Reg. No. :

Question Paper Code : 40451

B.E./B.Tech. DEGREE EXAMINATIONS, NOVEMBER/DECEMBER 2021.

Fifth Semester

Electronics and Communication Engineering

EC 8553 — DISCRETE-TIME SIGNAL PROCESSING

(Common to : Biomedical Engineering/Computer And Communication Engineering/ Electronics and Telecommunication Engineering/Medical Electronics)

(Regulations 2017)

Time : Three hours Maximum : 100 marks

Answer ALL questions.

PART A — (10 × 2 = 20 marks)

1. Compute the circular convolution for the given sequence $x(n)=\{1,2,5,6\}$ and $h(n)=\{1,0,-1,-2\}$.

2. The number of points is given by N=64. Compute the number of complex multiplications and additions required to perform DFT and FFT.

3. Calculate the Butterworth polynomial of a Low pass filter with order N=3 and cut off frequency of $\Omega c=1$ rad/sec.

4. Give the significance of impulse invariant method.

5. Define Gibbs Phenomenon.

6. Draw the direct form realization for the following linear phase filter $h(n)=\{1,2,3,4,3,2,1\}$.

7. What do you infer from overflow error?

8. Differentiate between fixed point and floating point number representation.

9. What is the need for pipelining in digital signal processors?

10. What is the difference between Harvard and Von Newman architecture?

PART B — ($5 \times 13 = 65$ marks)

11. (a) Compute 8-point DFT of a sequence $x(n)$ = {1, 3, 6, 8, –3, –7, –9, 1}. Use DIT-FFT algorithm. Also compare DIT-FFT and DIF Algorithms.

Or

(b) Find the output $y(n)$ for the given input sequence $x(n)$ = {1, 2, 3, 4, 5, 6, 7, 8, 9, 1, 11, 8, 9, 12, 14, –8, 3, 6, 44} and $h(n)$={1, 2, 1} Using overlap add method. Also give the comparison between overlap add and overlap save method.

12. (a) Design an analog Chebyshev Type-I LPF that has -3dB passband attenuation at 4.8kHz and –16dB stopband attenuation at 6kHz. Use bilinear transformation and find its digital filter transfer function H(z) with period T = 1 sec.

Or

(b) Obtain the direct form-I, direct form-II, cascade and parallel structure for the following system.

$$y(n)=\frac{3}{4}y(n-1)-\frac{1}{8}y(n-2)+x(n)+\frac{1}{3}x(n-1)$$

13. (a) Determine the filter coefficients $h(n)$ of a linear phase FIR filter of length 15 which has a symmetric unit sample response and a frequency response that satisfies the condition.

$$H_r\left(\frac{2\pi k}{15}\right)=\begin{cases}1, & k=0,1,2,3\\ 0, & k=4,5,6,7\end{cases}.$$

Or

(b) Design an FIR linear phase digital filter approximating the ideal frequency response.

$$H_d(\omega)=\begin{cases}1 \ for & |\omega|\le\frac{\pi}{6}\\ 0 \ for & \frac{\pi}{6}<|\omega|\le\pi\end{cases}$$

Determine the coefficients of a 25-tap filter using hamming window.

14. (a) Explain in detail about the three quantization error with relevant mathematical expressions.

Or

(b) Discuss in detail about limit cycle oscillations due to product quantization and summation with an example.

15. (a) With neat function block diagram, elaborate in detail about any one of the latest DSP architectures.

Or

(b) Explain how programming is done in digital signal processors. Also explain any one application.

PART C — (1 × 15 = 15 marks)

16. (a) Design an analog Butterworth LPF that has –2dB passband attenuation at 3.184 Hz and –10dB stopband attenuation at 4.78 Hz. Analyse how HPF is designed from LPF.

Or

(b) Design a linear phase FIR filter using Fourier series method. Analyze any one real time application of FIR filter.

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Reg. No. :

Question Paper Code : X10368

B.E./B.Tech. DEGREE EXAMINATIONS, NOVEMBER/DECEMBER 2020
Fifth Semester
Electronics and Communication Engineering
EC 8553 – DISCRETE-TIME SIGNAL PROCESSING
(Common to Biomedical Engineering/Electronic Telecommunication Engineering/Medical Electronics)
(Regulations 2017)

Time : Three Hours | Maximum : 100 Marks

Answer ALL questions

PART – A **(10×2=20 Marks)**

1. State and prove the circular time shift property of DFT.
2. Justify the statement, "with zero padding, Discrete Fourier Transform can be used to perform linear filtering".
3. Using Bilinear transform obtain H(z) if $H(s) = \frac{1}{(s+1)^2}$ and the sampling period T = 0.1s.
4. Draw the frequency response of a digital Butterworth low pass filter with a cut-off frequency of 2 rad/sec.
5. What is the Gibb's phenomenon ? Show how it can be reduced by using smooth windowing function in the design of FIR filters ?
6. Write the Hamming Window function and outline its characteristic features.
7. Express – 0.125 in floating point binary representation.
8. Outline the characteristics of error in product quantization.
9. List the merits of instruction pipelining.
10. Compare fixed point and floating point DSP processors.

