BTS(C) – VI – (S) – 08 – 045 – C

B. Tech Degree VI Semester (Supplementary) Examination, September 2008

CS/EI/EE 601 DIGITAL SIGNAL PROCESSING

(2002 Scheme)

Time : 3 Hours | Maximum Marks : 100

I. (a) What is impulse response of a system and give its significance? (3)

(b) The input sequence $\left(\underset{\uparrow}{3},1,2,-1\right)$ is applied to a discrete time processor with unit sample response $h(n)=\left(\underset{\uparrow}{1},2,1\right)$. Compute the resulting output sequence of the processor. (7)

(c) Let $e(n)$ be an exponential sequence $e(n)=\alpha^n$ for all n, let $x(n)$ and $y(n)$ denote two arbitrary sequences. Show that $\left[e(n)x(n)*e(n)y(n)\right]=e(n)\left[x(n)*y(n)\right]$ where $*$ denotes convolution. (10)

OR

II. (a) List the condition for (i) Linearity (ii) Stability (iii) Causality for an LTI system. Check the linearity, stability and causality of the system defined by $y(n)=n\,x(n)$. (12)

(b) Find the Z-transform of $x(n)=Cos\,wn\;\;u(n)$. (8)

III. Find the linear and circular convolution of the sequence $x(n)=\left(\underset{\uparrow}{1},0.5\right)$ and $h(n)=\left(\underset{\uparrow}{0},5,1\right)$ using DFT. (20)

OR

IV. Compute the 8 point DFT of the sequence $x(n)=(1,1,1,1,0,0,0,0)$ using decimation in time FFT algorithm. Write the necessary butterfly diagram. (20)

V. Design a low pass FIR filter for N=7 and $W_c = 1 rad/\sec$ using

(i) Rectangular window (ii) Hamming window (20)

OR

VI. (a) Prove that FIR filter have a linear phase characteristic for a symmetrical impulse response with even number of samples. (10)

(b) Explain the frequency sampling method of FIR filter design. (5)

(c) What is Cubb's phenomenon? (5)

VII. (a) Explain the impulse invariant transformation. (10)

(b) Convert the analog filter with system transfer function $Ha(s)=\dfrac{s+0.1}{(s+0.1)^2+9}$ into a digital IIR filter by means of impulse invariant transformation. (10)

OR

VIII. (a) Explain the bilinear transformation. (10)

(b) Convert the analog filter with system function $Ha(s)=\dfrac{s+0.3}{(s+0.3)^2+16}$ into a digital IIR filter using bilinear transformation. (10)

IX. (a) What are the important characteristics of a DSP processor? (6)

(b) Briefly explain any one application of digital signal processing. (14)

OR

X. (a) Explain the quantization process via truncation and rounding. (12)

(b) What is meant by limit cycle oscillation explain? (8)

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