

**Sixth Semester B.E. Degree Examination, June/July 2019**  
**Digital Signal Processing**

Time: 3 hrs.

Max. Marks: 80

Note: Answer any FIVE full questions, choosing  
 ONE full question from each module.

**Module-1**

1. a. Determine DFT of sequence  $x(n) = \frac{1}{3}$  for  $0 \leq n \leq 2$  for  $N = 4$ . Plot magnitude and phase spectrum. (08 Marks)

- b. Two length - 4 sequence are defined below :

$$x(n) = \cos\left(\frac{\pi n}{2}\right) \quad n = 0, 1, 2, 3$$

$$h(n) = 2^n \quad n = 0, 1, 2, 3$$

- i) Calculate  $x(n) *_4 h(n)$  using circular convolution directly.

- ii) Calculate  $x(n) *_4 h(n)$  using Linear convolution.

(08 Marks)

**OR**

2. a. Compute circular convolution using DFT + IDFT for following sequence :

$$x_1(n) = \begin{cases} 2, & n=0 \\ 3, & n=1 \\ 1, & n=2 \\ 1, & n=3 \end{cases}, \quad x_2(n) = \begin{cases} 1, & n=0 \\ 3, & n=1 \\ 5, & n=2 \\ 3, & n=3 \end{cases}. \quad (08 \text{ Marks})$$

- b. Find the output of the LTI system whose impulse  $h(n) = \{1, 1, 1\}$  and the input signal is  $x(n) = \{3, -1, 0, 1, 3, 2, 0, 1, 2, 1\}$ . Using the overlap save method. Use 6-pt circular convolution. (08 Marks)

**Module-2**

3. a. What are FFT algorithms? Explain the advantages of FFT algorithms over the direct computations of DFT for a sequence  $x(n)$ . (04 Marks)

- b. What are the differences and similarities between DIT and DIF - FFT algorithms? (04 Marks)

- c. Find the 8-pt DFT of the sequence  $x(n) = \{1, 2, 3, 4, 4, 3, 2, 1\}$ . Using DIT - FFT radix - 2 algorithm. (08 Marks)

**OR**

4. a. Find the 4-pt circular convolution of  $x(n)$  and  $h(n)$  given. Using radix-2 DIF - FFT algorithm. (08 Marks)

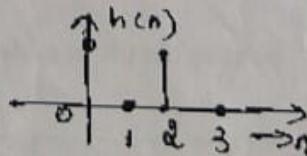
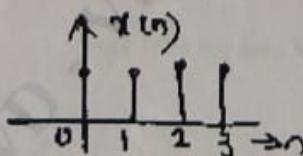


Fig.Q4(a)

- b. Given  $x(n) = (n + 1)$  and  $N = 8$ . Determine  $X(K)$ . Using DIF - FFT algorithm.

(08 Marks)

**Module-3**

- 5 a. Convert the analog filter with system transfer function :

$$H(s) = \frac{(s + 0.1)}{(s + 0.1)^2 + 3^2}$$

into a digital IIR filter by mean of the impulse invariant method. (06 Marks)

- b. Design a butter worth digital IIR lowpass filter using bilinear transformation by taking  $T = 0.1\text{ sec}$ , to satisfy the following specification :

$$\begin{cases} 0.6 \leq |H(e^{j\omega})| \leq 1.0; & \text{for } 0 \leq \omega \leq 0.35\pi \\ |H(e^{j\omega})| \leq 0.1; & \text{for } 0.7\pi \leq \omega \leq \pi \end{cases}$$

(10 Marks)

- 6 a. Compare analog and digital filters.

**OR**

(04 marks)

- b. Determine the poles of lowpass Butterworth filter for  $N = 2$ . Sketch the location of poles on s-plane and hence determine the normalized transfer function of lowpass filter. (08 Marks)

- c. Write difference between IIR and FIR filter. (04 Marks)

- 7 a. Design a Chebyshev digital IIR lowpass filter using impulse invariant transformation by taking  $T = 1\text{ sec}$  to satisfy the following specifications,

$$\begin{cases} 0.9 \leq |H(e^{j\omega})| \leq 1.0; & \text{for } 0 \leq \omega \leq 0.25\pi \\ |H(e^{j\omega})| \leq 0.24; & \text{for } 0.5\pi \leq \omega \leq \pi \end{cases}$$

- b. Draw direct form - I and II structure of the filter.

(12 Marks)

- b. Write the relation between analog and digital frequency in Bilinear transformation. (04 Marks)

- 8 a. Obtain the direct form - I, direct form II realization of the LTI system governed by the relation.

$$y(n) = -\frac{3}{8}y(n-1) + \frac{3}{32}y(n-2) + \frac{1}{64}y(n-3) + x(n) + 3x(n-1) + 2x(n-2). \quad (08 \text{ Marks})$$

- b. Realize the given system in cascade and parallel form :

$$H(z) = \frac{1 + 0.25z^{-1}}{(1 - 2z^{-1} + 0.25z^{-2})(1 - 3z^{-1} + 0.25z^{-2})}$$

(08 Marks)

**15EE63****Module-5**

- 9 a. The frequency response of a filter is described by :  $H(\omega) = j\omega$ ,  $-\pi \leq \omega \leq \pi$ . Design the filter using a rectangular window. Take  $N = 7$ . **(08 Marks)**
- b. Design a lowpass digital filter to be used in A/D –  $H(z)$  – D/A structure that will have – 3dB cutoff at  $30\pi$  rad/sec and attenuation factor of 5dB at  $45\pi$  rad/sec. The filter is required to have a linear phase and the system will use sampling frequency of 100 samples/sec. **(08 Marks)**

**OR**

- 10 a. Deduce the equation for the following frequency spectrum for rectangular window sequence defined by :

$$w_f(n) = \begin{cases} 1, & \frac{-(N-1)}{2} \leq n \leq \frac{N-1}{2} \\ 0, & \text{otherwise} \end{cases} \quad \text{(06 Marks)}$$

- b. A lowpass filter has the desired frequency response :

$$H_d(\omega) = \begin{cases} e^{-j\omega/3}, & 0 < \omega < \pi/2 \\ 0, & \text{otherwise} \end{cases}$$

Determine  $h(n)$  based on frequency sampling method. Take  $K = 7$ . **(06 Marks)**

- c. Realize the linear phase FIR filter having the following impulse response :

$$h(n) = \delta(n) + \frac{1}{4}\delta(n-1) - \frac{1}{8}\delta(n-2) + \frac{1}{4}\delta(n-3) + \delta(n-4). \quad \text{(04 Marks)}$$

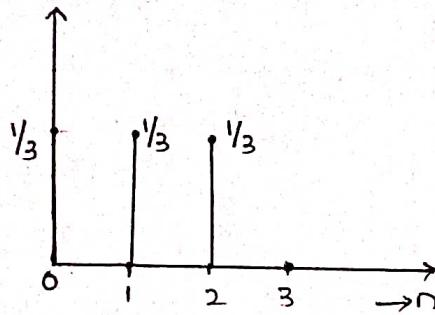
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## DIGITAL SIGNAL PROCESSING

## MODULE - I

1a) Determine DFT of sequence  $x(n) = \frac{1}{3}$  for  $0 \leq n \leq 2$ . for  $N=4$ .  
 Plot magnitude and phase spectrum (8M)

$\Rightarrow$  Given  $x(n) = \frac{1}{3}$  for  $0 \leq n \leq 2$ ,  $N=4$



$$\therefore x(n) = \left\{ \frac{1}{3}, \frac{1}{3}, \frac{1}{3}, 0 \right\}$$

Using linear transformation,

$$[x(k)] = [W_N^{kn}] [x(n)]$$

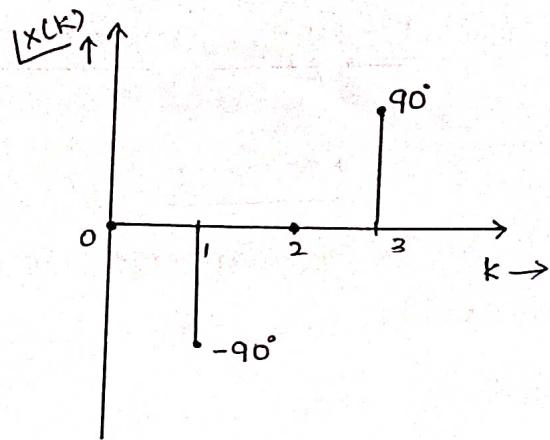
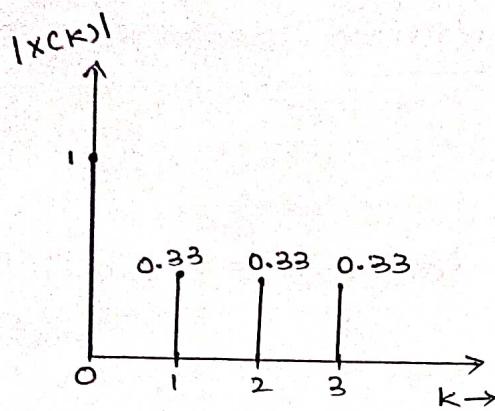
$$\begin{bmatrix} x(0) \\ x(1) \\ x(2) \\ x(3) \end{bmatrix} = \begin{bmatrix} W_4^0 & W_4^0 & W_4^0 & W_4^0 \\ W_4^1 & W_4^1 & W_4^2 & W_4^3 \\ W_4^2 & W_4^2 & W_4^4 & W_4^6 \\ W_4^3 & W_4^3 & W_4^6 & W_4^9 \end{bmatrix} \begin{bmatrix} x(0) \\ x(1) \\ x(2) \\ x(3) \end{bmatrix}$$

$$\begin{bmatrix} x(0) \\ x(1) \\ x(2) \\ x(3) \end{bmatrix} = \begin{bmatrix} 1 & 1 & 1 & 1 \\ 1 & -j & -1 & j \\ 1 & -1 & 1 & -1 \\ 1 & j & -1 & -j \end{bmatrix} \begin{bmatrix} \frac{1}{3} \\ \frac{1}{3} \\ \frac{1}{3} \\ 0 \end{bmatrix} = \begin{bmatrix} \frac{1}{3} + \frac{1}{3} + \frac{1}{3} + 0 \\ \frac{1}{3} - j \frac{1}{3} - \frac{1}{3} + 0 \\ \frac{1}{3} - \frac{1}{3} + \frac{1}{3} + 0 \\ \frac{1}{3} + j \frac{1}{3} - \frac{1}{3} + 0 \end{bmatrix}$$

$$\begin{bmatrix} x(0) \\ x(1) \\ x(2) \\ x(3) \end{bmatrix} = \begin{bmatrix} 1 \angle 0^\circ \\ 0.33 \angle -90^\circ \\ 0.33 \angle 0^\circ \\ 0.33 \angle 90^\circ \end{bmatrix}$$

$$\therefore x(k) = \left\{ 1 \angle 0^\circ, 0.33 \angle -90^\circ, 0.33 \angle 0^\circ, 0.33 \angle 90^\circ \right\}$$

