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10EC52

Fifth Semester B.E. Degree Examination, December 2012

Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note: 1. ***Answer FIVE full questions, selecting at least TWO questions from each part.***
2. ***Use of normalized Chebyshev and Butterworth prototype tables are NOT ALLOWED.***

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. Find the N-point DFT of x(n) if $x(n) = \begin{cases} \frac{1}{3}; & 0 \le n \le 2 \\ 0; & \text{otherwise} \end{cases}$. **(08 Marks)**

b. Two finite sequences x(n) = [x(0), x(1), x(2), x(3)] and h(n) = [h(0), h(1), h(2), h(3)] have DFTs given by X(R) = DFT {x(n)} = {1, J, -1, -J}; H(R) = DFT {h(n)} = {0, 1+J, 1, 1-J}. Use the properties of the DFT and find the following:

i) $X_1(R)$ = DFT {h(0), -h(1), h(2), -h(3)}

ii) $X_2(R)$ = DFT {y(n)} where $y(n) = x(n) \underset{4}{\circledast} h(n)$

iii) $X_3(R)$ = DFT{x(0), h(0), x(1), h(1), x(2), h(2), x(3), h(3)} **(12 Marks)**

2 a. Consider a length - 12 sequence defined for
$0 \le n \le 11$, x(n) = {8, 4, 7, -1, 2, 0, -2, -4, -5, 1,4 , 3}
with 12-point DFT given by X(R), $0 \le R \le 11$, evaluate the following function without computing DFT, $\sum_{R=0}^{11} e^{-\frac{J4R}{6}} x(R)$ **(05 Marks)**

b. Determine $x_3(n) = x_1(n) \underset{8}{\circledast} x_2(n)$ for the sequences, $x_1(n) = e^{j\pi n}$; $0 \le n \le 7$;
$x_2(n) = u(n) - u(n-5)$. Sketch all the sequences. Use time domain approach. **(08 Marks)**

c. Show that:

i) Real and even sequence has real DFT.

ii) Multiplication of two DFT's in frequency domain corresponds to circular convolution in time domain. **(07 Marks)**

3 a. Consider a FIR filter with impulse response h(n) = {3, 2, 1, 1} if the input is
x(n) = {1, 2, 3, 3, 2, 1, -1, -2, -3, 5, 6, -1, 2, 0, 2, 1},
find the output y(n). Use overlap-add method assuming the length of block is 7. **(09 Marks)**

b. Write a note on Chirp z-transform. **(06 Marks)**

c. What is in-place computation? What is the total number of complex additions and multiplications required for N = 512 point, if DFT is computed directly and if FFT is used? Also find the number of stages required and its memory requirement. **(05 Marks)**

4 a. Derive DIT-FFT algorithm for N = 8 and draw the complete signal graph. **(12 Marks)**

b. Find the IDFT of X(R) = {0, 2 + 2j, –j4, 2 – 2j, 0 2+2j, j4, 2 – 2j} using inverse Radix – 2 DIT-FFT algorithm. **(08 Marks)**

PART – B

5 a. Design a Chebyshev analog low pass filter that has -3dB cut off frequency of 100 rad/sec and a stopband attenuation of 25 dB or greater for all radian frequencies past 250 rad/sec. Verify the design. **(10 Marks)**

b. Derive the s to z plane transformation based on finite backward difference method. Also show that the entire left half s-plane poles are mapped inside the smaller circle of radius ½ centered at z = ½ inside the unit circle in the z-plane. **(10 Marks)**

6 a. Obtain the direct form II (canonic) and cascade realization of

$$H(z)=\frac{(z-1)(z^2+5z+6)(z-3)}{(z^2+6z+5)(z^2-6z+8)}$$

the cascade system should consist of two biquadratic sections. **(10 Marks)**

b. Given $H(z) = (1 + 0.6z^{-1})^5$

i) Realize in direct form

ii) Realize as a cascade of first order sections only

iii) As a cascade of 1st and 2nd order sections. **(10 Marks)**

7 a. Using rectangular window technique, design a lowpass filter with passband gain of unity, cut off frequency of 1000 Hz and working at a sampling frequency of 5 kHz. The length of impulse response should be 7. **(10 Marks)**

b. With necessary mathematical analysis, explain the frequency sampling technique of FIR filter design. **(10 Marks)**

8 a. Design a digital filter H(z) that when used in A/D – H(z) – D/A structure, gives an equivalent analog filter with the following specifications:

PB Ripple ≤ 3.01 dB
PB Edge : 500 Hz
SB attenuation ≥ 15 dB
SB Edge : 750 Hz
Sample rate : 2 kHz

The filter is to be designed by performing a bilinear transformation on an analog system function. Use Butterworth prototype. Also obtain the difference equation. **(15 Marks)**

b. If $H_a(s)=\frac{1}{(s+2)(s+1)}$; find the corresponding H(z) using impulse invariance method for sampling frequency of 5 samples/sec. **(05 Marks)**

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