## B. Tech Degree VI Semester Examination, April 2008

## CS/EI/EE 601 DIGITAL SIGNAL PROCESSING

(2002 Scheme)

Time: 3 Hours Maximum Marks: 100 I. Check the linearity, time invariance, causality and stability of the following systems: (a) y(n) = nx(n)(i) (4)  $v(n) = x(n^2)$ (ii) (4) $y(n) = \sum_{k=0}^{n} x(k)$ (4) Determine the steady state response for the system with impulse function (b)  $h(n) = \left(\frac{j}{2}\right) u(n)$  for an input  $x(n) = (\cos \pi n)u(n)$ . (8) II. Explain the properties of z – transform. (a) (10)Define system function and find the system function and impulse response of the (b) system described by the difference equation v(n) = x(n) + 3x(n-1) - 2x(n-2) + x(n-3). (6) Find the stability of the system whose impulse response  $h(n) = 2^n u(n)$ . (c) (4) Explain the properties of discrete fourier transform. III. (a) (10)(b) Find the IDFT of the sequence  $X(k) = \{5,0,1-j,0,1,0,1+j,0\}.$ (10)Find the output, y(n) of a filter whose impulse response is  $h(n) = \{1, 1, 1\}$  and IV. (a) input signal  $x(n) = \{3, 2, 1, 0, 1, 2, 3, -1, 2, 1\}$  using overlap save method. (10)Find the DFT of the sequence x(n) using DIT FFT algorithm. (b)  $x(n) = \{2, 2, 2, 2, 1, 1, 1, 1\}.$ (10)V. (a) Explain the relevance of window function and explain each window. (10)Design a high pass filter using hamming window with a cut off frequency of (b) 1.2 radians/sec. (10)OR VI. (a) Explain frequency sampling method of FIR filter design. (10)Obtain the direct and cascade form realization of the system function (b)

 $H(z) = 1 + \frac{5}{2}z^{-1} + 2z^{-2} + 2z^{-3}$ 

(10)

(Turn Over)

VII. (a) Realize the system in cascade and parallel form

$$H(z) = \frac{1 + \frac{1}{2}z^{-1}}{\left(1 - z^{-1} + \frac{1}{4}z^{-2}\right)\left(1 - z^{-1} + \frac{1}{2}z^{-2}\right)}.$$
 (10)

(b) Convert the analog filter with system function H<sub>a</sub> (S) in to digital filter using bilinear transformation

$$H_a(S) = \frac{S + 0.1}{\left(S + 0.1\right)^2 + 16}. (10)$$

OR

- VIII. (a) Explain warping effect on magnitude and phase response in an IIR filter. How this can be eliminated?
  - (b) Design a chebyshev digital filter using impulse invariant transformation with following specification:

$$0.9 \le |H(w)| \le 1.0; \ 0 \le w \le 0.25\pi$$

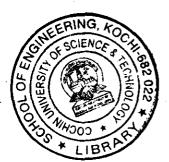
$$|H(w)| \le 0.24; \ 0.5\pi \le w \le \pi$$
(12)

IX. Write short notes on:

- (i) Fixed point and floating point arithmetic (5)
- (ii) Truncation and rounding errors in digital filters (5)
- (iii) Product quantization error (5)
- (iv) Limit cycle oscillation (5)

OR

- X. (a) Draw and explain the architecture of a typical DSP processor. (10)
  - (b) Explain any one application of digital signal processing. (10)



(8)