

**B.Tech. Degree VI Semester Examination
June 2005**

**CS/EI/EE 601 DIGITAL SIGNAL PROCESSING
(2002 Admissions)**

Time: 3 Hours

Maximum Marks: 100

- I. (a) Check whether the following systems are static, causal, linear, time invariant and stable :

$$(i) \quad y(n) = n e^{x(n)}$$

$$(ii) \quad y(n) = \sum_{k=-\infty}^{n+1} x(k) \quad (15)$$

- (b) The unit step response of a system is given by $2^{-n} u(n)$, find the impulse response of the system. (5)

OR

- II. (a) Show that the LTI systems are stable if and only if -

$$\sum_{k=-\infty}^{\infty} |h(k)| < \infty \quad (5)$$

- (b) Find the inverse Z - transform of $X(z)$ given by

$$X(z) = \frac{z^2}{(z - \frac{1}{2})(z - \frac{1}{4})}; \text{ROC: } |z| > \frac{1}{2}. \quad (8)$$

- (c) Find the Z - transform of the sequence $x(n)$ defined below:

$$x(n) = \begin{cases} 3^n, & n < 0 \\ \left(\frac{1}{3}\right)^n, & n = 0, 2, 4, \dots \\ \left(\frac{1}{2}\right)^n, & n = 1, 3, 5, \dots \end{cases} \quad (7)$$

(Turn Over)

- III. (a) Explain in detail the properties of DFT. (10)
 (b) Compute the linear convolution of the two sequences
 $x(n) = \{1, 2, 2, 1\}$ and $h(n) = \{1, 2, 3\}$ using DIT-FFT. (10)

OR

- IV. (a) Describe the DIT - FFT algorithm. (10)
 (b) Compute the 8 point DFT $X(k)$ of the real sequence
 $x(n) = \{1, 1, 1, 1, -1, -1, -1, -1\}$ by using DIF - FFT algorithm. (10)

- V. (a) Explain the frequency sampling method of FIR filter design. (10)
 (b) Design a low pass FIR filter for the following specifications :

$$H(j\omega) = 1 \text{ for } \Omega \leq 5 \text{ rad/sec}$$

$$= 0 \text{ for } \Omega > 5 \text{ rad/sec.}$$

Sampling frequency 20 rad/sec. Use Fourier series method with $N = 9$ and Hamming window for the design. (10)

OR

- VI. (a) Explain the principle of FIR filter design using windows. (10)
 (b) Using rectangular window technique design a low pass filter with pass band gain of unity, cut-off frequency of 1500 Hz and working at a sampling frequency of 5 KHz. The length of the impulse response should be 7. (10)

- VII. (a) Explain the bilinear transformation method for the design of digital filters. (10)
 (b) A third order butterworth low pass filter with 3dB frequency of 1 KHz is to be realized using digital systems. Assume sampling frequency to be 5 KHz. Realize the filter using impulse invariant technique. (10)

OR

- VIII. (a) Obtain the direct form I, direct form II, cascade and parallel form realization of the following systems :

$$y(n) = y(n-1) - \frac{1}{2}y(n-2) + x(n) - x(n-1) + x(n-2) \quad (16)$$

- (b) Compare IIR and FIR filters. (4)

- IX. (a) Draw and explain the block diagram of a typical DSP processor. (12)
 (b) Discuss the effect of round-off noise in digital filter implementation. (8)

OR

- X. (a) Briefly explain *any three* applications of Digital Signal processing. (12)
 (b) What is meant by limit cycle oscillations? Illustrate with examples. (8)
