Roll No. ☐☐☐☐☐☐☐☐☐☐☐☐ **Total No. of Pages : 03**

Total No. of Questions : 09

B.Tech.(Electronics & Computer Engg.) (2011 Onwards) (Sem.–6)

DIGITAL SIGNAL PROCESSING

Subject Code : BTEC-502

M.Code : 71164

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) What are the advantages of digitial signal processing?

b) Calculate DTFT of the sequence $x(n) = \left(\frac{1}{3}\right)^n U(n-2)$.

c) What is the z-transform of the sequence $x(n) = A\delta(n - m + p)$, where n, m and p are integers?

d) Match the following for the window functions.

Window function	**Peak of the sidelobe**
Rectangular	-58dB
Hamming	-32dB
Hanning	-43dB
Bartlett	-27dB
Blackman	-13dB

e) Define infinite impulse response and finite impulse response filters and compare.

f) What are the limitations of impulse invariant method ?

g) What is frequency warping? How it will arise?

h) Find the frequency response of a rectangular window.

i) What are the advantages of DSP processors over conventional processors?

j) List the status register bits of TMS320C5X and their functions.

SECTION-B

2. An FIR filter has the unit impulse response sequence $h(n) = \{1, -3, 1\}$. Determine the output sequence in response to the input sequence $x(n) = \{1, 3, 4, -7, -1, 1, 4, -5, 4, 2, -9, 3, 4\}$ using overlap add method.

3. Determine the 8-point DFT of the signal $x(n) = \{1\ 1\ 1\ 1\ 1\ 1\}$.

4. Determine the z-transform of the following sequences :

a) $x(n) = \frac{1}{2} n \left(\frac{1}{3}\right)^{n-1} U(n-1)$

b) $x(n) = -\alpha^n U(-n-3)$

5. Determine the causal signal $x(n)$ if its z-transform X(z) is given by $X(z) = \frac{z^{-6} + z^{-7}}{1 - z^{-1}}$.

6. A low-pass filter is to be designed with the following desired frequency response :

$$Hd(e^{jw}) = \begin{cases} e^{-2jw}, & for -\frac{\pi}{4} \le w \le \frac{\pi}{4} \\ 0, & for -\frac{\pi}{4} \le |w| \le \pi \end{cases}$$

Determine the filter coefficients $h(n)$ if the window function is defined as :

$$w(n)=\begin{cases} 1, & 0\le n\le 4 \\ 0, & otherwise \end{cases}$$

SECTION-C

7. Consider an FIR filter with system function

   $(z) = 1 + 2.88z^{-1} + 3.4048\, z^{-2} + 1.74\, z^{-3} + 0.4\, z^{-4}$

   Sketch the direct-form and lattice realizations of the filter and determine in detail the corresponding input-output equations.

8. a) What is bilinear transformation technique? Obtain the mapping formula and discuss the stability for this transformation technique.

   b) Transform the analog filter with the transfer function H(s) into a digital filter using backward difference for the derivative technique.

$$H(s) = \frac{1}{(s+2)(s+3)}$$

9. Write short notes on :

   a) Goertzel algorithm

   b) Region of convergence

   c) Effect of round off noise in digital filters

NOTE : Disclosure of Identity by writing Mobile No. or Making of passing request on any page of Answer Sheet will lead to UMC against the Student.