PART – B **(5×13=65 Marks)**

11. a) i) Obtain the response of a digital filter having the impulse response h(n) = {1, 2, 4} to the input sequence x(n) = {1, 2}. **(7)**

ii) Compute the DFT of $x(n) = \cos(n\pi/4); 0 \le n \le 7$ using DIT-FFT algorithm. **(6)**

(OR)

b) i) Obtain the output y(n) of a filter whose impulse response is h(n) = {1, 1, 1} and input signal x(n) = {3, –1, 0, 1, 3, 2, 0, 1, 2, 1} using overlap-save method. **(7)**

ii) Given X(k) = {36, –4 + j9.656, –4 + j4, –4 + j1.656, –4, – 4 – j1.656, – 4 – j4, – 4 – j9.656}, find x(n), using DIF-FFT algorithm. **(6)**

12. a) i) Determine the transfer function of the second order normalized analog Chebyshev low pass filter. **(7)**

ii) Determine the direct form I and II realization for a third-order IIR transfer function. **(6)**

$$H(z) = \frac{(0.28z^2 + 0.3z + 0.04)}{(0.5z^3 + 0.3z^2 + 0.7z - 0.2)}$$

(OR)

b) Determine H(z) for a Butterworth filter satisfying the following constraints. **(13)**

$$\sqrt{0.5} \le \left|H(e^{j\omega})\right| \le 1 \qquad 0 \le \omega \le \pi/2$$

$$\left|H(e^{j\omega})\right| \le 0.2 \quad 0.75\pi \le \omega \le \pi$$

With T = 1s. Apply impulse invariant transformation.

13. a) The desired frequency response of a low pass filter is **(13)**

$$H_d(e^{j\omega}) = \begin{cases} 1, -\pi/2 \le \omega \le \pi/2 \\ 0, \ \pi/2 \le \omega \le \pi \end{cases}$$

Determine $h_d(n)$. Also determine h(n) using the symmetric rectangular window, with window length = 7.

(OR)

b) i) Use the Fourier series method to design a low pass digital filter to approximate the ideal specifications given by

$$H(e^{j\omega}) = \begin{cases} 1, & \text{for } |f| \le f_p \\ 0, & f_p \le |f| \le F/2 \end{cases}$$, where f_p is the passband frequency and F is the sampling frequency. **(7)**

ii) Obtain FIR linear phase and cascade realizations of the system function. **(6)**

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

14. a) Discuss the effects of finite word length in the implementation of the FFT algorithms using fixed point arithmetic. **(13)**

(OR)

b) Explain the effects of coefficient quantization in Direct Form Realization of IIR filter. **(13)**

15. a) i) Illustrate the addressing modes of DSP processors. **(7)**

ii) Sketch the structure of the MAC unit and DSP processor and explain its functions. **(6)**

(OR)

b) i) Explain the architecture of fixed point and floating point DSP processors. **(7)**

ii) With suitable diagrams show how to implement FIR filter in DSP processor. **(6)**

PART – C **(1×15=15 Marks)**

16. a) The first order filter shown in Fig. 1 below is implemented in four-bit (including sign bit) fixed point two's complement fractional arithmetic. Products are rounded to four-bit representation using the input $x(n) = 0.10\delta(n)$. Determine

i) the first five outputs if $\alpha = 0.5$. Does the filter go into a limit cycle ?

ii) the first five outputs if $\alpha = 0.75$. Does the filter go into a limit cycle ?

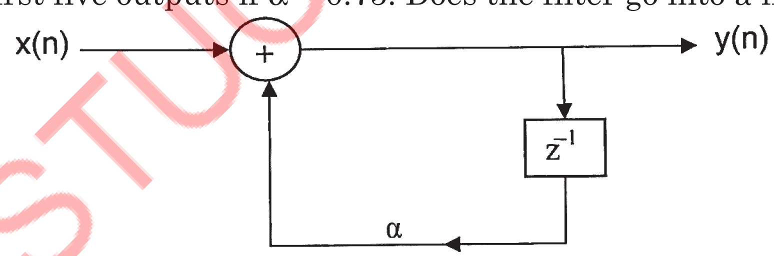

Fig. 1

(OR)

b) i) Express the magnitude response of an FIR filter of length 11 exhibiting the linear phase property. **(7)**

ii) A band reject FIR filter of length seven is required. It is to have lower and upper cut-off frequencies as 3 KHz and 6 KHz respectively. The sampling frequency is 18 KHz. Determine the filter co-efficients using Hanning window. Draw the structure of the filter. **(8)**

15. a) Sketch the architecture of a DSP processor and explain its special features that support signal processing applications. (13)

(OR)

b) i) Explain the concept of circular buffering in a DSP processor. Explain how this is useful in implementation of FIR filters. (7)

ii) Comment on the factors that decide the choice of a DSP processor for a system. (6)

PART – C (1×15 = 15 Marks)

16. a) Given, $H_d(e^{j\omega}) = \begin{cases} e^{-j3\omega}, & -\pi/4 \le \omega \le \pi/4 \\ 0, & \text{otherwise} \end{cases}$. Design a FIR filter using Hamming window with N = 7. (15)

