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06EE64

Sixth Semester B.E. Degree Examination, December 2010

Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note: ***1. Answer any FIVE full questions, selecting at least TWO questions from each part.***
2. Normalized Butterworth letter table may be provided.

PART – A

1 a. State and prove the following properties of DFT.
i) Linearity
ii) Frequency shifting. **(06 Marks)**

b. Computer the circular convolution between the following sequences using DFT and IDFT methods
x[n] = (1, 2, 3, 4) and y[n] = (-1, -2, -3, -4). x[n] and y[n] are periodic sequences with period N = 4. **(14 Marks)**

2 a. Explain the concept of Overlap – Add method, with the necessary diagram. **(08 Marks)**

b. Compute the N pt DFT of a^n and a. n. **(12 Marks)**

3 a. Find the DFT of x[n] = [1, 2, 3, 4, 4, 3, 2, 1] using the DIT– FFT algorithm. **(10 Marks)**

b. Develop an 8 point DIF–FFT algorithm, starting from DFT. State clearly all the steps. Explain how it reduces the number of computations. **(10 Marks)**

4 a. Obtain the direct form–I, direct form–II, cascade and parallel form realizations for the following system :
y[n] = 0.75 y[n – 1] – 0.125 y[n – 2] + 6x[n] + 7x[n – 1] + x[n – 2]. **(14 Marks)**

b. Describe with necessary diagram and equations, the linear phase structure of FIR filter for even order. **(06 Marks)**

PART – B

5 a. Let $H_a(s) = \dfrac{b}{(s+a)^2+b^2}$ be a causal second order analog transfer function. Show that the causal second order digital filter transfer function H(t), obtained from $H_a(s)$, through impulse invariance method is given by

$$H(t) = \frac{e^{-aT}\sin bT\ t^{-1}}{1-2e^{-aT}\text{col}bT\,t^{-1}+e^{-2aT}\,t^{-2}}.$$

Also find H(z) when $H_a(s) = \dfrac{1}{s^2+2s+2}$. **(12 Marks)**

b. The system function of the analog filter is given as

$$H_a(s) = \frac{s+0.1}{(s+0.1)^2+16}.$$

Obtain the system function of the digital filter, using bilinear transformation, which is resonant at $W_r = \pi/2$. **(08 Marks)**

6 a. Design a Butterworth analog high pass filter, that will meet the following specifications :
   i) Maximum passband attenuation = 2 dB
   ii) Passband edge frequency = 200 red/sec
   iii) Minimum stopband attenuation = 20 dB
   iv) Stopband edge frequency = 100 rad/sec. **(10 Marks)**

b. Design a Cheebyshev analog lowpass filter that has – 3dB cutoff frequency of 100 rad/sec, and a stop band attenuation of 25 dB or greater for all radian frequency past 250 rad/sec. **(10 Marks)**

7 a. Explain why windows are necessary in FIR filter design. What are the different windows in practice? Explain the design procedure for the design of FIR filters using windows. **(12 Marks)**

b. Design an FIR (low-pall) filter using rectangular window with passband gain of 0 dB, cut-off frequency of 200 Hz, sampling frequency of 1 KHz. Assume the length of the pulse response as 7. **(08 Marks)**

8 a. Distinguish between the analog and digital filters. **(08 Marks)**

b. Explain the architecture of TMS 320 C 5X processor, with a neat diagram. **(12 Marks)**

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