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 Ωc=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.

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

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.