Roll No. Total No. of Pages : 02

Total No. of Questions : 09

B.Tech.(Electrical & Electronics Engg.) (2012 Onwards) (Sem.–6)

DIGITAL SIGNAL PROCESSING

Subject Code : BTEEE-601

M.Code : 72835

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. Answer briefly :

a) Explain the function of D/A converter in DSP.

b) Why convolution is used? Explain.

c) Discuss the importance of ROC in Z transform.

d) What do you mean by transition band? Explain.

e) Explain unit step and unit ramp signals.

f) List the advantages and disadvantages of IIR filters.

g) What do you understand by quantization of filter coefficients? Explain.

h) Write down the advantages of DSP processors.

i) Comment upon the computational requirements of different FFT algorithms.

j) Draw the single stage lattice structure and write down where it is used.

SECTION-B

2. Explain the advantages and disadvantages of digital signal processing over analog processing. Also discuss the applications of DSP.

3. Determine the pole zero plot for the signal

$$x(n) = \begin{cases} a^n, & 0 \le n \le M-1 \\ 0, & elsewhere \end{cases}$$

Where $a > 0$.

4. Discuss in detail the frequency domain sampling and reconstruction of discrete time signal.

5. Discuss the direct form and cascade form structures for IIR systems.

6. What do you mean by frequency transformation? Transfer the single pole low pass Butterworth filter with the system function $H(s) = \frac{\Omega_p}{s + \Omega_p}$ into a bandpass filter with upper and lower band edge frequencies Ω_u and Ω_l respectively.

SECTION-C

7. Compute the 8-point DFT of the sequence.

$$x(n) = \begin{cases} 2n, & 0 \le n \le 7 \\ 0, & otherwise \end{cases}$$

by direct method and verify it with any of the FFT algorithms.

8. A requirement exists to simulate in a digital computer an analog system with the following normalized characteristics :

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

Obtain a suitable transfer function using :

a) The impulse invariant method, and

b) The bilinear transform method,

Assume a sampling frequency of 5 kHz and a 3dB cutoff frequency of 1 kHz.

9. Discuss the architectures of ADSP and TMS series of DSP processors.

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.