PRATHYUSHA ENGINEERING COLLEGE

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DEPARTMENT OF

ELECTRONICS AND COMMUNICATION ENGINEERING

QUESTION BANK

V SEMESTER

EC8553 – DISCRTE TIME SIGNAL PROCESSING

Regulation – 2017

Academic Year 2019 – 20

UNIT I

DISCRETE FOURIER TRANSFORM

PART – A

1. How many multiplications and additions are required to compute N –point DFT using radix – 2 FFT?
2. State the advantages of FFT over DFTs?
3. Distinguish between DFT and DTFT?
4. What is zero padding? What are its uses?
5. What is twiddle factor?
6. What are the differences between Overlap – add and Overlap – save method?
7. State the properties of DFT.
8. Draw the basic butterfly diagram for the computation in the decimation in frequency FFT algorithm and explain.
9. How will you perform linear convolution using circular convolution?
10. State Parseval's relation with respect to DFT?
11. Write down the relation between z-transfrom & DFT.
12. What is meant by in – place computation?

PART - B

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

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

   (ii) Summarize the properties of DFT

2. (i) Explain the Overlap add and Overlap save method

   (ii) Compute the eight point DFT of the sequence

   $$x(n) = \left\{\frac{1}{2}, \frac{1}{2}, \frac{1}{2}, \frac{1}{2}, 0, 0, 0, 0\right\}$$

   Using radix – 2 DIT algorithm.

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

   (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 & Overlap save method.

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

   (ii) Compute an 8 point DFT of the sequence

   x(n)={0,1,2,3,4,5,6,7}

5. (i) Determine the N – point DFT of the following sequences

   (a) x(n) = δ(n) (b) x(n) = δ(n-n_0)(ii)Compute 8 – point DFT of the sequence x(n) = {0,1, 2, 3, 4,5, 6, 7} using radix – 2 DIF algorithm?

6. (i) Compute the 8 point DFT for the following sequences using DIT – FFT algorithm

   $$x(n) = \begin{cases} 1 & -3 \le n \le 3 \\ 0 & Otherwise \end{cases}$$

   (ii) Compute the IDFT of the sequence {1,1,1,0}.

7. (i) Compute the DFT of the sequence whose values are given by x(n) = {1,1,-2,-2}

   (ii) Compute the eight point DFT of the sequence x(n)= {1,2,3,4,4,3,2,1} using Radix-2 DIT algorithm.

UNIT II

INFINITE IMPULSE RESPONSE FILTERS

PART – A

1. Compare Butterworth with Chebyshev filters.
2. What is known as pre warping in digital filters?
3. Draw the direct form structure of IIR filter.
4. Why do we go for analog approximation to design a digital filter?
5. What is the advantage of direct form II realization when compared to direct form I realization?
6. Mention the advantages of cascade realization.
7. Give the steps in design of a digital filter from analog filters.
8. Compare IIR and FIR filters.
9. Distinguish between recursive realization and non recursive realization?
10. What is meant by impulse invariance method of designing IIR filter?

PART – B

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

ii) Convert the following analog transfer function in to digital using impulse invariant mapping with T=1sec.

$$H(s) = \frac{3}{(S+3)(S+5)}$$

2. (i) Design a digital Butterworth filter using impulse invariance method satisfying the constraints Assume T=1sec.

$$0.8 \leq \left|H(e^{jw})\right| \leq 1 \qquad 0 \leq w \leq 0.2\pi$$

$$\left|H(e^{jw})\right| \leq 0.2 \qquad 0.6\pi \leq w \leq \pi$$

(ii)Obtain the direct form I , direct form II and cascade form realization of the following system functions

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

3. Determine the system function H(z) of the chebyshev low pass digital filter with the specifications
   1. α_p =1dB ripple in the pass band $0 \leq w \leq 0.2\pi$
   2. α_s=1dB ripple in the stop band $0.3\pi \leq w \leq \pi$

Using bilinear transformation (assume T=1sec)

4. (i)Explain the bilinear transform method of IIR filter design. What is wrapping effect? Explain the poles and zeros mapping procedure clearly.
   (ii)Explain the procedure for designing Butterworth filter using IIV method.

5. i) Obtain the cascade form realization of the digital system

$$y(n)=3/4\ y(n-1)-(1/8)y(n-2)+1/3x(n-1)+x(n)$$

   ii) Convert the given analog filter with transfer function $H(s) = \frac{2}{(s+1)(s+2)}$ in to a digital IIR filter using bilinear transformation. Assume T=1sec.

