain in brief the magnitude characteristics of physically realizable filters.

j) Explain the fixed point representation of numbers.

SECTION-B

2. Differentiate the following :

   a) Energy and power signals

   b) Periodic and aperiodic signals

3. Determine the causal signal $x(n)$ having the z-transform $X(z)=\frac{1}{(1+z^{-1})(1-z^{-1})^2}$

4. Discuss the use of DFT in linear filtering in detail.

5. Determine the lattice coefficient corresponding to the FIR filter with system function H(z) = A3(z) = $1+\frac{13}{24}z^{-1}+\frac{5}{8}z^{-2}+\frac{1}{3}z^{-3}$

6. Compare the internal architectures of ADSP and TMS series of processors.

SECTION-C

7. Explain the various advantages and disadvantages of DSP over analog processing. Also discuss various elementary discrete time signals and manipulations of discrete time signals.

8. Compute the 16-point DFT of the sequence using any of the FFT algorithm.

$$x(n) = \begin{cases} n+3, & 0 \le n \le 7 \\ 0, & otherwise \end{cases}$$

9. The normalized transfer function of an analog filter is given by :

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

   Obtain the transfer function, H(z), of an equivalent digital filter using the matched z-transform method. Assume a sampling frequency of 1.28 kHz and a 3dB cutoff frequency of 150 Hz.

NOTE : Disclosure of identity by writing mobile number or making passing request on any page of Answer sheet will lead to UMC against the Student.