Roll No. Total No. of Pages : 02

Total No. of Questions : 09

B.Tech. (Electronics Engg.) (2012 Onwards)

B.Tech. (ECE/ETE) (2011 Onwards) (Sem.–5)

DIGITAL SIGNAL PROCESSING

Subject Code : BTEC-502

Paper ID : [A2104]

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) $x(n) = (0.4)^n u(n)$. State whether it is an energy or power signal. Justify.

b) What do you mean by causality and stability of a system?

c) State the differentiation property of Z-transform.

d) State the sampling theorem of signals. What will be the sampling rate if the maximum frequency component in an analog signal is 8 KHz?

e) Give the relation between linear and circular convolution.

f) What is meant by frequency warping? What is the cause of this effect?

g) Give the computational efficiency of FFT over DFT.

h) In the implementation of a digital system what are the effects of quantization of coefficients?

i) Give the advantages and disadvantages of FIR digital filter over IIR digital filter.

j) Explain the concept of pipelining in DSP processor.

SECTION-B

2. Determine the Z-transform of the signal

$$x(n) = n(-1)^n u(n)$$

3. Compute the linear convolution $y(n)$ using circular convolution of the signals

$$x(n) = \begin{cases} \frac{1}{4}n, & -1 \le n \le 5 \\ 0, & \text{elsewhere} \end{cases}$$

$$h(n) = \begin{cases} 1, & -2 \le n \le 3 \\ 0, & \text{elsewhere} \end{cases}$$

4. Explain the Goertzel Algorithm to compute the DFT using linear filtering approach.

5. The transfer function of analog filter is given as:

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

find the corresponding $H(z)$ using impulse invariance method for sampling frequency of 6 samples/sec.

6. Obtain the direct form – 1 & 2, cascade form realization for the following system

$$y(n) - \frac{3}{4}y(n-1) + \frac{1}{8}y(n-2) = x(n) + \frac{1}{2}x(n-1)$$

SECTION-C

7. Draw the 8-point: radix-2 decimation-in-time (DIT) FFT algorithm flow diagram and compute the DFT of the following sequence $x(n) = (1, 0, -2, 1, -1, 2, 0, 1)$.

8. Discuss various steps for the design of linear phase FIR filters using window method. Design the symmetric FIR low pass filter using rectangular window, whose desired frequency response is given as,

$$\mathrm{H}_d(\omega) = \begin{cases} e^{-j\omega\tau} & \text{for } |\omega| \le \omega_c \\ 0 & \text{otherwise} \end{cases}$$

The length of the filter should be 9 and $\omega_c = 1$ radian/sample.

9. Write short notes on the following :

a) Effect of round off noise in digital filters

b) Architecture of TMS series of digital signal processors.