## Magnitude and Phase Spectrums:



1b) Two length-4 sequence are defined below:

$$x(n) = \cos\left(\frac{\pi n}{2}\right), n=0,1,2,3$$

$$h(n) = 2^n, n=0,1,2,3$$

- i) Calculate  $x(n) *_4 h(n)$  using circular convolution directly  
ii) calculate  $x(n) *_4 h(n)$  using linear convolution. (8M)

$\Rightarrow$  Given  $x(n) = \cos\left(\frac{\pi n}{2}\right)$  for  $n=0,1,2,3$

$$\text{i.e } x(0) = \cos\left(\frac{\pi \times 0}{2}\right); n=0 \Rightarrow x(0)=1$$

$$x(1) = \cos\left(\frac{\pi \times 1}{2}\right) = \cos\left(\frac{\pi}{2}\right) = 0; n=1 \Rightarrow x(1)=0$$

$$x(2) = \cos\left(\frac{\pi \times 2}{2}\right) = \cos(\pi) = -1; n=2 \Rightarrow x(2)=-1$$

$$x(3) = \cos\left(\frac{\pi \times 3}{2}\right) = \cos\left(\frac{3\pi}{2}\right) = 0; n=3 \Rightarrow x(3)=0$$

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

4)  $h(n) = 2^n$  for  $n=0,1,2,3$

$$h(0) = 2^0 = 1; n=0$$

$$h(1) = 2^1 = 2; n=1$$

$$h(2) = 2^2 = 4; n=2$$

$$h(3) = 2^3 = 8; n=3$$

$$\therefore h(n) = \{1, 2, 4, 8\}$$

i) Using circular convolution directly.

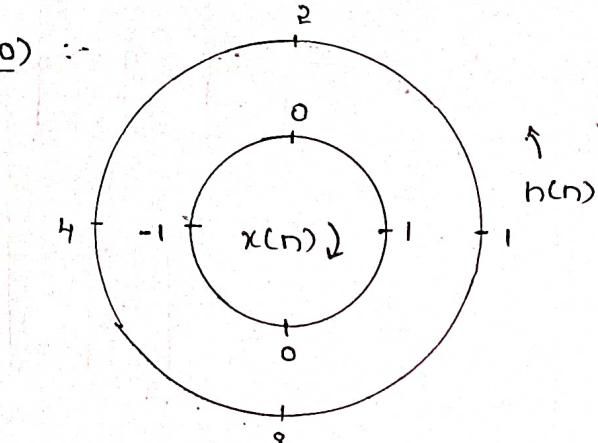
$$y(n) = x(n) \circledast h(n)$$

$$y(n) = x(n) \otimes h(n).$$

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

$$h(n) = \{ 1, 2, 4, 8 \}$$

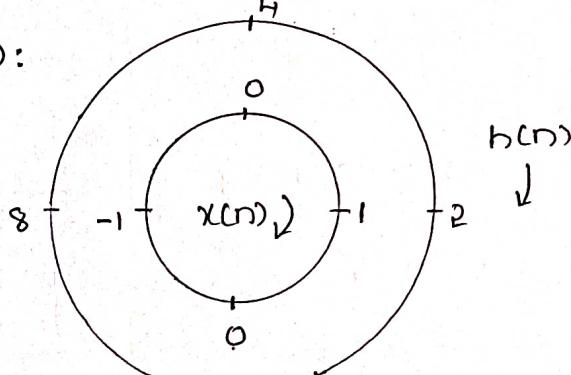
a)  $y(0)$ :



$$\begin{aligned} y(0) &= 1(1) + 0(8) + (-1)(4) + 0(2) \\ &= 1 + 0 - 4 + 0 \end{aligned}$$

$$y(0) = -3$$

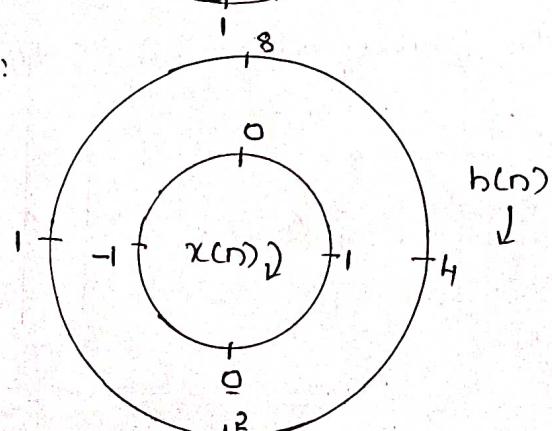
b)  $y(1)$ :



$$\begin{aligned} y(1) &= 1(2) + 0(-1) + (-1)(8) + 0(4) \\ &= 2 + 0 - 8 + 0 \end{aligned}$$

$$y(1) = -6$$

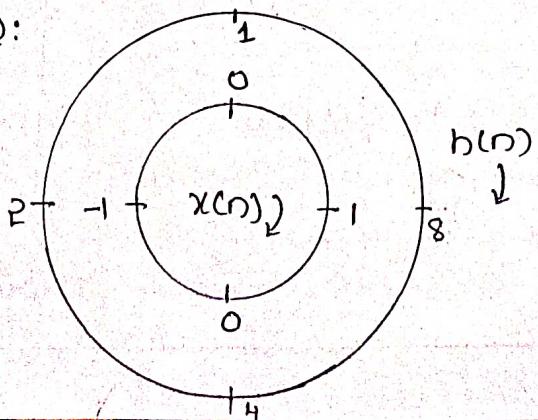
c)  $y(2)$ :



$$\begin{aligned} y(2) &= 1(4) + 0(2) + (-1)(1) + 0(8) \\ &= 4 + 0 - 1 + 0 \end{aligned}$$

$$y(2) = 3$$

d)  $y(3)$ :



$$\begin{aligned} y(3) &= 1(8) + 0(4) + (-1)(2) + 0(1) \\ &= 8 + 0 - 2 + 0 \end{aligned}$$

$$y(3) = 6$$

$$\therefore y(n) = \{ -3, -6, 3, 6 \}$$

ii> calculate  $x(n) \otimes h(n)$  using linear convolution.

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

$$h(n) = \{ 1, 2, 4, 8 \}$$

$x(n) \rightarrow$	1	0	-1	0	
$\downarrow h(n)$	1	1	0	-1	0
1	1	0	-1	0	
2	2	0	-2	0	
4	4	0	-4	0	
8	8	0	-8	0	

$$\therefore y(n) = \{ 1, 2, (4-1), (8-2), -4, -8, 0 \}$$

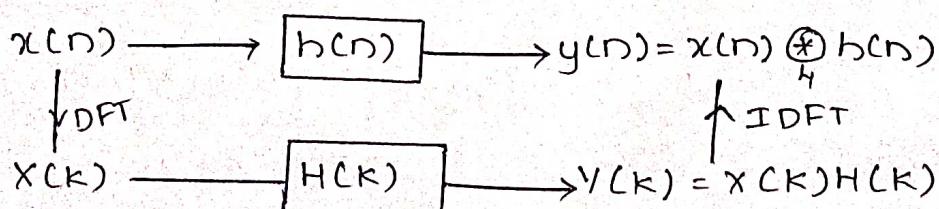
$$\therefore y(n) = \{ 1, 2, 3, 6, -4, -8, 0 \}$$

2a) Compute circular convolution using DFT+IDFT for following sequence:

$$x_1(n) = \{ 2, 3, 1, 1 \} \quad x_2(n) = \{ 1, 3, 5, 3 \} \quad (8M)$$

$$\Rightarrow \text{Given } : x_1(n) = \{ 2, 3, 1, 1 \} = x(n)$$

$$x_2(n) = \{ 1, 3, 5, 3 \} = h(n)$$



i) Finding DFT of  $x(n)$ :

$$[x(k)] = [W_N^{kn}] [x(n)] \Rightarrow \text{linear form}.$$

$$\begin{bmatrix} x(0) \\ x(1) \\ x(2) \\ x(3) \end{bmatrix} = \begin{bmatrix} W_4^0 & W_4^0 & W_4^0 & W_4^0 \\ W_4^0 & W_4^1 & W_4^2 & W_4^3 \\ W_4^0 & W_4^2 & W_4^4 & W_4^6 \\ W_4^0 & W_4^3 & W_4^6 & W_4^9 \end{bmatrix} \begin{bmatrix} x(0) \\ x(1) \\ x(2) \\ x(3) \end{bmatrix}$$

$$\begin{bmatrix} x(0) \\ x(1) \\ x(2) \\ x(3) \end{bmatrix} = \begin{bmatrix} 1 & 1 & 1 & 1 \\ 1 & -j & -1 & j \\ 1 & -1 & 1 & -1 \\ 1 & j & -1 & -j \end{bmatrix} \begin{bmatrix} 2 \\ 3 \\ 1 \\ 1 \end{bmatrix} = \begin{bmatrix} 2+3+1+1 \\ 2-j3-1+j \\ 2-3+1-1 \\ 2+j3-1-j \end{bmatrix}$$