(OR)

b) Propose a DSP based system to process signals in an audio system that includes recording, storage, transmission and reproduction of signals and explain. (15)

Reg. No. :

Question Paper Code : 90190

B.E./B.Tech. DEGREE EXAMINATIONS, NOVEMBER/DECEMBER 2019

Fifth Semester

Medical Electronics

EC 8553 – DISCRETE-TIME SIGNAL PROCESSING

(Common to Biomedical Engineering/Computer and Communication Engineering/ Electronics and Communication Engineering/Electronics and Telecommunication Engineering)

(Regulations 2017)

Time : Three Hours Maximum : 100 Marks

Answer ALL questions

PART – A (10×2 = 20 Marks)

1. Determine the IDFT of Y(K) = {1, 0, 1, 0}.

2. Draw the 4-point radix 2 DIT-FFT and 4-point radix 2 DIF-FFT butterfly structures for FFT.

3. Summarize the procedure to design digital filters from analog filters. Recall in this context what is meant by backward difference.

4. What is bilinear transformation ? List the properties of bilinear transformation.

5. Obtain the direct form realization of the filter $H(Z) = \frac{1}{2} + \frac{1}{4}z^{-1} + \frac{1}{4}z^{-2} + \frac{1}{2}z^{-3}$ with minimum number of multipliers.

6. How to design an FIR filter using frequency sampling method ? For what type of filters frequency sampling method is suitable ?

7. Define input quantization error and product quantization error.

8. Interpret how the digital filter is affected by quantization of filter coefficients ?

9. Distinguish between fixed point and floating point arithmetic.

10. List the applications of Digital Signal Processing.

PART – B (5×13 = 65 Marks)

11. a) i) Compute the DFT of the sequence x(n) = {0, 1, 2, 1}. Sketch the magnitude and phase spectrum. (7)

ii) For the given $x_1(n)$, $x_2(n)$ and N, compute the circular convolution of $x_1(n)$ and $x_2(n)$. (6)

1) $x_1(n) = \delta(n) + \delta(n-1) + \delta(n-2)$, N = 3

$x_2(n) = 2\,\delta(n) - \delta(n-1) + 2\,\delta(n-2)$

2) $x_1(n) = \delta(n) + \delta(n-1) + \delta(n-2) - \delta(n-3)$, N = 5

$x_2(n) = \delta(n) - \delta(n-2) + \delta(n-4)$.

(OR)

b) i) Perform Linear convolution of the following sequences by overlap-add method. (7)

x(n) = {1, – 2, 3, 2, – 3, 4, 3, – 4} and h(n) = {1, 2, – 1}.

ii) Compute the 8 point DFT of the sequence x(n) = {1, 1, 1, 1, 1, 1, 1, 0} using DIT, FFT algorithm. (6)

12. a) For the given specifications $0.9 \le |H(j\Omega)| \le 1$, for $0 \le \Omega \le 0.2\pi$

$|H(j\Omega)| \le 0.2$, for $0.4\pi \le \Omega \le \pi$

Plot the magnitude response and design an analog Butterworth filter. (13)

(OR)

b) i) For the analog transfer function $H_a(s) = \frac{2}{(s+1)(s+3)}$. Determine H(z), if T = 1s, using Impulse invariant method. (7)

ii) Realize the system with difference equation (6)

$$y(n) = \frac{3}{4}y(n-1) - \frac{1}{8}y(n-2) + x(n) + \frac{1}{3}x(n)$$ In cascade form.

13. a) Discuss on the frequency response of linear phase FIR filter when the impulse response is symmetrical and N is odd. (13)

(OR)

b) List the steps in the design of FIR filter using windows. Point out the characteristics of the following window functions. (13)

i) Rectangular Window

ii) Hanning Window

iii) Hamming Window.

14. a) i) A digital system is characterized by the difference equation y(n) = 0.8y(n – 1) + x(n). Determine the limit cycle behavior and the dead band of the system with x(n) = 0 and initial condition y(– 1) = 10. Assume that the result y(n) is rounded off to the nearest integer. (7)

ii) Given $H(Z) = \frac{0.5 + 0.4z^{-1}}{1 - 0.312z^{-1}}$ is the transfer function of a digital filter.

Find the scaling factor S_0 to avoid overflow in adder 1 of the digital filter shown in fig. 1. (6)

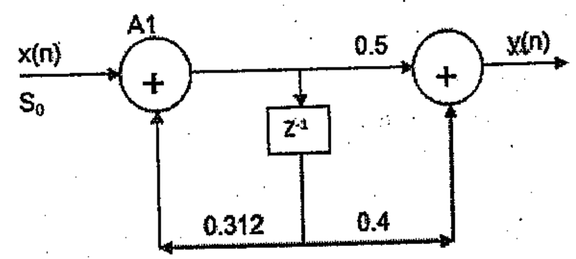

Fig. 1

(OR)

b) Discuss the effect of coefficient quantization on pole locations of the following IIR system, when it is realized in direct form – 1.

$H(Z) = \frac{1}{1 - 0.7z^{-1} + 0.12z^{-2}}$. Assume a word length of 4-bits through truncation. (13)