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

7. (ii)Design a low pass Butterworth digital filter with the following specification :

   Ws= 4000, Wp= 3000

   Ap= 3 d B, As= 20 d B, T= 0.0001 sec.

8. Obtain the cascade and parallel realizations for the system function given by $H(z) = \frac{1+\frac{1}{4}Z^{-1}}{(1+\frac{1}{2}Z^{-1})(1+\frac{1}{2}Z^{-1}+\frac{1}{4}Z^{-2})}$

UNIT III
FINITE IMPULSE RESPONSE DIGITAL FILTERS
PART – A

1. Give the equations of Hamming window and Hanning Window.
2. Determine the transversal structure of the system function

$$H(z) = 1 + 2Z^{-1} - 3Z^{-2} - 4Z^{-3}$$

3. State the properties of FIR filter.
4. What is meant by Gibbs Phenomenon?
5. What are the desirable characteristics of window?
6. Draw the direct form implementation of the FIR system having difference equation.

$$y(n) = x(n) - 2x(n-1) + 3x(n-2) - 10x(n-6)$$

7. What are the techniques of designing FIR filters?
8. What is the principle of designing FIR filter using windows?
9. What are advantages and disadvantages of FIR filter?

10. What is the reason that FIR filter is always stable?

PART - B

1. (i)Design a high pass filter with a frequency response

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

Find the values of *h(n)* for N = 11 using hamming window. Find *H(z)* and determine the magnitude response.

(ii) Obtain the linear phase realization of the system function.

$$H(z) = \frac{1}{2} + \frac{1}{3}z^{-1} + z^{-2} + \frac{1}{4}z^{-3} + z^{-4} + \frac{1}{3}z^{-5} + \frac{1}{2}z^{-6}$$

2. (i)Explain the designing of FIR filters using windows.

(ii)Obtain the frequency response linear phase FIR filter using symmetric condition when

N is odd& anti-symmetric condition when N is even.

3. Design an ideal high pass filter using Hanning window with a frequency response

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

Assume N = 11.

4. (i) Design a FIR low pass filter having the following specifications using Hanning window

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

Assume N = 7

6. (i) Explain with neat sketches the implementation of FIR filters in direct form and Lattice form.

(ii) Design a digital FIR band pass filter with lower cut off frequency 2000Hz and upper cut off frequency 3200 Hz using Hamming window of length N = 7. Sampling rate is 10000Hz. 1

7. (i)Determine the frequency response of FIR filter defined by

$$y(n) = 0.25x(n) + x(n-1) + 0.25x(n-2)$$

(ii) Discuss the design procedure of FIR filter using frequency sampling method.

8. Design an FIR filter using Hanning window with the following specification

$$H_d(e^{j\omega}) = \begin{cases} 1 & for \quad \frac{\pi}{4} \le \omega \le \frac{\pi}{4} \\ e^{-j2\omega} & for \quad \frac{\pi}{4} \le |\omega| \le \pi \end{cases}$$

Assume N = 5.

9. (i) Using a rectangular window technique, 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.

ii) Consider an FIR lattice filter with coefficients $k_1 = 1/2$; $k_2 = 1/3$; $k_3 = 1/4$. Determine the FIR filter coefficients for the direct form structure.

10. A low pass filter has the desired response as given below, Determine the filter co efficient h(n) for M=7,using type 1 frequency sampling technique.

$$H_d(e^{j\omega}) = \begin{cases} e^{-j3\omega}, & 0 \le \omega < \frac{\pi}{2} \\ 0, & \frac{\pi}{2} \le \omega \le \pi \end{cases}$$

UNIT – IV

FINITE WORD LENGTH EFFECTS

PART – A

1. What is truncation?
2. What is product quantization error?
3. What is overflow oscillations?
4. What are the advantages of floating point arithmetic?
5. What is dead – band of a filter?
6. What do you understand by input quantization error?
7. What is quantization step size?
8. What are limit cycles?

9. What is meant by block floating point representation? What are its advantages? 1

10. What is coefficient quantization error? What is its effect?

11. Why rounding is preferred to truncation in realizing digital filter?

12. State the need for scaling in filter implementation.

PART – B

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

   (ii) Explain how reduction of product round-off error is achieved in digital filters?
3. (i) Explain the effects of co-efficient quantization in FIR filters?

   (ii) Distinguish between fixed point and floating point arithmetic.