$$\begin{bmatrix} x(0) \\ x(1) \\ x(2) \\ x(3) \end{bmatrix} = \begin{bmatrix} 7 \\ 1-j2 \\ -1 \\ 1+j2 \end{bmatrix}$$

$$\therefore x(k) = \{7, 1-j2, -1, 1+j2\}$$

ii) Finding DFT of  $h(n)$

$$\text{From Linear form: } [h(k)] = [W_N^{kn}] [h(n)]$$

$$\begin{bmatrix} h(0) \\ h(1) \\ h(2) \\ h(3) \end{bmatrix} = \begin{bmatrix} W_4^0 & W_4^0 & W_4^0 & W_4^0 \\ W_4^0 & W_4^1 & W_4^2 & W_4^3 \\ W_4^0 & W_4^2 & W_4^4 & W_4^6 \\ W_4^0 & W_4^3 & W_4^6 & W_4^9 \end{bmatrix} \begin{bmatrix} h(0) \\ h(1) \\ h(2) \\ h(3) \end{bmatrix}$$

$$\begin{bmatrix} H(0) \\ H(1) \\ H(2) \\ H(3) \end{bmatrix} = \begin{bmatrix} 1 & 1 & 1 & 1 \\ 1 & -j & -1 & j \\ 1 & -1 & 1 & -1 \\ 1 & j & -1 & -j \end{bmatrix} \begin{bmatrix} 1 \\ 3 \\ 5 \\ 3 \end{bmatrix} = \begin{bmatrix} 1+3+5+3 = 12 \\ 1-j3-5+3j = -4 \\ 1-3+5-3 = 0 \\ 1+3j-5-3j = -4 \end{bmatrix}$$

$$H(k) = \{ 12, -4, 0, -4 \}$$

iii) Finding  $H(k)$  :-

$$Y(k) = X(k) H(k)$$

$$Y(k) = \begin{bmatrix} 7 & 1-j2 & -1 & 1+j2 \end{bmatrix} \times \begin{bmatrix} 12 & -4 & 0 & -4 \end{bmatrix}$$

$$Y(k) = [84 \quad -4+j8 \quad 0 \quad -4-j8]$$

iv) Taking IDFT of  $Y(k)$

$$y(n) = \frac{1}{N} [W_N^{-kn}] [Y(k)]$$

$$\begin{bmatrix} y(0) \\ y(1) \\ y(2) \\ y(3) \end{bmatrix} = \frac{1}{N} \begin{bmatrix} W_4^0 & W_4^0 & W_4^0 & W_4^0 \\ W_4^0 & W_4^{-1} & W_4^{-2} & W_4^{-3} \\ W_4^0 & W_4^{-2} & W_4^{-4} & W_4^{-6} \\ W_4^0 & W_4^{-3} & W_4^{-6} & W_4^{-9} \end{bmatrix} \begin{bmatrix} Y(0) \\ Y(1) \\ Y(2) \\ Y(3) \end{bmatrix}$$

$$\begin{bmatrix} y(0) \\ y(1) \\ y(2) \\ y(3) \end{bmatrix} = \frac{1}{4} \begin{bmatrix} 1 & 1 & 1 & 1 \\ 1 & j & -1 & -j \\ 1 & -1 & 1 & -1 \\ 1 & -j & -1 & j \end{bmatrix} \begin{bmatrix} 84 \\ -4+j8 \\ 0 \\ -4-j8 \end{bmatrix} = \frac{1}{4} \begin{bmatrix} 84 - 4 + j8 + 0 - 4 - j8 = 76 \\ 84 - 4j - 8 + 0 + j4 - 8 = 68 \\ 84 + 4 - j8 + 0 + 4 + j8 = 92 \\ 84 + j4 + 8 + 0 - j4 + 8 = 100 \end{bmatrix}$$

$$\begin{bmatrix} y(0) \\ y(1) \\ y(2) \\ y(3) \end{bmatrix} = \frac{1}{4} \begin{bmatrix} 76 \\ 68 \\ 92 \\ 100 \end{bmatrix} = \begin{bmatrix} 19 \\ 17 \\ 23 \\ 25 \end{bmatrix}$$

$$\therefore y(n) = \{19, 17, 23, 25\}$$

2b) Find the output of the LTI system whose impulse  $h(n) = \{1, 1, 1, 1\}$  and input signal is  $x(n) = \{3, -1, 0, 1, 3, 2, 0, 1, 2, 1\}$ . Using overlap save method. Use 6-pt circular convolution (8M)

$$\Rightarrow \text{Given: } x(n) = \{3, -1, 0, 1, 3, 2, 0, 1, 2, 1\}$$

$$h(n) = \{1, 1, 1, 1\}$$

$$x(n) \xrightarrow{h(n)} y(n) = x(n) \oplus h(n)$$

$$L=10 \quad M=4 \quad L+M-1=10+4-1=13$$

$$\therefore x(n) = \{3, -1, 0, 1, 3, 2, 0, 1, 2, 1, 0, 0\}$$

For  $N=6$  Given :- 6 point circular convolution.

Step 1 :- Finding the blocks

$$x_1(n) = \underbrace{0, 0}_{(m-1)\text{ zero's}}, 3, -1, 0, 1$$

$$x_2(n) = 0, 1, 3, 2, 0, 1$$

$$x_3(n) = 0, 1, 2, 1, 0, 0$$

Step 2 : Block circular convolution

$$y_1(n) = x_1(n) \otimes_{\text{6}} h(n)$$

$$y_3(n) = x_3(n) \otimes_{\text{6}} h(n)$$

$$y_2(n) = x_2(n) \otimes_{\text{6}} h(n)$$

Finding  $y_1(n) = \sum_{k=0}^6 x_1(k)h(n-k)$

$$x_1(n) = \{ 0 \ 0 \ 3 \ -1 \ 0 \ 1 \}$$

$$h(n) = \{ 1 \ 1 \ 1 \ 0 \ 0 \ 0 \}$$

$h(n)$						$x_1(n)$	$y_1(n)$
1	0	0	0	1	1	0	1
1	1	0	0	0	1	0	1
1	1	1	0	0	0	3	3
0	1	1	1	0	0	-1	2
0	0	1	1	1	0	0	2
0	0	0	1	1	1	1	0

Finding  $y_2(n) = \sum_{k=0}^6 x_2(k)h(n-k)$

$$x_2(n) = \{ 0 \ 1 \ 3 \ 2 \ 0 \ 1 \}$$

$$h(n) = \{ 1 \ 1 \ 1 \ 0 \ 0 \ 0 \}$$

$h(n)$						$x_2(n)$	$y_2(n)$
1	0	0	0	1	1	0	1
1	1	0	0	0	1	1	2
1	1	1	0	0	0	3	4
0	1	1	1	0	0	2	6
0	0	1	1	1	0	0	5
0	0	0	1	1	1	1	3

Finding  $y_3(n) = \sum_{k=0}^6 x_3(k)h(n-k)$

$$x_3(n) = \{ 0 \ 1 \ 2 \ 1 \ 0 \ 0 \}$$

$$h(n) = \{ 1 \ 1 \ 1 \ 0 \ 0 \ 0 \}$$

$$\left[ \begin{array}{cccccc} h(m) & & & & & \\ \hline 1 & 0 & 0 & 0 & 1 & 1 & 1 \\ 1 & 1 & 0 & 0 & 0 & 1 & 1 \\ 1 & 1 & 1 & 0 & 0 & 0 & 2 \\ 0 & 1 & 1 & 1 & 0 & 0 & 1 \\ 0 & 0 & 1 & 1 & 1 & 0 & 0 \\ 0 & 0 & 0 & 1 & 1 & 1 & 0 \end{array} \right] = \left[ \begin{array}{c} x_3(n) \\ y_3(n) \end{array} \right]$$

Step 3: Combining  $y_1(n)$ ,  $y_2(n)$  &  $y_3(n)$ .

$$y_1(n) = \underbrace{1 \quad 1}_{(m-1) \text{ discard}} \quad 3 \quad 2 \quad 2 \quad 0$$

$$y_2(n) = \underbrace{1 \quad 2}_{(m-1) \text{ discard}} \quad 4 \quad 6 \quad 5 \quad 3$$

$$y_3(n) = \underbrace{0 \quad 1}_{(m-1) \text{ discard}} \quad 3 \quad 4 \quad 3 \quad 1$$

$$y(n) = \{3, 2, 2, 0, 4, 6, 5, 3, 3, 4, 3, 1\}$$

$$\therefore y(n) = \{3, 2, 2, 0, 4, 6, 5, 3, 3, 4, 3, 1\} = 12.$$

$$\therefore L+M-1 = 10+3-1 = 12.$$

## MODULE - II

3a) What are FFT algorithms? Explain the advantages of FFT algorithms over direct computations of DFT for a sequence  $x(n)$  (4M)

=> FFT [Fast Fourier Transform] is an algorithm that computes the discrete Fourier transform [DFT] of a sequence or its inverse (IDFT). Fourier analysis converts a signal from its original domain to frequency domain and vice versa. The DFT is obtained by decomposing a sequence of values into components of different frequencies, so FFT rapidly

Computes such transformation by factorizing the DFT matrix into a product of sparse factors. Hence it reduces the complexity of computing the DFT. It makes use of the symmetry and periodicity properties of twiddle factor to reduce DFT computation.

### Advantages of FFT algorithm over direct computation of DFT

- The computation complexity of FFT algorithm greatly reduced compare to direct computation
- FFT algorithm can be used to find DFT as well as IDFT very effectively.
- FFT helps in converting the time domain to frequency domain.

3b) What are the differences and similarities between DIT and DIF-FFT algorithms? (4M).

