C 41260 **(Pages : 2)** Name..............................

Reg. No..............................

SIXTH SEMESTER B.TECH. (ENGINEERING) DEGREE EXAMINATION MAY 2013

AI 09 601—DIGITAL SIGNAL PROCESSING

(2009 admission onwards)

Time : Three Hours Maximum : 70 Marks

Part A

Answer **all** *questions.*

1. Show that, for symmetric $x(n), n = 0, 1, \ldots, N-1$, the DFT $X(k) = 0$ for $k = N/2$.
2. Obtain the circular convolution of $x[n] = \{1, 2, 1\}$ with $y[n] = \{1, -1\}$.
3. Draw the lattice structure realization of the FIR filter $H(z) = 1 + \frac{1}{2}z^{-1}$.
4. Write the transformation equation to convert a digital low-pass filter into a digital high-pass filter.
5. What are the different buses in TMS 320 C 54 processor ?

(5 × 2 = 10 marks)

Part B

Answer any **four** *questions.*

6. Show that 8-point DFT can be expressed in terms of two 4-point DFTs.
7. Let N-point DFT of $x(n)$ is $X(k)$. Express DFT of $x^*(n)$ and $e^{-j4\pi mn/N}x(n)$ in terms of $X(k)$.
8. What is overflow error ? How it is prevented ?
9. Prove that a stable analog filter will be mapped to a stable digital filter through impulse invariant transform.
10. Convert the analog filter having transfer function $H(s) = \dfrac{1}{s^2 + 3s + 2}$ into digital IIR filter using impulse invariant method.
11. With an example explain how a specific DSP hardware will increase the processing speed of a DSP algorithm implementation.

(4 × 5 = 20 marks)

Part C

Answer any **one** *question from each module.*

Module I

12. (a) (i) State and prove convolution property of DFT. (5 marks)

(ii) Show that DFT of even part of a signal $x(n)$ is equal to the real part of the DFT of $x(n)$.

(5 marks)

Turn over

(b) (i) Show that DFT of two real sequences of length N can be computed using one N-point DFT.

(6 marks)

(ii) State and prove time shifting property of DFT. (4 marks)

Module II

13. (a) Draw the direct form and lattice-ladder form realizations of the IIR filter :

$$H(z) = \frac{1 + 2z^{-1} + 3z^{-2} + 2z^{-3}}{1 + 0.9z^{-1} - 0.8z^{-2} + 0.5z^{-3}}.$$

(10 marks)

(b) Explain the limit cycle oscillations of a digital filter with respect the system described by the difference equation $y[n] = 0.95y[n-1] + x[n]$. Also determine the dead band of the filter.

(10 marks)

Module III

14. (a) Design an FIR linear phase filter using Hamming window approximating the ideal frequency response :

$$H(w) = \begin{cases} 1, \text{ for } |w| \le \frac{\pi}{4}, \\ 0, \text{ for } \frac{\pi}{4} < |w| \le \pi \end{cases}$$

Assume filter length L = 13. Draw the filter structure in Direct Form.

(10 marks)

(b) Design an digital IIR filter with the following specifications :

pass band = 0 – 12 kHz, stop band = 12.6 – 16 kHz, pass band ripple < 0.1 dB stop band attenuation > 30 dB, sampling frequency = 32 kHz. Draw the filter structure for the filter.

(10 marks)

Module IV

15. (a) Describe the function of on chip peripherals of TMS 320 series processors. (10 marks)

(b) What are the DSP specific processing units and instructions present in a typical digital signal processor ? Explain with appropriate examples.

(10 marks)

[4 × 10 = 40 marks]