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06EC52

Fifth Semester B.E. Degree Examination, June/July 2011
Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note: *1. Answer FIVE full questions selecting at least TWO questions from each part.*
2. Standard notations are used.
3. Missing data be suitably assumed.
4. Draw neat diagram wherever necessary.

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

PART – A

1 a. Prove that the sampling of Fourier transform of a sequence x(n) results in N point DFT using which both the sequence and the transform can be reconstructed. **(10 Marks)**

b. Derive the relationship between N point DFT and Z transform. **(04 Marks)**

c. Evaluate the following function without computing the DFT: $\sum_{k=0}^{11} e^{\frac{-j4\pi k}{6}} x(k)$ for a given 12 point sequence $x(n) = [8, 4, 7, -1, 2, 0, -2, -4, -5, 1, 4, 3]$. **(06 Marks)**

2 a. Let x(n) be a length 'N real sequence with N point DFT X(k). Prove that :
i) $X(N-k) = X^*(k)$ and ii) X(0) is real. **(06 Marks)**

b. P.T $\sum_{n=0}^{N-1} x(n)y^*(n) = \frac{1}{N}\sum_{k=0}^{N-1} x(k)y^*(k)$. **(06 Marks)**

c. Determine the N point circular correlation of $x_1(n)$ and $x_2(n)$ defined by
$x_1(n) = \text{Cos}\frac{2\pi n}{N}$ $\quad x_2(n) = \text{Sin}\frac{2\pi n}{N}$. **(08 Marks)**

3 a. Using DFT properties which relates linear convolution to circular convolution, obtain the output of a linear filter given the impulse response h(n) = [1, 1, 1] and an input to be a long sequence x(n) = [1, 2, 0, -3, 4, 2, -1, 1, -2, 3, 2, 1, -3]. **(09 Marks)**

b. Find the 4 point DFT of e real sequences using single 4 point DFT. Given :
g(n) = [1, 2, 0, 1] and h(n) = [2, 2, 1, 1]. **(09 Marks)**

c. How many multiplications and additions are needed for 64 point sequence in calculation of DFT using FFT algorithm and using direct DFT computation? Also specify number of real registers needed to perform these computations. **(02 Marks)**

4 a. Derive the radix – 2 DIT FFT algorithm to compute DFT of an N = 8 point sequence and draw the complete signal flow graph. **(07 Marks)**

b. Using DIT FFT find the sequence x(n) corresponding to 8 point DFT, where x(k) is given by x(k) = [4, 1-j 2.414, 0, 1-j 0.414, 0, 1 + j 0.414, 0, 1+j 2.414]. **(08 Marks)**

c. Derive the impulse response and hence the transfer functions for Goertzel filter and realize the same in DF – II. **(05 Marks)**

PART – B

5 a. Derive the expression for poles from the squared magnitude response of Butterworth L.P.F. **(06 Marks)**

b. Transform the third order Butterworth normalized low pass filter to high pass filter with passband edge at 2 rad/sec. (Transfer function can be directly written). **(07 Marks)**

c. Derive an expression for the order of Chebyshev type 1 low pass filter. **(07 Marks)**

6 a. Derive the frequency response of a symmetric FIR low pass filter for both N even and Nodd. **(08 Marks)**

b. A low pass filter is to be designed with the following desired frequency response :

$$H_d(e^{jw}) = H_d(w) = e^{-j2w} \ldots\ldots |w| < \pi/4$$
$$= 0 \ldots\ldots \pi/4 \le w \le \pi$$

Determine the filter coefficients h(n), given rectangular window w(n) defined by
$w(n) = 1 \quad 0 \le n \le 4$
$= 0$ Otherwise. **(07 Marks)**

c. Explain the frequency sampling design of FIR filters and realize it in DF structure. **(05 Marks)**

7 a. Derive the transformation of IIR filter using approximation of derivatives by backward difference and verify whether it satisfies the sufficient and necessary conditions of mapping. **(06 Marks)**

b. Convert the following transfer function into digital using impulse invariance method :

$$H(s) = \frac{s+a}{(s+a)^2 + b^2}.$$ **(05 Marks)**

c. Design a digital Butterworth filter H(z) given an equivalent analog filter with following specifications : passband ripple ≤ 3db, stopband edge frequency of 750 hz, stop band attenuation of 15 db, passband edge frequency = 500 hz and sampling rate is 2 kHz. Design using bilinear transformation. **(09 Marks)**

8 a. Obtain a DF – II and cascade realization for the system function :

$$H(z) = \frac{(1+z^{-1})}{\left(1-\frac{1}{4}z^{-1}\right)\left(1-z^{-1}+\frac{1}{2}z^{-2}\right)}.$$ **(06 Marks)**

b. Obtain linear phase realization of the impulse response using ladder structure symmetric structure : $h(n) = \delta(n) - \frac{1}{4}\delta(n-1) + \frac{1}{2}\delta(n-2) + \frac{1}{2}\delta(n-3) - \frac{1}{4}\delta(n-3) + \delta(n-5)$. **(06 Marks)**

c. Determine the coefficients k_m of lattice filter whose transfer function corresponding to FIR filter described by the transfer function $H(z) = 1 + 2z^{-1} + \frac{1}{3}z^{-2}$. Also draw the corresponding II order lattice structure. **(08 Marks)**

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