⇒	DIT	DIF
	<ul style="list-style-type: none"> <li>→ In this algorithm time index 'n' is decimated into even &amp; odd at various level.</li> <li>→ Butterfly structure:</li> </ul> <p><u>Basic :-</u></p> <p><u>Modified :-</u></p>	<ul style="list-style-type: none"> <li>→ In this algorithm frequency index 'k' is decimated into even &amp; odd at various level.</li> <li>→ Butterfly Structure:</li> </ul> <p><u>Modified :-</u></p>

DIT	DIF
<ul style="list-style-type: none"> <li>→ In this algorithm the input sequence has to be bit reversal. Order and output sequence will be normal.</li> </ul>	<ul style="list-style-type: none"> <li>→ In this algorithm the input sequence is normal and output sequence is bit reverse.</li> </ul>
<u>Similarities:-</u> <ul style="list-style-type: none"> <li>→ Memory requirement is same.</li> <li>→ In DIT computation for modified structure.</li> </ul> <p>Multiplication = <math>\frac{N}{2} \log_2 N</math></p> <p>Addition = <math>N \log_2 N</math></p>	<ul style="list-style-type: none"> <li>→ Memory requirement is same.</li> <li>→ In DIF computation for modified structure.</li> </ul> <p>Multiplication = <math>\frac{N}{2} \log_2 N</math></p> <p>Addition = <math>N \log_2 N</math></p>

3c) Find the 8 point DFT of the sequence  $x(n) = \{1, 2, 3, 4, 3, 2, 1\}$   
 Using DIT - FFT radix - 2 algorithm (8M).

⇒ Given:-

$$x(n) = \{1, 2, 3, 4, 3, 2, 1\}$$

To solve using DIF - FFT radix - 2 algorithm.

Normal  
 $\chi(n)$

$\chi(k)$

Bit  
reversal

$$\chi(0) = 1$$

$$\chi(0) = 1$$

$$\chi(1) = 2$$

$$\chi(4) = 4$$

$$\chi(2) = 3$$

$$\chi(2) = 3$$

$$\chi(3) = 4$$

$$\chi(6) = 2$$

$$\chi(5) = 1$$

$$\chi(1) = 2$$

$$\chi(4) = 3$$

$$\chi(5) = 3$$

$$\chi(3) = 2$$

$$\chi(3) = 1$$

$$\chi(2) = 1$$

$$\chi(7) = 1$$

III decimation.

II Decimation

I Decimation.

$N=2$ ,  $r=0$ .

$N=4$ ,  $r=0,1$

$$W_N^r = W_2^0$$

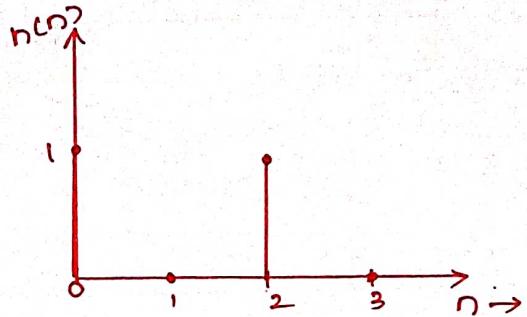
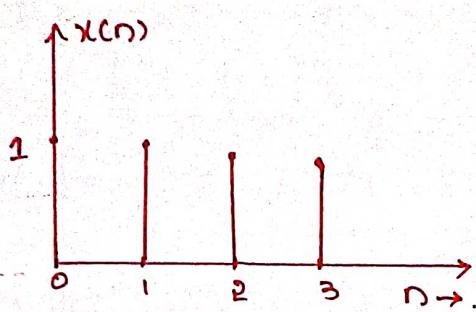
$$W_N^r = W_4^0, W_4^1$$

$$N=8, r=0, 1, 2, 3$$

$$W_N^r = W_8^0, W_8^1, W_8^2, W_8^3$$

$$X(k) = \{20, -5.828 - j2.414, 0, -0.172 + j0.414, 0, -5.828 + j2.414\}$$

4a) Find the 4-pt circular convolution of  $x(n)$  and  $h(n)$  given using radix-2 DIF - FFT algorithm.

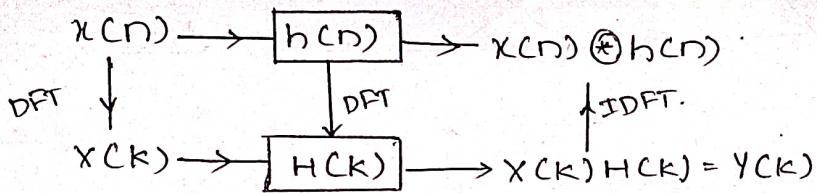


(8M)

Soln :-

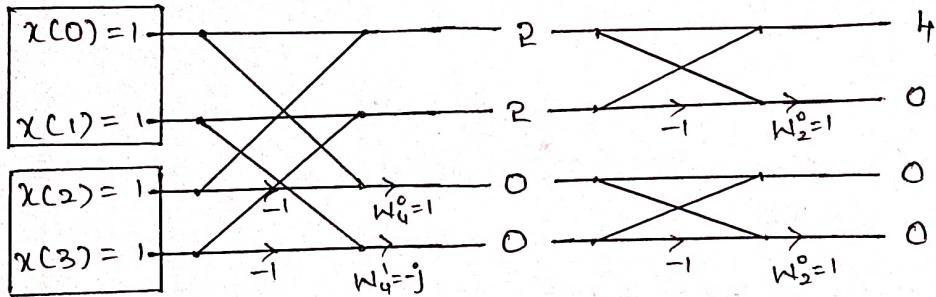
$$\text{Given: } x(n) = \{1, 1, 1, 1\}, h(n) = \{1, 0, 1, 0\}$$

Using DIF-FFT algorithm, we get the following flow diagram:



i) Finding DFT  $X(K)$  by DIF-FFT Radix-2 Algorithm.

$x(n)$



$x(k)$

Bit reversal

$$x(0) = 4$$

$$x(2) = 0$$

$$x(1) = 0$$

$$x(3) = 0$$

I decimation

$$N=4, r=0,1$$

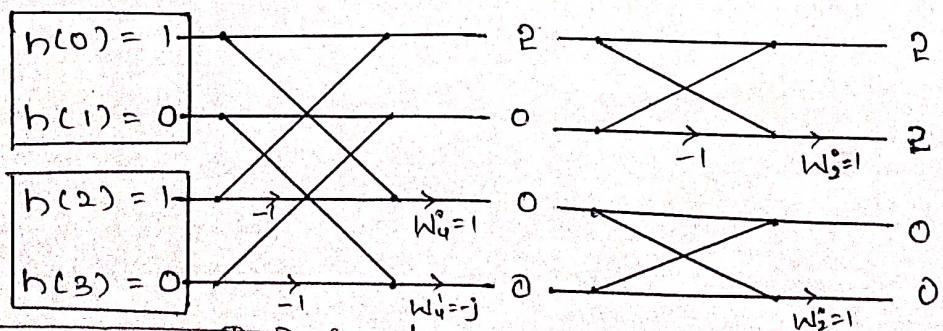
II decimation

$$N=2, r=0$$

$$\therefore X(K) = \{4, 0, 0, 0\}$$

ii) Finding DFT  $H(K)$  by DIF-FFT Radix-2 Algorithm:-

$h(n)$



$H(k)$

Bit Reversal

$$H(0) = 2$$

$$H(2) = 2$$

$$H(1) = 0$$

$$H(3) = 0$$

I Decimation

$$N=4, r=0,1$$

II Decimation

$$N=2, r=0$$

$$H(k) = \{2, 0, 2, 0\}$$

iiv) Finding  $Y(k) = X(k)H(k)$

$$Y(k) = \{4, 0, 0, 0\} \{2, 0, 2, 0\}$$

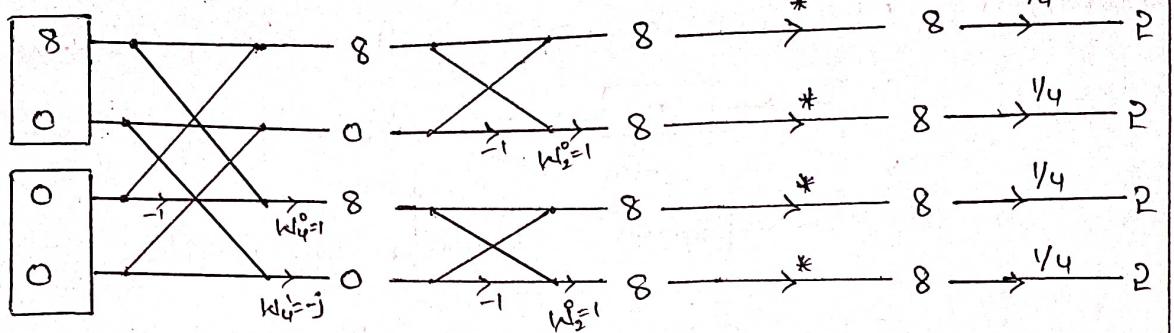
$$Y(k) = \{8, 0, 0, 0\}$$

v) Finding IDFT of  $Y(k)$  to find  $y(n)$

$$Y(k)$$

$$Y^*(k)$$

$$Y(0) = 8$$



$$y(n)$$

Bit reversal  $y(n)$

$$2$$

$$y(0) = 2$$

$$2$$

$$y(2) = 2$$

$$2$$

$$y(1) = 2$$

$$2$$

$$y(3) = 2$$

$$\therefore y(n) = x(n) \otimes h(n) = \{2, 2, 2, 2\} //$$

4b) Given  $x(n) = (n+1)$  and  $N=8$ . Determine  $X(k)$ . Using DIF-FFT Algorithm (8M)

$\Rightarrow$  Given  $x(n) = (n+1)$ ,  $N=8$ .

