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) $y(n) = n e^{x(n)}$

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

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

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

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

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

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

(Turn Over)

III. (a) Explain in detail the properties of DFT. (10)

(b) Compute the linear convolution of the two sequences

$x(n) = \{\underset{\uparrow}{1},2,2,1\}$ and $h(n) = \{\underset{\uparrow}{1},2,3\}$ 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 :

$$H(jw) = 1 \text{ for } \Omega \le 5\,rad/\sec$$

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

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

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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)

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