EE61

RAMAIAH Institute of Technology

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SEMESTER END EXAMINATIONS – JULY / AUGUST 2022

Program	: B.E: Electrical and Electronics Engineering		Semester	: VI
Course Name	: Digital Signal Processing		Max. Marks	: 100
Course Code	: EE61		Duration	: 3 Hrs

Instructions to the Candidates:

- Answer one full question from each unit.

UNIT – I

1. a) Discuss briefly about digital signal processing and its applications. CO1 (04)

   b) Determine the response of the FIR filter whose impulse response is given as h(n) = (1, -2, 2) when the input applied is x(n)= (1, 2, 1). Use circular convolution and verify the result using linear convolution. CO2 (08)

   c) Given two sequences $x_1(n)$ = (2, 1, 2, 1), $x_2(n)$ = (1,2,3,4). Find x_3 (n) such that $X_3(k) = X_1(k)\, X_2(k)$. CO1 (08)

2. a) Perform the circular convolution of the following two sequences using DFT-IDFT method. Given $x_1(n)$ = (2, 3, 1, 1), $x_2(n)$ = (1,3,5,3) CO1 (08)

   b) The first five points of the 8 point DFT of a real valued sequence are {0.25, 0.125-j0.3018, 0, 0.125-j0.0518, 0}. Determine remaining three points of the DFT. CO2 (06)

   c) Find the 4 point IDFT of the sequence X(k)= (4, -j2,0,j2). CO2 (06)

UNIT – II

3. a) A long sequence is filtered through a filter with impulse response is $h(n) = \{1, 2, -1\}$. Determine the output Y(n) of the filter using overlap add method. Use 4-point circular convolution input $X(n) = \{1, -2, 3, 2, -3, 4, 3, -4\}$ CO2 (08)

   b) Prove the following properties of phase factor (W_N): CO2 (04)

      i. $W_N^{K+N} = W_N^k$

      ii. $W_N^{K+\frac{N}{2}} = -W_N^k$

   c) Explain the computational complexity of DITFFT algorithm compared to direct method. CO2 (08)

4. a) Suppose that we are given a program to find the DFT of a complex-valued sequence $x(n)$. How can we use this program to find the inverse DFT of $X(K)$? CO2 (04)

   b) Design the 8-point DFT of the sequence algorithm $x(n) = \{2,2,2,2,1,1,1,1\}$ use DIT FFT. CO2 (08)

   0 1 2 3 4 5 6 7

   c) Find the 8 point DFT of a real sequence x[n] = (1, 2, 2, 2, 1, 0, 0, 0) using decimation in frequency FFT algorithm. CO1 (08)

UNIT – III

5. a) Obtain a linear phase FIR structure having odd value of impulse samples 'M'. CO3, 4 (10)

   b) Realize a linear phase FIR filter with the impulse response $h(n) = \delta(n) + \frac{1}{4}\delta(n-1) - \frac{1}{8}\delta(n-2) + \frac{1}{4}\delta(n-3) + \delta(n-4)$. Give necessary equations. CO3, 4 (10)

6. a) Discuss any two structures of realizing an IIR system with necessary equations and sketches. CO4 (10)

   b) Obtain DF-I, DF-II and cascade structure of the system represented by the difference equation $y(n) = \frac{3}{4}y(n-1) - \frac{1}{8}y(n-2) + x(n) + \frac{1}{3}x(n-1)$ CO4 (10)

UNIT- IV

7. a) A filter is to be designed with the following desired frequency response: CO3 (08)

   $$Hd(w) = \begin{cases} 0, & -\pi/4 \le w \le \pi/4 \\ e^{-j2w}, & \pi/4 \le |w| \le \pi, \end{cases}$$

   Find the frequency response of the FIR filter designed using a Hamming window.

   b) Design a 17-tap linear-phase FIR filter with a cutoff frequency $w_c = \pi/2$. The design is to be done based on frequency sampling technique. (consider low pass filter). CO3 (08)

   c) Explain Gibb's phenomenon. CO3 (04)

8. a) Obtain the coefficients of an FIR filter to meet the specifications given below using the window method CO4 (10)

   $0.99 \le |Hd(w)| \le 1.01, \quad 0 \le |w| \le 0.375\pi$

   $|Hd(w)| \le 0.01, \quad 0.5\pi \le |w| \le \pi.$

   b) An equiripple high pass filter of order N=64 is to be designed. The stopband ripple is to be no larger than $\delta_s = 0.001$ & the passband ripple no larger than $\delta_p = 0.01$. If a passband cut-off frequency $w_p = 0.72\pi$. Find approximate value of stop band cut off frequency. CO4 (10)

UNIT – V

9. a) Design a chebyshev analog lowpass filter that has a -3dB cutoff frequency of 100rad/sec and a stop band attenuation of 25dB (or) greater for all radian frequencies past 250rad/sec. CO1, 3,4 (08)

   b) Design an analog Butter worth filter that has -3dB better cutoff frequency of 20rad/sec and at least -10dB attenuation at 30rad/sec CO1, 3,4 (06)

   c) Consider a fifth order lowpass butterworth filter with a passband of 1 KHz and maximum passband attenuation of 1 db. Find the actual attenuation in db, of the lowpass filter at a frequency of 2 KHz. CO1, 3,4 (06)

10. a) A second order butterworth low pass filter with a half power frequency of 1 rad/sec is converted to a digital filter h(z) using the bilinear transformation at a sampling rate 1/T=1Hz CO4 (06)

      i. what is the transfer function H(s) of the analog filter
      ii. what is the transfer function H(z) of the digital filter
      iii. c. DC gains of H(z) and H(s) is identical?

   b) Explain TMS320LF2407 digital signal controller. CO5 (06)

   c) A digital IIR lowpass filter is required to meet the following frequency domain specifications. CO4 (08)

      i. Passband ripple: ≤ 1 db
      ii. Passband edge frequency: 0.33π rad
      iii. Stopband attenuation: ≥ 40 db
      iv. Stopband edge frequency: 0.5π rad

   The digital filter is to be designed by applying bilinear transformation on an analog system function. Determine the order, N of butterworth and chebyshev filters needed to meet the specifications in the digital implementation.

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