

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.**

PART - A

- 1 a. Find the N-point DFT of $x(n)$ if $x(n) = \begin{cases} \frac{1}{3}; & 0 \leq n \leq 2 \\ 0; & \text{otherwise} \end{cases}$. (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 $X(R) = \text{DFT} \{x(n)\} = \{1, J, -1, -J\}$; $H(R) = \text{DFT} \{h(n)\} = \{0, 1+J, 1, 1-J\}$. Use the properties of the DFT and find the following:
- i) $X_1(R) = \text{DFT} \{h(0), -h(1), h(2), -h(3)\}$
- ii) $X_2(R) = \text{DFT} \{y(n)\}$ where $y(n) = x(n) \otimes_4 h(n)$
- iii) $X_3(R) = \text{DFT} \{x(0), h(0), x(1), h(1), x(2), h(2), x(3), h(3)\}$ (12 Marks)
- 2 a. Consider a length - 12 sequence defined for $0 \leq n \leq 11$, $x(n) = \{8, 4, 7, -1, 2, 0, -2, -4, -5, 1, 4, 3\}$ with 12-point DFT given by $X(R)$, $0 \leq R \leq 11$, evaluate the following function without computing DFT, $\sum_{R=0}^{11} e^{-\frac{J4R}{6}} \times (R)$ (05 Marks)
- b. Determine $x_3(n) = x_1(n) \otimes_8 x_2(n)$ for the sequences, $x_1(n) = e^{j\pi n}$; $0 \leq n \leq 7$; $x_2(n) = u(n) - u(n-5)$. 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)

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.

- 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 $\frac{1}{2}$ centered at $z = \frac{1}{2}$ 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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