$$\therefore x(0) = (0+1) = 1$$

$$x(1) = 1+1 = 2$$

$$x(2) = 2+1 = 3$$

$$x(3) = 3+1 = 4$$

$$x(4) = 4+1 = 5$$

$$x(5) = 5+1 = 6$$

$$x(6) = 6+1 = 7$$

$$x(7) = 7+1 = 8$$

$$\therefore x(n) = \{1, 2, 3, 4, 5, 6, 7, 8\}$$

Normal  
 $\chi(k)$

$X(k)$   
Bit reversal

$\chi(0) = 1$

$X(0) = 36$

$\chi(1) = 2$

$\chi(1) = -4$

$\chi(2) = 3$

$X(2) = -4$

$\chi(2) = -4 + 4j$

$\chi(3) = 4$

$X(3) = -4 - 4j$

$\chi(3) = -4 + 1.0656j$

$\chi(4) = 5$

$X(4) = -4 + 9.0656j$

$\chi(4) = -4 + 9.0656j$

$\chi(5) = 6$

$X(5) = -4 - 1.0656j$

$\chi(5) = -4 - 1.0656j$

$\chi(6) = 7$

$X(6) = -4 + 4j$

$\chi(6) = -4 + 4j$

$\chi(7) = 8$

$X(7) = -4 - 9.0656j$

$\chi(7) = -4 - 9.0656j$

### I Decimation

### II Decimation

### III Decimation

$$\therefore X(k) = \{ 36, -4 + 9.0656j, -4 + 4j, -4 + 1.0656j, -4, -4 - 1.0656j, -4 - 4j, -4 - 9.0656j \}$$

$N=8, r=0, 1, 2, 3$

$N=4, r=0, 1$

$N=2, r=0$

### MODULE - 03

5a) Convert the analog filter with system transfer function

$$H(s) = \frac{(s+0.1)}{(s+0.1)^2 + 3^2} \quad \text{into a digital IIR filter by mean of the impulse invariant method.} \quad (6M)$$

$\Rightarrow$  Given  $H(s) = \frac{(s+0.1)}{(s+0.1)^2 + 3^2}$

w.k.t  $\frac{(s+a)}{(s+a)^2 + b^2} \rightarrow \frac{1 - e^{aT}(\cos\omega_b T)z^{-1}}{1 - 2e^{aT}(\cos\omega_b T)z^{-1} + e^{2aT}z^{-2}}$

Comparing the above formula with  $H(s)$  we get.

$$a = 0.1 \quad ; \quad b = 3$$

$$H(z) = \frac{1 - e^{-0.1T}(\cos\omega_b T)z^{-1}}{1 - 2e^{-0.1T}(\cos\omega_b T)z^{-1} + e^{-2(0.1)T}z^{-2}}$$

$$H(z) = \frac{1 - e^{-0.1T}(\cos\omega_b T)z^{-1}}{1 - 2e^{-0.1T}(\cos\omega_b T)z^{-1} + e^{-0.2T}z^{-2}} \quad //$$

5b) Design a butter worth digital IIR low pass filter using bilinear transformation by taking  $T=0.1$  sec. to satisfy the following specification:

$$0.6 \leq |H(e^{j\omega})| \leq 1.0; \quad \text{for } 0 \leq \omega \leq 0.35\pi$$

$$|H(e^{j\omega})| \leq 0.1; \quad \text{for } 0.7\pi \leq \omega \leq \pi. \quad (10M)$$

$\Rightarrow$  Given :-  $T = 0.1$  sec,  $0.6 \leq |H(e^{j\omega})| \leq 1$ ; for  $0 \leq \omega \leq 0.35\pi$   
 $|H(e^{j\omega})| \leq 0.1$ ; for  $0.7\pi \leq \omega \leq \pi$ .

Given :-  $A_p = 0.6$  for  $\omega_p \leq 0.35\pi$

$$A_s = 0.1 \quad \text{for } \omega_s \geq 0.7\pi$$

i) To obtain specification of equivalent analog filter for bilinear transformation:-

$$\omega_p = \tan \frac{\omega_p}{2} = \tan \frac{0.35\pi}{2} = 0.6128$$

$$\omega_s = \tan \frac{\omega_s}{2} = \tan \frac{0.7\pi}{2} = 1.9626$$

Hence specification of equivalent analog filter are.

$$A_p = 0.6, \omega_p = 0.6128$$

$$A_s = 0.1, \omega_s = 1.9626$$

ii) To obtain order of filter:-

$$N = \frac{1}{2} \log \left[ \frac{\left( \frac{1}{A_s^2} - 1 \right) \left( \frac{1}{A_p^2} - 1 \right)}{\log \left( \frac{\omega_s}{\omega_p} \right)} \right] = \frac{1}{2} \left[ \frac{\log \left[ \left( \frac{1}{0.1^2} - 1 \right) \left( \frac{1}{0.6^2} - 1 \right) \right]}{\log \left( \frac{1.9626}{0.6128} \right)} \right]$$

$$= \frac{1}{2} \left[ \frac{\log (176)}{\log (3.2022)} \right] = \frac{1}{2} [4.44]$$

$$N = 2.22 \approx 2$$

To obtain  $\omega_c$ .

$$\omega_c = \frac{1}{2} \left[ \frac{\omega_p}{\left( \frac{1}{A_p^2} - 1 \right)^{\frac{1}{2N}}} + \frac{\omega_s}{\left( \frac{1}{A_s^2} - 1 \right)^{\frac{1}{2N}}} \right] = \frac{1}{2} \left[ \frac{0.6128}{\left( \frac{1}{0.6^2} - 1 \right)^{\frac{1}{2 \times 2}}} + \frac{1.9626}{\left( \frac{1}{0.1^2} - 1 \right)^{\frac{1}{2 \times 2}}} \right]$$

$$\omega_c = 0.576$$

To obtain poles of  $H_a(s)$

$$P_k = \pm \omega_c e^{j(CN+2k+1)\pi/2N}; k = 0, 1, \dots, N-1$$

$$\text{For } N=2, P_k = \pm \omega_c e^{j(2+2k+1)\pi/2 \times 2}.$$

$$P_k = \pm \omega_c e^{j(3+2k)\pi/4}$$

$$P_k = \pm 0.576 e^{j(3+2k)\pi/4}; k = 0, 1.$$

$$\text{For } k=0; P_0 = \pm 0.576 e^{j(3+2 \cdot 0)\pi/4} = \pm 0.576 e^{j\frac{3\pi}{4}}$$

$$P_0 = + (0.576 \cos \frac{3\pi}{4} - j \sin \frac{3\pi}{4}), - (0.576 \cos \frac{3\pi}{4} - j \sin \frac{3\pi}{4})$$

$$P_0 = (-0.407 - j0.407), (0.407 + j0.407)$$

$$\text{For } k=1; P_1 = \pm 0.576 e^{j(3+2(1)\pi/4)} = \pm 0.576 e^{j\frac{5\pi}{4}}$$

$$P_1 = + (0.576 \cos \frac{5\pi}{4} - j 0.576 \sin \frac{5\pi}{4}), - (0.576 \cos \frac{5\pi}{4} - j 0.576 \sin \frac{5\pi}{4})$$

$$P_1 = (-0.407 + j 0.407), (0.407 - j 0.407).$$

$$\therefore s_1 = -0.407 - j 0.407 \text{ and } s_1^* = -0.407 + j 0.407.$$

For obtain system function  $H_a(s)$ .

For  $N=2$ ,

$$H_a(s) = \frac{s^2 c^2}{(s-s_1)(s-s_1^*)}$$

$$H_a(s) = \frac{0.576^2}{(s+0.407+j0.407)(s+0.407-j0.407)}$$

$$= \frac{0.576^2}{(s+0.407)^2 - (j0.407)^2} = \frac{0.576^2}{(s+0.407)^2 + (0.407)^2}$$

$$\approx \frac{0.3317}{s^2 + 0.1656 + 0.814s + 0.1656}.$$

$$H_a(s) = \frac{0.3317}{s^2 + 0.814s + 0.3312}$$

To obtain  $H(z)$  by bilinear transformation.

$$H(z) = H_a(s) \Big|_{s=\frac{1-z^{-1}}{1+z^{-1}}}$$

$$= \frac{0.3317}{\left(\frac{1-z^{-1}}{1+z^{-1}}\right)^2 + 0.814\left(\frac{1-z^{-1}}{1+z^{-1}}\right) + 0.3312} = \frac{0.3317(1+\bar{z}^1)^2}{(1-z^{-1})^2 + 0.814(1-\bar{z}^1)(1+\bar{z}^1) + 0.3312(1+\bar{z}^1)^2}$$

$$= \frac{0.3317(1+\bar{z}^2 + 2\bar{z}^1)}{1+z^2 - 2z^{-1} + 0.814(1-\bar{z}^2) + 0.3312(1+\bar{z}^2 + 2\bar{z}^{-1})}$$

$$H(z) = \frac{0.3317(1+\bar{z}^2 + 2\bar{z}^{-1})}{2.1452 - 1.3376z^{-1} + 0.5172z^{-2}}$$

$$\therefore H(z) = \frac{0.3317z^2 + 0.6634z^{-1} + 0.3317}{0.5172z^2 - 1.3376z^{-1} + 2.1452}$$

## 6a) Compare analog and digital filters:-

### Analog Filter

+> It processes continuous time signal using passive elements ( $R, L, C$ ) called as passive filter or using active elements (opamp) called as active filter.

#### +> Types of Analog Filters:-

- 1) LPF - Low pass filters
- 2) HPF - High pass filters
- 3) BPF - Band pass filters ...

+> Infinite frequency range because FT is non periodic.

+> Mathematically, it is defined by transfer function in 's' domain ' $H(s)$ '.

+> Frequency response characteristic is defined by  $H(j\omega)$  where  $\omega$  is Analog frequency (rad/sec)

+> For stable analog filter, all poles lie on left half of 's' plane

+> Design of Analog filter, means for given frequency response char.  $H_d(j\omega)$ , finding  $H(s)$ .

### Digital Filter

+> It processes digital signal through numerical computation using digital circuits [FPGA or DSA].

#### +> Types of Digital Filters:-

- 1) IIR - Infinite impulse response Filters.
- 2) FIR - Finite impulse response filters.

+> Finite frequency range i.e  $(-\pi \text{ to } \pi)$  because DTFT is periodic in  $2\pi$ .

