USN

10EC52

Fifth Semester B.E. Degree Examination, Dec.2019/Jan.2020
Digital Signal Processing

Time: 3 hrs. Max. Marks:100

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

PART – A

1 a. What do you mean by sampling in frequency domain? Derive the relationship for reconstruction of the signal from the samples of the spectrum. **(10 Marks)**

b. Construct W_3 matrix for finding 3 point DFT. Using this matrix, find the DFT of x(n) = {0, 2, 2}. **(06 Marks)**

c. Establish relationship between DFT and the Fourier series coefficients of a periodic sequence. **(04 Marks)**

2 a. Find the DFT of x(n) = {0, 1, 2, 0} and from DFT of x(n) find the DFT of y(n) = {0, 0, 1, 2} using properties of DFT. Hence prove the property used. **(10 Marks)**

b. Given x(n) = {0, 3, 3} and h(n) = {-1, 1, 1} find the DFTs of x(n) and h(n). Hence calculate circular convolution y(n) between x(n) and h(n) using their DFTs. **(10 Marks)**

3 a. A long sequence x(n) is filtered through a filter with impulse response h(n) to yield the output y(n). If h(n) = {1, 2} and x(n) = {1, 4, 3, 0, 7, 4, -7, -7, -1, 3, 4, 3}, compute y(n) using overlap add technique. Use only a 5 point circular convolution in your approach. **(10 Marks)**

b. Explain computation complexities of Direct DFT calculation and DIT FFT algorithm. Compare the results and calculate the speed improvement factor for N = 64. **(10 Marks)**

4 a. A filter with impulse response h(n) = {1, 1} is given an input x(n) = {0, 2, 4}. Find the output of the filter from the DFTs of h(n) and x(n). Use DIT FFT algorithms to calculate DFT and IDFT. **(10 Marks)**

b. What are the similarities and differences between DIT and DIF, FFT algorithms? **(04 Marks)**

c. Write a note on chirp-z transform. **(06 Marks)**

PART – B

5 a. Describe the transformation relation used for converting a LPF into a HPF. **(06 Marks)**

b. Distinguish between Butterworth and Chebyshev type I filter. **(04 Marks)**

c. Design an analog Chebyshev filter for which the squared magnitude response $|H_a(j\Omega)|^2$ satisfies the condition

$$20\log_{10}|H_a(j\Omega)|_{\Omega=0.2\pi} \ge -1$$

$$20\log_{10}|H_a(j\Omega)|_{\Omega=0.3\pi} \le -15$$

(10 Marks)

6 a. Consider the system function

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

i) Realize the system in direct form – 1
ii) Realize in cascade form
iii) Realize in parallel form. **(12 Marks)**

b. Consider an FIR lattice filter with coefficients $K_1 = 0.65$, $K_2 = -0.34$ and $K_3 = 0.8$. Find its impulse response. Draw the equivalent direct form structure. **(08 Marks)**

7 a. What are the advantages and disadvantages with the design of FIR filter using window function? **(06 Marks)**

b. Explain the following windows with their frequency responses, used in FIR filter design:
i) Rectangular window
ii) Hanning window
iii) Hamming window. **(06 Marks)**

c. Design a lowpass FIR filter using frequency sampling technique having a cutoff frequency of $\frac{\pi}{2}$ rad/sample. The filter should have linear phase and length of 17. **(08 Marks)**

8 a. The system function of the analog filter is given as

$$H_a(s)=\frac{s+0.1}{(s+0.1)^2+9}$$

Obtain the system function of the IIR digital filter by using impulse invariance method. **(06 Marks)**

b. Explain the bilinear transform method of IIR filter design. What is warping effect? Explain the poles and zeros mapping procedure. **(10 Marks)**

c. Compare the impulse invariance and bilinear transform methods. **(04 Marks)**

* * * * *