C 29194 **(Pages : 2)** Name.................................

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

SEVENTH SEMESTER B.TECH. (ENGINEERING) DEGREE EXAMINATION JUNE 2012

EE 2K 702—DIGITAL SIGNAL PROCESSING

Time : Three Hours Maximum : 100 Marks

Answer **all** *questions.*

Part A

I. (a) Explain with suitable example :

(i) Linear shift invariant system.

(ii) Stable system.

(b) Explain :

(i) Sampling of continuous time signals.

(ii) Fourier transform of a signal.

(c) Discuss the following :

(i) FFT.

(ii) Convolution.

(d) Discuss any *one* FFT algorithm.

(e) Explain the parameter quantisation effect.

(f) Explain the features of a fixed point DSP core architecture.

(g) Explain about frequency prewarping.

(h) What are FIR and IIR filters Compare FIR and IIR filters.

(8 × 5 = 40 marks)

Part B

II. (a) What is z-transform of a signal ? State and prove its properties.

Or

(b) (i) Find the Z-transform of $x(n) = 2^n u(n-2)$.

(ii) Find the inverse z- transform of the following X (z) by partial fraction expansion method

$$X(z) = \frac{z+2}{2z^2 - 7z + 3} \text{ if ROC is } \tfrac{1}{2} < |z| < 3.$$

Turn over

III. (a) What is DFT of a signal ? State and prove its properties.

Or

(b) (i) Given $x(n) = 2^n$ and N = 8, find X (K) using DIT FFT algorithm.

(ii) Find the DFTs of the sequence $x(n) = \cos \frac{n\pi}{2}$, where N = 4 using DIF FFT algorithm.

IV. (a) Find the lattice-ladder structure of the following system function

$$H(z) = \frac{1 + 2z^{-1} + 3z^{-2}}{1 + 0.5\, z^{-1} + 0.125\, z^{-2}}.$$

Or

(b) Explain the architecture of a floating point digital signal processor.

V. (a) Explain the finite word length effect in DSP.

Or

(b) (i) Use the backward difference for derivative and convert the analog filter into digital filter

$$H(s) = \frac{1}{(s + 0.1)^2 + 9}.$$

(ii) Using impulse invariant technique find H (z). Assume T = 1s $H(s) = \frac{1}{(s+1)(s+2)}$.

(4 × 15 = 60 marks)