BTS (C) – VI – (S) – 05 – 072 (B)

B. Tech Degree VI Semester (Supplementary) Examination, November 2005

CS/EC/EI/EE 601 DIGITAL SIGNAL PROCESSING

(Prior to 2002 Admissions)

Time : 3 Hours   Maximum Marks:100

I. (a) Check linearity, causality, time invariance and stability of the following systems.

$(i)\ y(n)=e^{x(n)}$ $(ii)\ y(n)=\sum_{K=-\infty}^{n} x(k)$ (10)

(b) Find out inverse z transform of -

$(i)\ X(z)=\dfrac{z+0.2}{z^2-0.5z-0.5}\ ROC:|Z|>1$

$(ii)\ x(z)=\dfrac{1}{(1-2z^{-1})(1-z^{-1})^2}\ ROC:|Z|>1$ (10)

OR

II. (a) Find out z transforms of the following including ROC

$(i)\ x(n)=a^n\cos(nq)u(n)$ $(ii)\ y(n)=\left(\dfrac{1}{2}\right)^n[u(n)-u(n-10)]$ (10)

(b) Find out impulse response and unit slip response of the given system described by the difference equation -

$$y(n)=\frac{1}{2}y(n-1)+x(n)$$ (10)

III. (a) Develop decimation in frequency FFT algorithm. (12)

(b) Explain the filtering of long input sequence by over lap save method. (8)

OR

IV. (a) State and prove circular convolution property and time reversal property of DFT. (10)

(b) Obtain the DFT of the sequence $x(n)=\{1,2,-2,-1,0,0,2,1\}$ by using DIT-FFT (10)

V. (a) Explain how FIR filler can be designed using windowing technique. Explain Gibbs phenomenon and its reason. (10)

(b) Design an FIR filter with

$$H(c^{jw})=e^{-j(\frac{n-1}{2})w} \quad 0\le|w|\le\frac{\pi}{2}$$
$$=0 \quad \frac{\pi}{2}\le|w|\le\pi$$

using frequency sampling with N=7 (10)

OR

VI. (a) State and prove the property of impulse response of an FIR fitter to have linear phase characteristics. (10)

(b) Design an FIR low pass fitter with cut off frequency 100Hz and sampling frequency 500Hz using hamming window and 10 delays. (10)

VII. (a) Obtain direct form I, direct form II cascade and parallel form of realization for the following system.

$$y(n)=-0.1y(n-1)+0.2y(n-2)+3x(n)+3.6x(n-1)\_+0.6x(n-2).$$ (12)

(b) Transfer function of an analog filter is $H(s)=\dfrac{1}{s^2+3s+3}$. Find out corresponding transfer function of digital filter by using (i) impulse invariant method with T = 1 (ii) Bilinear transform with T = 2. (8)

OR

VIII. (a) Using Bilinear transformation design a high pass filter monotonic in pass band with cut off frequency 1.25 KHz and down 10 db at 400 Hz. The sampling frequency is 5 KHz. (12)

(b) Prove that impulse invariant transformation transforms a stable analog filter into a stable digital filter. (8)

IX. (a) Draw and explain the block diagram of a typical DSP processor. (10)

(b) Explain effects of coefficient quantization error in IIR filters. (10)

OR

X. (a) Explain overflow lenit cycle oscillation and zero input lenit cycle oscillation using suitable examples. (12)

(b) Explain the major application areas of DSP. (8)

***