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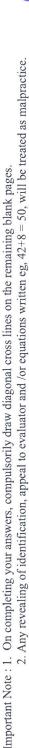# Fifth Semester B.E. Degree Examination, Dec.2019/Jan.2020
Digital Signal Processing

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

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

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

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