4. With respect to finite word length effects in digital filters, with examples discuss about
   (i) Over flow limit cycle oscillation
   (ii) Signal scaling
5. Consider a second order IIR filter with

$$H(z) = \frac{1.0}{(1 - 0.5Z^{-1})(1 - 0.45Z^{-1}).}$$

   Find the effect on quantization on pole locations of the given system function in direct form and in cascade form. Assume $b = 3$ bits.

6. (i) What is called quantization noise? Derive the signal to quantization noise ratio of A/D converter.

   (ii) Represent the following numbers in floating point format with five bits for mantissa and three bits for exponent.

   (a) 7_{10}

   (b) 0.25_{10}

   (c) -7_{10}

   (d) -0.25_{10}

7. 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. 1

8. (a) i) Explain the characteristics of limit cycle oscillation with respect to the system described by the difference equation : y(n) = 0.95 y(n-1) + x (n) ; x(n)= 0 and y(n-1)= 13. Determine the dead range of the system.

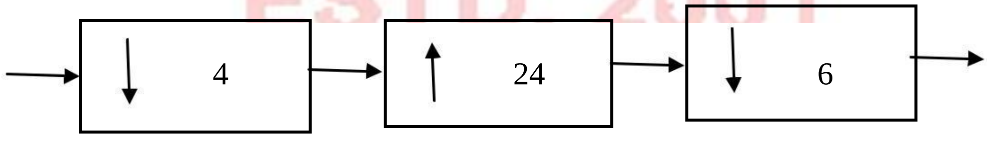
   ii) Explain the effects of coefficient quantization in FIR filters.

UNIT – V

MULTI RATE SIGNAL PROCESSING

PART – A

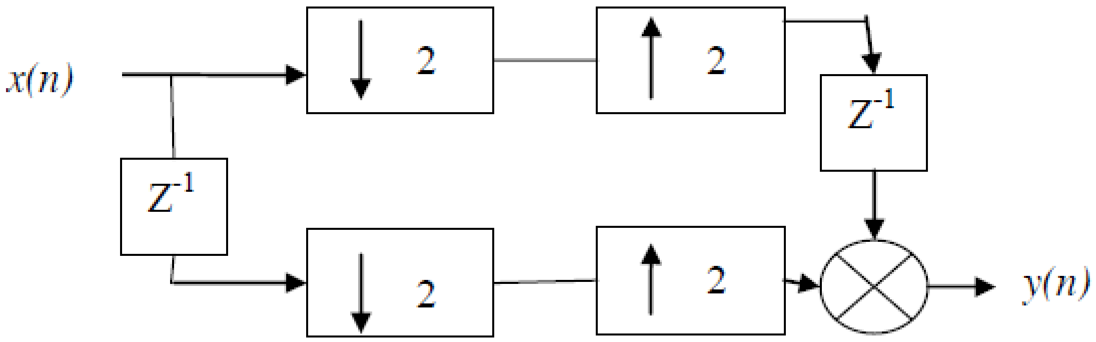
1. What is decimation? Draw the symbolic representation of a decimator.
2. What is Sub band coding?
3. State the various applications of adaptive filters.
4. What is echo cancellation?
5. What is multi rate signal processing & its advantages?
6. What is meant by down sampling and up sampling?
7. Give the applications of multi rate sampling?
8. What are called poly phase filters?
9. Find the expression for the following multi rate system?



10. Give the applications of multi rate DSP
11. Give the steps in multistage sampling rate converter design.
12. What is interpolator? Draw the symbolic representation of an interpolator?
13. What is anti aliasing filter & anti – imaging filter?

PART – B

1. (i)Explain in detail about applications of adaptive filters for echo cancellation and equalization.
   (ii)Explain subband coding in detail.
2. For the multi rate system shown in figure, find the relation between $x(n)$ and $y(n)$.



3. Explain sampling rate conversion by a rational factor and derive input and output relation in both time and frequency domain
4. (i) Explain the design of narrow band filter using sampling rate conversion.
   (ii)Explain the design steps involved in the implementation of multistage sampling rate converter.
5. Implement a two stage decimator for the following specifications.

   Sampling rate of the input signal= 20,000Hz.
   M=100
   Passband=0 to 40 Hz
   Transition band = 40 to 50 Hz.
   Pass band ripple = 0.01
   Stop band ripple= 0.002.
6. 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.

   Obtain the interpolated signal with a factor of 2.