+> Mathematically, it is defined by transfer function in 'z' domain ' $H(z)$ '.

+> Frequency response characteristic is defined by  $H(e^{j\omega})$  where  $\omega$  is digital frequency (rad)

+> For stable digital filter all poles lie within the unit circle in 'z' plane.

+> For given frequency response char.  $H_d(e^{j\omega})$  design of digital filter means finding ' $H(z)$ '

Qb) Determine the poles of lowpass butterworth filter for  $N=2$ . Sketch the location of poles on s-plane and hence determine the normalized transfer function of lowpass filter. (8M)

Design of low pass butterworth filter for  $N=2$ .  
Low pass filter can be designed using the butterworth polynomial.

$$|H(j\omega)| = \frac{1}{\sqrt{1 + \left(\frac{\omega}{\omega_c}\right)^{2N}}} , N = \text{Order.}$$

$\omega_c = -3\text{dB cutoff frequency of the filter.}$

To find equation for 'N' ..

Take log on both sides.

$$20 \log |H(j\omega)| = -\log \left[ 1 + \left( \frac{\omega}{\omega_c} \right)^{2N} \right] = k.$$

$$\text{at } \omega = \omega_p, k = k_p$$

$$\text{at } \omega = \omega_s, k = k_s.$$

we get .

$$k_p = -10 \log \left[ 1 + \left( \frac{\omega_p}{\omega_c} \right)^{2N} \right]$$

$$\left( \frac{\omega_p}{\omega_c} \right)^{2N} = 10^{-k_p/10} - 1 \quad \rightarrow \textcircled{1}$$

Similarly we get .

$$\left( \frac{\omega_s}{\omega_p} \right)^{2N} = 10^{-k_s/10} - 1 \quad \rightarrow \textcircled{2}$$

Divide equation  $\textcircled{1}$  by  $\textcircled{2}$ .

$$\left( \frac{\omega_p}{\omega_c} \right)^{2N} = \frac{(10^{-k_p/10} - 1)}{(10^{-k_s/10} - 1)}$$

Taking log on both side .

$$2N \log \left( \frac{\omega_p}{\omega_c} \right) = \log \left[ \frac{(10^{-k_p/10} - 1)}{(10^{-k_s/10} - 1)} \right]$$

$$N = \frac{1}{2} \left[ \log \left[ \frac{\frac{10^{-kp/10} - 1}{10^{-ks/10} - 1}}{\log \left( \frac{\omega_p}{\omega_c} \right)} \right] \right]$$

To find equation for  $\omega_c$

From equation ① we have

$$\left( \frac{\omega_p}{\omega_c} \right)^{2N} = 10^{-kp/10} - 1$$

$$\frac{\omega_p}{\omega_c} = \left( 10^{-kp/10} - 1 \right)^{1/2N}$$

$$\omega_c = \frac{\omega_p}{\left( 10^{-kp/10} - 1 \right)^{1/2N}}$$

To find the transfer function  $H(s)$

Normalised transfer function is determined by setting  $\omega_c = 1$ .  
Wkt, the poles of transfer function lie on a circle of radius  $\omega_c = 1$ . The total number of poles will be equal to  $N$ .

Then angle between the poles is  $\theta = \frac{360}{2N}$ .

The location of  $2N$  pole around a circle is made as per the following rule.

For  $N = \text{odd}$ .

The first pole will lie along +ve real axis of  $s$  plane.

For  $N = \text{even}$ .

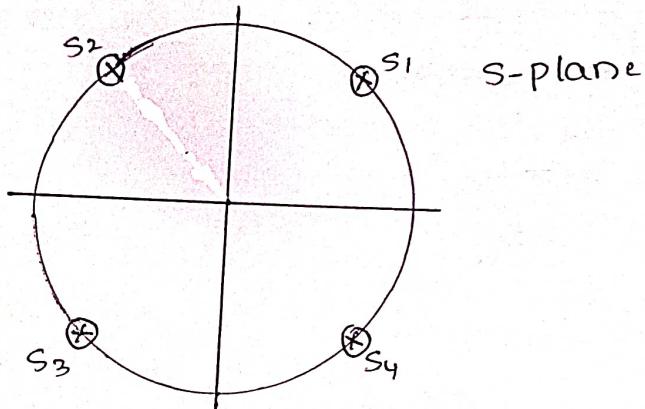
The first pole lies at angle  $\theta/2$  from the +ve real axis.

Now to find for  $N=2$ :-

Total poles =  $2N = 4$ .

$$\text{Angle between poles } \Theta = \frac{360^\circ}{2N} = \frac{360^\circ}{4} = 90^\circ.$$

First pole will lie at an angle  $\frac{\Theta}{2} = 45^\circ$  from the real axis.



For stable filter, we have to consider poles lying on LHS.

$$S_2 = -0.707 + j0.707$$

$$S_3 = -0.707 - j0.707$$

$$\therefore \text{Transfer function } H(s) = \frac{1}{(s-s_2)(s-s_3)}.$$

$$H(s) = \frac{1}{(s+0.707 - j0.707)(s+0.707 + j0.707)}$$

$$H(s) = \frac{1}{s^2 + \sqrt{2}s + 1}$$

Here  $H(s) = s^2 + \sqrt{2}s + 1$  is called normalized butterworth polynomial for  $N=2_{II}$ .

### 6c) Write difference between IIR and FIR Filter (4M)

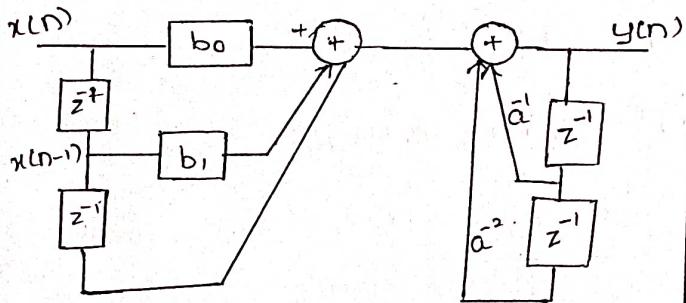
#### IIR

- It is an acronym for Infinite Impulse response.
- It is defined by differential equation

$$y(n) = \sum_{k=0}^M b_k x(n-k) - \sum_{k=1}^N a_k y(n-k)$$

Since present output depends upon past values of output. It has a feedback. Hence this filter is called recursive type of filter.

→ Block Diagram.



→ Transfer function is given by

$$H(z) = \frac{\sum_{k=0}^M b_k z^{-k}}{\sum_{k=0}^N a_k z^{-k}}$$

so this type of filter has zeros and poles. For stable system the poles should be made to lie within the unit circle.

→ computational efficiency is comparatively more

→ Memory requirement is less

→ Controllability is difficult

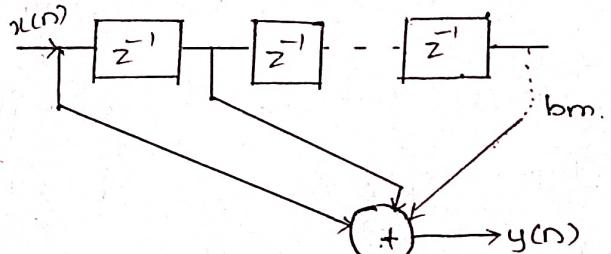
#### FIR

- It is an acronym for Finite Impulse Response.
- It is defined by differential equation

$$y(n) = \sum_{k=0}^M b_k x(n-k)$$

→ As there is no feedback so it is called as non-recursive type of filter.

→ Block diagram.



→ Transfer function is given by

$$H(z) = \sum_{k=0}^M b_k z^{-k}$$

This type of filter has only zeros. Poles lies at z=0 in z plane. so it is always stable

→ computational efficiency is less.

→ Memory requirement is more

→ Controllability is easy.

## MODULE - 4

7a) Design a chebyshev digital filter IIR lowpass filter using bilinear transformation to satisfy the following specifications;

$$0.9 \leq |H(e^{j\omega})| \leq 1.0 ; \text{ for } 0 \leq \omega \leq 0.25\pi$$

$$|H(e^{j\omega})| \leq 0.24 ; \text{ for } 0.5\pi \leq \omega \leq \pi.$$

(12M)

$\Rightarrow$  i) Given :  $A_p = 0.9$ ,  $\omega_p = 0.25\pi$ .

$$A_s = 0.24, \omega_s = 0.5\pi$$

ii) Prewrapping:-

$$\Omega = \frac{2}{T} \tan \frac{\omega}{2} = \tan \frac{\omega}{2} \text{ assuming } \frac{2}{T} = 1$$

$$\Omega_p = \tan \frac{\omega_p}{2} = \tan \frac{0.25\pi}{2} = 0.414 \text{ rad/sec.}$$

$$\Omega_s = \tan \frac{\omega_s}{2} = \tan \frac{0.5\pi}{2} = 1 \text{ rad/sec.}$$

Thus specification of equivalent analog filter

$$A_p = 0.9, \Omega_p = 0.414 \text{ rad/sec.}$$

$$A_s = 0.24, \Omega_s = 1 \text{ rad/sec.}$$

iii) Order of chebyshev filter:-

$$\text{Let us first calculate } \epsilon = \sqrt{\frac{1}{A_p^2} - 1} = \sqrt{\frac{1}{0.9^2} - 1} = 0.484$$

$$4 \cdot \delta = \sqrt{\frac{1}{A_s^2} - 1} = \sqrt{\frac{1}{0.24^2} - 1} = 4.045$$

