MUTHAYAMMAL ENGINEERING COLLEGE

(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu.

Estd. 2000

Department of Electronics and Communication Engineering
Question Bank - Academic Year (2021-22)

Course Code & Course Name : **19ECD09 & Digital Signal Processing**

Year/Sem/Sec : **II/IV/**

Unit-I: Discrete Fourier Transform

Part-A (2 Marks)

1. Obtain the circular convolution of the following sequences x(n) = {1,2,1}; h(n) ={1, -2,2}
2. Define DFT and IDFT.
3. State the advantages of FFT over DFTs.
4. What is meant by bit reversal?
5. What is zero padding?
6. Find the 4 – point DFT sequence x(n) = {1, 1, -1, -1}
7. What are the differences between Overlap – add and Overlap – save method?
8. Distinguish between linear convolution and circular convolution?
9. State Parseval's relation with respect to DFT.
10. Draw the butterfly diagram for decimation in frequency FFT algorithm.

Part-B (16 Marks)

1. (i).Explain Radix – 2 DIF FFT algorithm. Compare it with DIT – FFT algorithms. (8)

   (ii).Compute the linear convolution of finite duration sequences *h(n)* = {1,2} and *x (n)* = {1, 2, -1, 2, 3, - 2, -3, -1, 1, 1, 2, -1} by Overlap add method? (8)

2. Compute the eight point DFT of the sequence by using the DIT and DIF – FFT algorithm. (16)

$$x(n) = \begin{cases} 1 & 0 \le n \le 7 \\ 0 & Otherwise \end{cases}$$

3. Explain Radix – 2 DIF FFT algorithms. Compare it with DIT – FFT algorithms. (16)

4. (i).Find the IDFT of the sequence X(K)= {6, -2+2j, -2, -2-2j} using Radix 2 DIF algorithm. (8)

   (ii).Compute an 8 point DFT of the sequence. (8)

$$x(n) = \{1,\ 0,\ \underset{\uparrow}{1},\ -1,\ 1,\ 1,\ 0,\ 1\}$$

5. (i).Summarize the Difference between overlap – save method and overlap – add method. (8)

(ii).Summarize the properties of DFT. (8)

Unit-II : Infinite Impulse Response Filters

Part-A (2 Marks)

1. Compare Butterworth with Chebyshev filters
2. What is meant by warping effect?
3. List out the properties of Chebyshev filter
4. Draw the direct form structure of IIR filter
5. Why the Butterworth response is called a maximally flat response?
6. What is the advantage of direct form II realization when compared to direct form I realization?
7. Compare IIR and FIR filters
8. Why impulse invariance method is not preferred in the design of FIR filter?
9. Sketch the frequency response of an odd and even order Chebyshev low pass filters
10. What is bilinear transformation? What is the main advantages and disadvantages of this technique?

Part-B (16 Marks)

1. (i).Explain the procedure for designing analog filters using the Chebyshev approximation. (8)

(ii).Convert the following analog transfer function in to digital using impulse invariant mapping with T=1sec H(s) = $\frac{3}{(s+3)(s+5)}$ (8)

2. Obtain the direct form I, direct form II ,cascade and Parallel form realization of the following system functions. (16)

y(n)=0.1y(n-1)+0.2y(n-2)+3x(n)+3.6x(n-1)+0.6x(n-2)

3. Explain the bilinear transform method of IIR filters design. What is wrapping effect? Explain the poles and zeros mapping procedure clearly. (16)

4. (i).Design a digital second order low pass Butterworth filter with cut off frequency 2200 Hz using bilinear transformation. Sampling rate is 8000 Hz. (8)

(ii).Determine the cascade form and parallel form implementation of the system governed by the transfer function H(s) = $\frac{1+Z^{-1}}{1+2Z^{-1}}$ (8)

5. Apply Bilinear Transformation and Impulse invariant to H(s) =2/(S+2) (S+3) with T=0.1 sec. (16)

Unit-III : Finite Impulse Response Digital Filters

Part-A (2 Marks)

1. Give the equations of Hamming window and Blackman Window.
2. State the properties of FIR filter
3. What is meant by Gibbs Phenomenon?
4. Determine the transversal structure of the system function

   $H(z) = 1 + 2Z^{-1} - 3Z^{-2} - 4Z^{-3}$
5. What are the features of FIR filter design using Kaiser's approach?
6. What are the techniques of designing FIR filters?
7. State the effect of having abrupt discontinuity in frequency response of FIR filters.
8. What is the principle of designing FIR filter using frequency sampling method?
9. Write the steps involved in FIR filter design
10. What are the possible types of impulse response for linear phase FIR filters?

