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06EE64

Sixth Semester B.E. Degree Examination, May/June 2010
Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note: ***Answer any FIVE full questions, selecting at least TWO questions from each part.***

Important Note : 1. On completing your answers, compulsorily draw diagonal cross lines on the remaining blank pages.
2. Any revealing of identification, appeal to evaluator and /or equations written eg, 42+8 = 50, will be treated as malpractice.

PART – A

1 a. Given x [n] = [1, 1, 1], obtain the five point DFT X(K). **(08 Marks)**

b. If DFT of x(n) = X(K) show that:

$$x[(-n)]_N \xleftrightarrow{DFT} X[(-K)]_N \quad \text{and} \quad x^*(n) \xleftrightarrow{DFT} X^*(N-K)$$ **(08 Marks)**

c. Obtain the 10 pt DFT of the sequence $x(n) = \delta(n) + 2\delta(n-5)$. **(04 Marks)**

2 a. Let $x(n) = 2\delta(n) + \delta(n-1) + \delta(n-3)$. Obtain the sequence y(n) whose 5 point DFT $Y(K) = [X(K)]^2$ where X(K) is the 5 point DFT of x(n). Use DFT and IDFT method. **(08 Marks)**

b. A long sequence x(n) is filtered through a filter with an impulse response h(n) to give an output y(n). If x(n) = (1, 1, 1, 1, 1, 3, 1, 1, 4, 2, 1, 1, 3, 1) and h(n) = (1, -1), compute y(n) using overlap add or overlap save method. Use only 5 point circular convolution in your approach. **(12 Marks)**

3 a. What are FFT algorithms? Explain the advantages of FFT algorithms over the direct computation of DFT for a sequence x(n). **(04 Marks)**

b. Compute the 8 point DFT of the sequence $x(n) = \left[\frac{1}{2}, \frac{1}{2}, \frac{1}{2}, \frac{1}{2}, 0, 0, 0, 0\right]$, using radix 2 DIT FFT algorithm. **(08 Marks)**

c. Develop a radix 3 DIF FFT algorithm for evaluating the DFT for N = 9. **(08 Marks)**

4 a. Consider $H(z) = \dfrac{\left(1 + \frac{1}{5}z^{-1}\right)}{\left(1 - \frac{1}{2}z^{-1} + \frac{1}{3}z^{-2}\right)\left(1 + \frac{1}{4}z^{-1}\right)}$.

i) Realize the system in direct form I.

ii) Realize the system in cascade form using the first order and second order form II structures.

iii) Realize the system in parallel form using the first order and second order form II structures. **(12 Marks)**

b. Realize the linear phase FIR filter having the following impulse response:

$$h(n) = \delta(n) - \frac{1}{4}\delta(n-1) + \frac{1}{2}\delta(n-2) + \frac{1}{2}\delta(n-3) - \frac{1}{4}\delta(n-4) + \delta(n-5)$$ **(08 Marks)**

PART – B

5 a. Design a Butterworth filter using bilinear transformation for the following specifications:

$$0.8 \le |H(\omega)| \le 1 \quad \text{for} \quad 0 \le \omega \le 0.2\pi$$
$$|H(\omega)| \le 0.2 \quad \text{for} \quad 0.6 \le \omega \le \pi$$ **(10 Marks)**

5 b. Use impulse invariance method to design a digital filter from an analog prototype that has a system function:

$$H_a(s) = \frac{s+a}{(s+a)^2 + b^2} \quad \text{and} \quad H_a(s) = \frac{b}{(s+a)^2 + b^2}$$ **(10 Marks)**

6 a. Explain the bilinear transform method. Derive an expression showing mapping from s plane to z plane. Show that there is no aliasing effect in bilinear transformation. **(08 Marks)**

b. Design a digital lowpass Chebyshev filter that meets the following specifications: Passband magnitude characteristic that is constant to within 1 dB for frequencies below, w = 0.2 π and stopband attenuation of atleast 15 dB for frequencies between w = 0.3 π and π. Use bilinear transformation. **(12 Marks)**

7 a. The frequency response of a filter is described by:

$$H(\omega) = j\omega \quad -\pi \le \omega \le \pi$$

Design the filter using a rectangular window. Take N = 7. **(08 Marks)**

b. Design a lowpass digital filter to be used in an A/D – H(Z) – D/A structure that will have a –3dB cutoff at 30 π rad/sec and an attenuation of 50 dB at 45 π rad/sec. The filter is required to have a linear phase and the sampling rate is 100 samples/sec. **(12 Marks)**

8 a. Explain Gibb's phenomenon and the methods of minimizing it. **(10 Marks)**

b. Explain the principle features of Harvard architecture and the modified Harvard architecture. **(10 Marks)**

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