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10EC52

Fifth Semester B.E. Degree Examination, Jan./Feb. 2021
Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note:1. ***Answer any FIVE full questions, selecting atleast TWO questions from each part.***
2. ***Use of Normalized Prototype filter tables are not allowed.***

PART – A

1 a. Compute 4 – point DFT of a sequence $x(n) = (-1)^n$. **(06 Marks)**

b. Compute the N – point DFT of the window sequence.

$$W(n) = \frac{1}{2} + \frac{1}{2}\cos\left[\frac{2\pi}{N}\left(n - \frac{N}{2}\right)\right], \quad 0 \le n \le N-1.$$ **(08 Marks)**

c. Derive the Relationship between DFT to DTFT and DFT to Z – transform. **(06 Marks)**

2 a. For the sequences $x_1(n) = \cos\left(\frac{2\pi n}{N}\right)$ and $x_2(n) = \sin\left(\frac{2\pi n}{N}\right)$. Find the N – point circular convolution $x_1(n) \circledast_N x_2(n)$. **(07 Marks)**

b. Given the 8 – point sequence

$$x(n) = \begin{cases} 1, & 0 \le n \le 3 \\ 0, & 4 \le n \le 7 \end{cases}.$$

Compute the DFT of the sequence $x_1(n)$ as

$$x_1(n) = \begin{cases} 0, & 0 \le n \le 1 \\ 1, & 2 \le n \le 5 \\ 0, & 6 \le n \le 7 \end{cases}.$$

Use Property of DFT. **(08 Marks)**

c. Let x(n) be a Complex – Valued Sequence. Prove that $DFT[x^*(n)] = X^*(N-K) = X^*((-K))_N$. **(05 Marks)**

3 a. Find the output Y(n) of a filter whose impulse response is h(n) = {3, 2, 1, 1} and input sequence x(n) = {1, 2, 3, 4, 5, 1, 2, 3, 4, 5, 1, 2,3, 4, 5}. Using overlap – add method. Use 8 – point circular convolution in your approach. **(08 Marks)**

b. What is meant by Sectional convolution? Explain any one method. **(06 Marks)**

c. Determine Number of complex multiplications and complex additions for N = 256 using Direct computation of DFT and using FFT algorithm and also calculate speed improvement factor for multiplication. **(06 Marks)**

4 a. Compute 8 – point DFT of the sequence x(n) = {2, 1, 2, 1, 0, 0, 0, 0} using Radix – 2 DIT FFT algorithms. Show clearly all the intermediate results. **(08 Marks)**

b. What is Goertzel algorithm? For the sequence x(n) = {5, 3 - j2, -3, 3 + j2}, determine x(2) using Goertzel algorithm. Assume the initial conditions are zero. **(08 Marks)**

c. Compute the 4 – point FDFT of the sequence X(K) = {4, 1 - j , -2, 1 + j}, using DIF – FFT algorithm. **(04 Marks)**

PART – B

5 a. Compare Butterworth Filter and Chebyshev type – 1 filter. **(04 Marks)**

b. A Chebyshev type – 1 analog Low pass filter is required for the following specifications :

i) Pass band attenuation of -2 dB at 3.4 KHz.

ii) Stop band attenuation of -15 dB at 8KHz. Design a filter. **(10 Marks)**

c. What are the different types of analog frequency transformations? Write its transformed frequency responses. **(06 Marks)**

6 a. Realize an IIR filter with

$$H(z) = \frac{(z^2-1)(z^2-2z)}{\left(z^2-\frac{1}{2}+j\frac{1}{2}\right)\left(z^2-\frac{1}{2}-j\frac{1}{2}\right)\left(z^2+\frac{1}{16}\right)}$$ in parallel form. **(06 Marks)**

b. Obtain the direct form – I, direct form -II and cascade form realization for the following system. $y(n) = -0.1y(n-1) + 0.2y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2)$. **(09 Marks)**

c. Realize an FIR filter with impulse response h(n) given by $h(n) = \left(\frac{1}{2}\right)^n [u(n)-u(n-5)]$. **(05 Marks)**

7 a. Compare FIR system with IIR system. **(06 Marks)**

b. The desired response of a Low pass filter is

$$H_d(e^{jW}) = \begin{cases} e^{-j3W}, & \frac{-3\pi}{4} \le W \le \frac{3\pi}{4} \\ 0, & \frac{3\pi}{4} \le W \le \pi \end{cases}$$

Determine $H(e^{jW})$ for M = 7 using a Hamming window. **(10 Marks)**

c. What is the need for employing window technique for FIR filter design? **(04 Marks)**

8 a. Design an IIR Butterworth Digital Filter that when used in the Pre – filter A/D – 1 + (z) – D/A structure will satisfy the following analog specifications.

i) LPF with -1dB cut off at 100 π r/s.

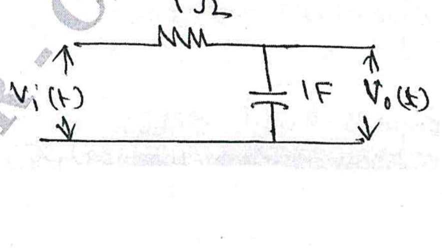
ii) Stop band attenuation of -35 dB or greater at 1000 π r/s.

iii) Monotonic in SB and PB.

iv) Sampling rate 2000 sampler/sec. **(15 Marks)**

b. Obtain the digital filter equivalent of the analog filter shown in Fig. Q8(b) using impulse invariant transformation. Assume $f_s = 8f_c$, where f_c = cut off frequency of the filter. **(05 Marks)**

Fig.Q8(b)



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