The order of chebyshev is given by

$$N = \frac{\cosh^{-1}(\delta/\epsilon)}{\cosh^{-1}(\Omega_s/\Omega_p)} = \frac{\cosh^{-1}\left(\frac{4.045}{0.484}\right)}{\cosh^{-1}\left(\frac{1}{0.414}\right)} = 1.84 \approx 2.$$

$$H_a(s) = \frac{k}{(s - 0.242)^2 + (0.361)^2} = \frac{k}{[s^2 - 0.484s + 0.1868] + 0.130}$$

$$H_a(s) = \frac{k}{[s^2 - 0.484s + 0.1868]}$$

$$\therefore b_0 = 0.1868$$

$$k = \frac{b_0}{\sqrt{1+e^2}} = \frac{b_0 1868}{\sqrt{1+0.484}} = 0.1533$$

$$\therefore H_a(s) = \frac{0.1533}{(s^2 - 0.484s + 0.1868)}$$

Vi> H(z) using bilinear transformation:

$$H(z) = H_a(s) \Big|_{s = \frac{2}{T} \left( \frac{1-z^{-1}}{1+z^{-1}} \right)}$$

$$= \frac{0.1533}{(s^2 - 0.484s + 0.1868)} \quad \Bigg|_{s = \frac{1-z^{-1}}{1+z^{-1}}} \quad \text{since } \frac{2}{T} = 1$$

$$= \frac{0.1533}{\left[ \frac{1-z^{-1}}{1+z^{-1}} \right]^2 - 0.484 \left[ \frac{1-z^{-1}}{1+z^{-1}} \right] + 0.1868}$$

$$= \frac{0.1533 (1+z^{-1})^2}{[1-z^{-1}]^2 - 0.484 (1-z^{-1})(1+z^{-1}) + 0.1868 (1+z^{-1})^2}$$

$$= \frac{0.1533 (1+z^{-1})^2}{1 - 2z^{-1} + z^{-2} - 0.484 (1-z^{-2}) + 0.1868 (1+z^{-2}+2z^{-1})}$$

$$H(z) = \frac{0.1533 z^{-2} + 0.3066 z^{-1} + 0.0285}{1.6738 z^{-2} - 1.626 z^{-1} + 0.6998}$$

### IV) Poles of $H_{ac(s)}$ .

Let us calculate the values of  $\mu = \frac{1 + \sqrt{1 + \varepsilon^2}}{\varepsilon} = \frac{1 + \sqrt{1 + 0.484^2}}{0.484}$

$$\mu = 4.36$$

$$a = \omega_p \left[ \frac{\mu^{\frac{1}{N}} - \mu^{-\frac{1}{N}}}{2} \right] = 0.414 \left[ \frac{4.36^{\frac{1}{2}} - 4.36^{-\frac{1}{2}}}{2} \right] = 0.33$$

$$b = \omega_p \left[ \frac{\mu^{1/N} + \mu^{-1/N}}{2} \right] = 0.414 \left[ \frac{4.36^{\frac{1}{2}} + 4.36^{-\frac{1}{2}}}{2} \right] = 0.53$$

$$\phi_k = \frac{(2k+N+1)\pi}{2N} ; k=0, 1, 2, \dots, N-1$$

$$\text{For } N=2 \quad \phi_k = \frac{(2k+2+1)}{2 \times 2} ; k=0, 1$$

$$\therefore \phi_k = \frac{(2k+3)}{4} ; k=0, 1$$

$k$	$\phi_0 = \frac{2k+3}{4}$	$\sigma_k = a \cos \phi_k$	$\omega_k = b \sin \phi_k$	$P_k = \sigma_k + j\omega_k$
0	$\phi_0 = 2(0)+\frac{3}{4}$ $\phi_0 = 0.75$	$\sigma_0 = 0.33 \cos\left(\frac{3}{4}\right)$ $\sigma_0 = 0.242$	$\omega_0 = 0.53 \sin\left(\frac{3}{4}\right)$ $\omega_0 = 0.361$	$P_0 = 0.242 + j0.361$
1	$\phi_1 = 2(1)+\frac{3}{4}$ $\phi_1 = 2.75$	$\sigma_1 = 0.33 \cos(2.75)$ $\sigma_1 = -0.305$	$\omega_1 = 0.53 \sin(2.75)$ $\omega_1 = 0.202$	$P_1 = -0.305 + j0.202$

$$s_1 = p_0 = 0.242 + j0.361 , s_1^* = 0.242 - j0.361$$

### V) System equation (function $H_{ac(s)}$ )

$$H_{ac(s)} = \frac{k}{(s-s_1)(s-s_1^*)} = \frac{k}{(s-0.242-j0.361)(s-0.242+j0.361)}$$

7b) Write the relation between analog and digital

frequency in Bilinear transformation.

(CHM)

⇒ Analog frequency is denoted by ( $\omega$ ) & digital frequency is denoted by ( $\omega_n$ ). Analog frequency is converted to digital signal by Sampling.

$$\cos(\omega t + \phi) \xrightarrow{\text{Sampling}} \begin{aligned} &\cos(\omega nT_s + \phi) \\ &t = nT_s \end{aligned}$$

$$\omega = \omega T_s$$

$$\omega = \frac{\omega}{f_s} \Rightarrow T_s = \frac{1}{f_s}$$

In analog frequency  $\omega$  <sup>unit</sup> is rad/sec. but in digital frequency  $\omega_n$  is rad. Analog frequency are transformed using Laplace transform while digital frequency are transformed using z transform. The bilinear transform is mathematical relationship which converts the transfer function of a particular filter from laplace domain to z-domain and viceversa.

8a) Obtain direct form-I and direct form-II realization

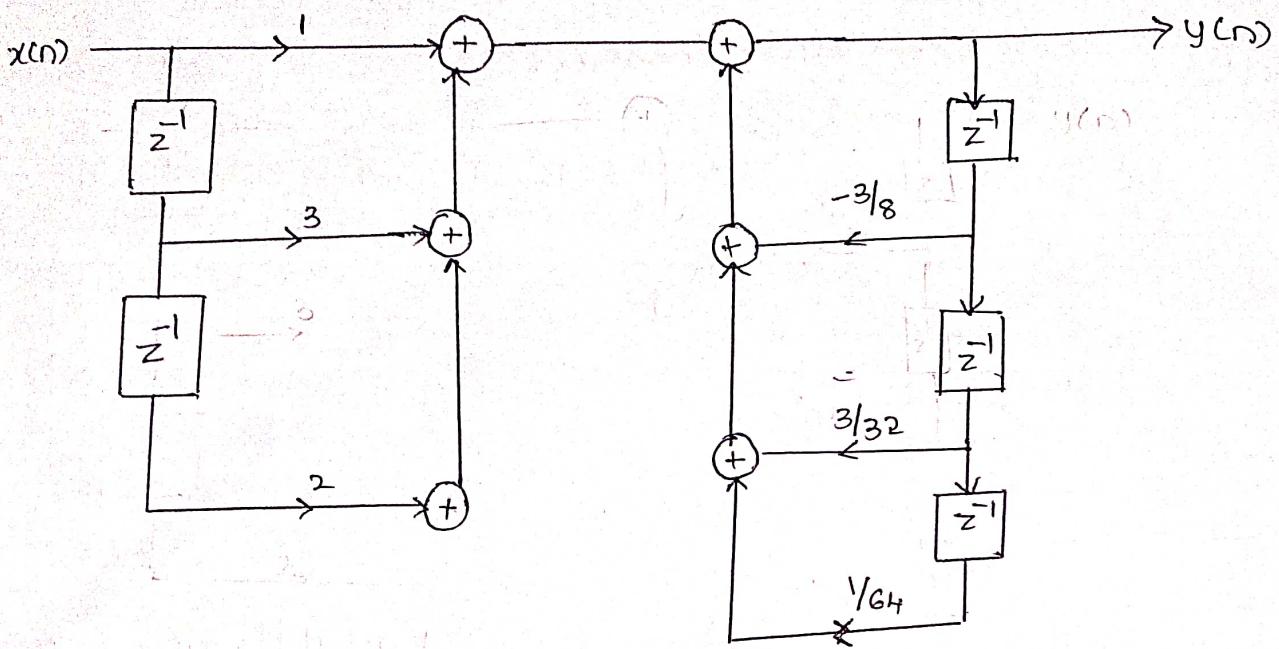
of the LTI system governed by the relation.

$$y(n) = -\frac{3}{8} y(n-1) + \frac{3}{32} y(n-2) + \frac{1}{64} y(n-3) + x(n) + 3x(n-1) + 2x(n-2) \quad (8M)$$

⇒ Direct form I :-

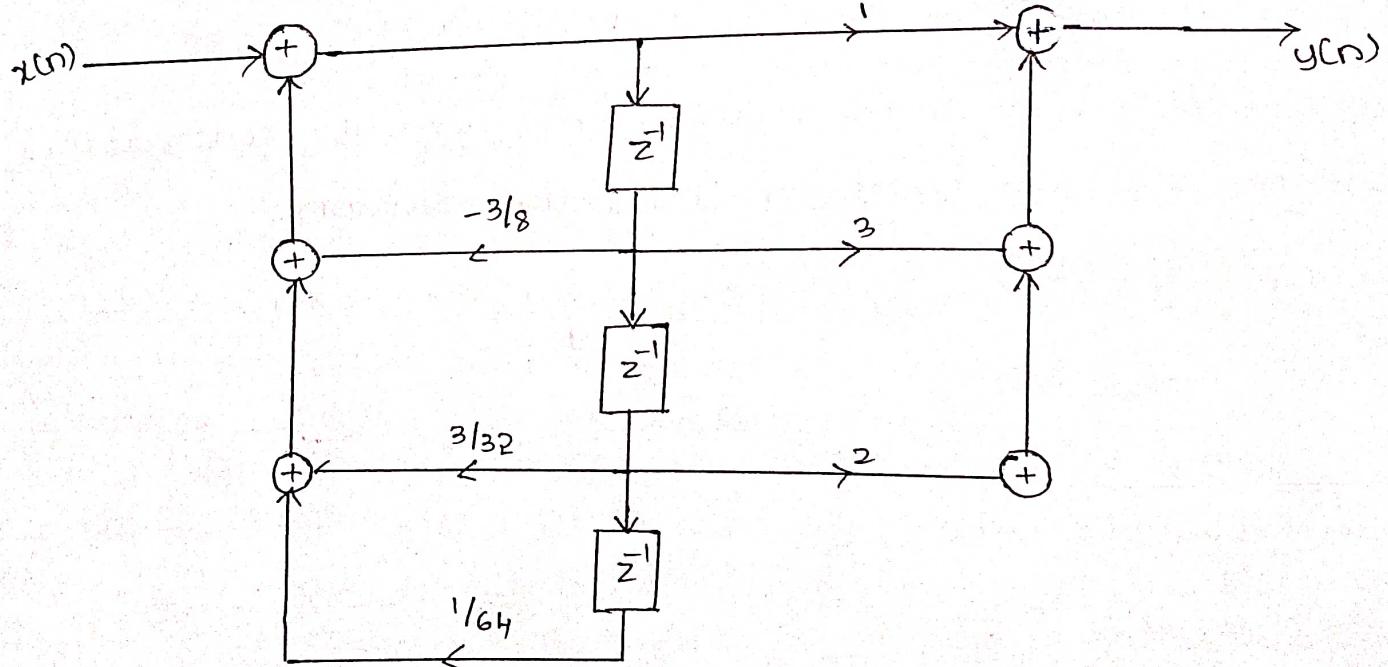
From given equation.