Part-B (16 Marks)

1. Design a high pass filter with a frequency response

   $$H_d(e^{j\omega}) = \begin{cases} 1 & for \quad \frac{\pi}{2} \le |\omega| \le \pi \\ 0 & for \quad |\omega| \le \frac{\pi}{4} \end{cases}$$ (16)

   Find the values of *h(n)* for N = 11 using hamming window. Find *H(z)* and determine the magnitude response.
2. (i).Realize the system function by linear phase FIR structure (8)

   $H(Z)=2/3\ Z^{-2}+2/3\ Z^{-1}+1$

   (ii).Explain the designing of FIR filters using windows? (8)
3. Design a FIR low pass filter having the following specifications using Hanning

   $$H_d(e^{j\omega}) = \begin{cases} 1 & for \quad -\frac{\pi}{6} \le |\omega| \le \frac{\pi}{6} \\ 0 & for \quad otherwise \end{cases}$$ (16)

   Windows Assume N = 7
4. Design a low pass filter with pass band gain of unity cut off frequency of 1000Hz and working at a sampling frequency of 5 kHz. The length of the impulse response should be 7 (16)
5. Consider an FIR lattice filter with coefficients $k1 = 1/2$; $k2 = 1/3$; $k3 = 1/4$. Determine the FIR filter coefficients for the direct form structure (16)

Unit-IV : Finite Word Length Effects

Part-A (2 Marks)

1. Define limit cycle oscillator.
2. What is dead – band of a filter?
3. Definition of overflow oscillations?
4. What is input quantization error?
5. Define Zero Input limit cycle.
6. State the need for scaling in filter implementation.
7. What differences exist between the three-quantization errors to finite word length registers in digital filters?
8. Why do you think rounding is preferred to truncation in realizing digital filter?
9. Compare the fixed point and floating point arithmetic.
10. Characterize Noise transfer function.

Part-B (16 Marks)

1. Discuss in detail the errors resulting from rounding and truncation? (16)
2. (i) Explain the limit cycle oscillations due to product round off and overflow errors? (8)

   (ii) Explain how reduction of product round-off error is achieved in digital filters? (8)
3. (i) Derive the signal to quantization noise ratio of A/D converter. (8)

   (ii) Compare the truncation and rounding errors using fixed point and floating point representation. (8)
4. Determine the dead band of the system $y(n) = 0.2y(n-1) + 0.5y(n-2) + x(n)$ Assume 8 bits are used for signal representation. (16)
5. What is called quantization noise? Derive the expression for quantization noise power. (16)

Unit-V : Multi rate Signal Processing

Part-A (2 Marks)

1. What is decimation?
2. What is Sub band coding?
3. State the various applications of adaptive filters.
4. What is anti – imaging filter?
5. Give the applications of multi rate DSP

6. What is anti aliasing filter?
7. What are the advantages of multi rate processing?
8. What is interpolator? Draw the symbolic representation of an interpolator.
9. Give the steps in multistage sampling rate converter design
10. What is echo cancellation?

Part-B (16 Marks)

1. Explain the poly phase structure of decimator and interpolator? (16)
2. Discuss the procedure to implement digital filter bank using multi rate signal processing. (16)
3. Explain sampling rate conversion by a rational factor and derive input and output relation in both time and frequency domain. (16)
4. (i). A signal $x(n)$ is given by $x(n) = \{0,1,2,3,4,5,6,0,1,2,3\ldots.\}$. Obtain the decimated signal with a factor of 2. (8)

   (ii).Explain the various applications of adaptive filters? State the applications of multirate signal processing? (8)
5. (i).Explain sub band coding in detail. (8)

   (ii).Explain the design steps involved in the implementation of multistage sampling rate converter. (8)

Course Faculty **HoD**