$$y(n) = \frac{1}{4} y(n-1) + \frac{3}{32} y(n-2) + \frac{1}{64} y(n-3) + x(n) + 3x(n-1) + 2x(n-2)$$



Direct Form II :-

$$y(n) = -\frac{3}{8}y(n-1) + \frac{3}{32}y(n-2) + \frac{1}{64}y(n-3) + x(n) + 3x(n-1) + 2x(n-2)$$



8b) Realise the given system in cascade and parallel form:-

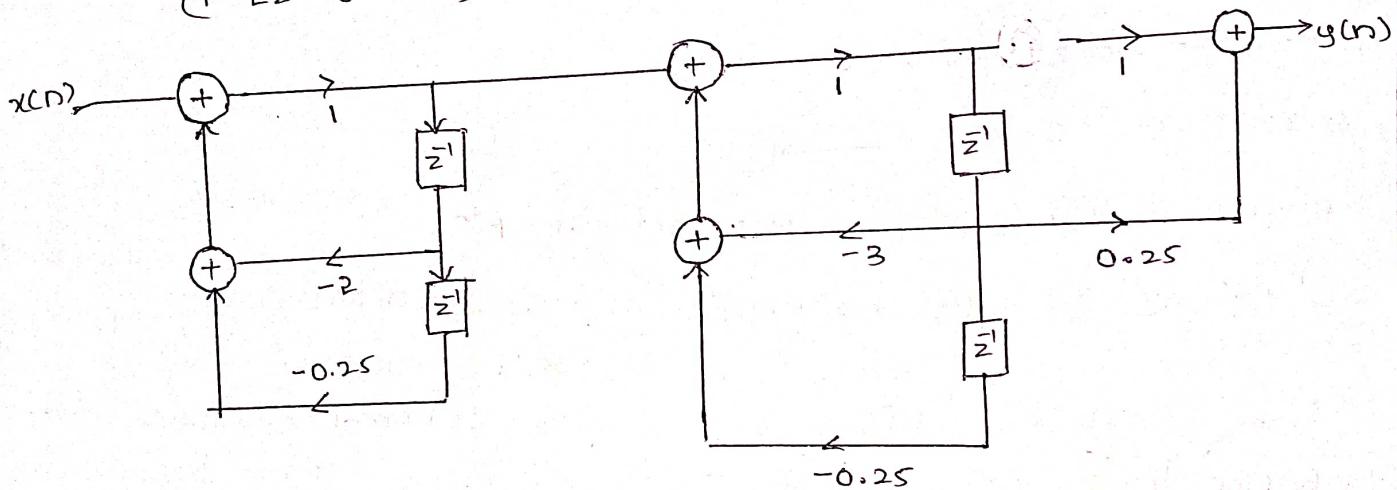
$$H(z) = \frac{1 + 0.25z^{-1}}{(1 - 2z^{-1} + 0.25z^{-2})(1 - 3z^{-1} + 0.25z^{-2})} \quad (8M)$$

Cascade:-

$$\Rightarrow \text{Given:- } H(z) = \frac{1 + 0.25z^{-1}}{(1 - 2z^{-1} + 0.25z^{-2})(1 - 3z^{-1} + 0.25z^{-2})}$$

$$H(z) = H_1(z) \cdot H_2(z).$$

$$\therefore H_1(z) = \frac{1}{(1 - 2z^{-1} + 0.25z^{-2})} \quad \text{and} \quad H_2(z) = \frac{1 + 0.25z^{-2}}{1 - 3z^{-1} + 0.25z^{-2}}.$$



Parallel form:-

$$H(z) = \frac{1 + 0.25z^{-1}}{(1 - 2z^{-1} + 0.25z^{-2})(1 - 3z^{-1} + 0.25z^{-2})}$$

$$H(z) = \frac{z^4 (1 + 0.25z^{-1})}{z^2 (1 - 2z^{-1} + 0.25z^{-2})(1 - 3z^{-1} + 0.25z^{-2})z^2}$$

$$\frac{H(z)}{z} = \frac{z^3 + 0.25z^2}{(z^2 - 2z + 0.25)(z^2 - 3z + 0.25)}$$

$$\frac{z^3 + 0.25z^2}{(z^2 - 2z + 0.25)(z^2 - 3z + 0.25)} = \frac{Az + B}{(z^2 - 2z + 0.25)} + \frac{Cz + D}{(z^2 - 3z + 0.25)}$$

$$z^3 + 0.25z^2 = (Az+B)(z^2 - 3z + 0.25) + (Cz+D)(z^2 - 2z + 0.25)$$

$$z^3 + 0.25z^2 = Az^3 - 3Az^2 + 0.25Az + Bz^2 - 3Bz + 0.25B + Cz^3 - 2Cz^2 + 0.25Cz + Dz^2 - 2Dz + 0.25D$$

$$z^3 + 0.25z^2 = (A+C)z^3 + (-3A+B-2C+D)z^2 + (0.25A-3B+0.25C-2D)z + (0.25B+0.25D)$$

Comparing coefficients on both sides.

$$A+C=1$$

$$-3A+B-2C+D=0.25$$

$$0.25A-3B+0.25C-2D=0$$

$$0.25B+0.25D=0$$

Solving above eq<sup>n</sup> we get  $A=1-C$ .

$$B+C+D=3.25 \rightarrow ①$$

$$-3B+0C-2D=-0.25 \rightarrow ②$$

$$0.25B+0.25D=0 \rightarrow ③$$

Solving eq<sup>n</sup> ①, ②, ③ we get.

$$B=0.25, C=3.25, D=-0.25 ; A=1-C=1-3.25=-2.25$$

$$H(z) = \frac{-2.25z + 0.25}{z^2 - 2z + 0.25} + \frac{3.25z - 0.25}{z^2 - 3z + 0.25}$$

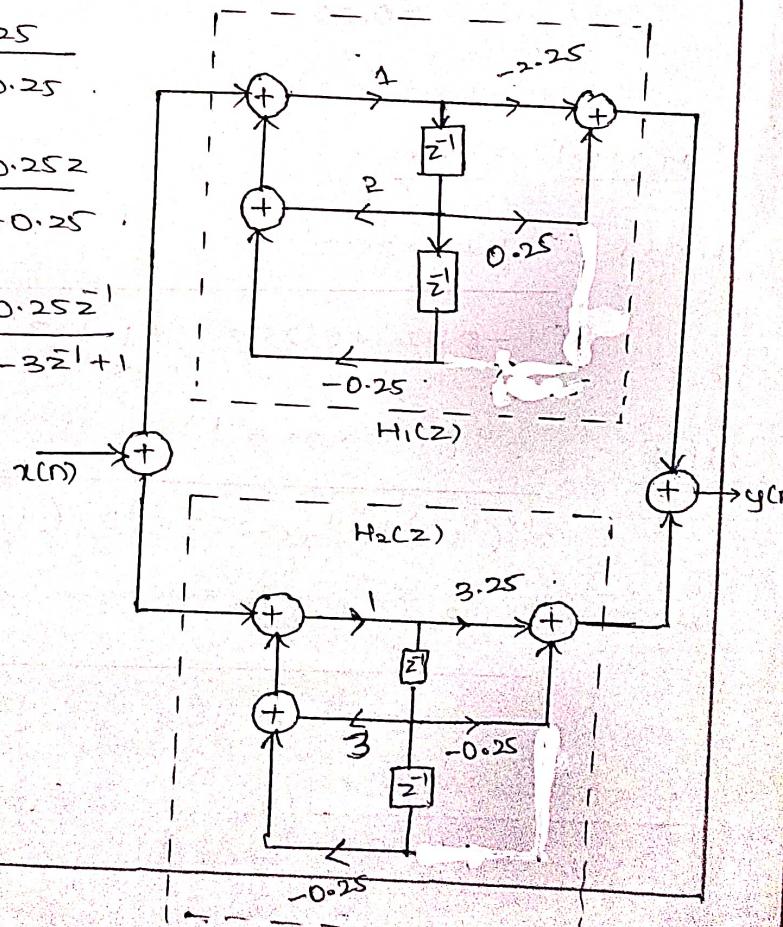
$$H(z) = \frac{-2.25z^2 + 0.25z}{z^2 - 2z + 0.25} + \frac{3.25z^2 - 0.25z}{z^2 - 3z + 0.25}$$

$$H(z) = \frac{-2.25 + 0.25z^{-1}}{0.25z^{-2} - 2z^{-1} + 1} + \frac{3.25 - 0.25z^{-1}}{0.25z^{-2} - 3z^{-1} + 1}$$

$$\Rightarrow H(z) = H_1(z) + H_2(z)$$

$$\therefore H_1(z) = \frac{0.25z^{-1} - 2.25}{0.25z^{-2} - 2z^{-1} + 1}$$

$$H_2(z) = \frac{-0.25z^{-1} + 3.25}{0.25z^{-2} - 3z^{-1} + 1}$$



## MODULE -05

9a) The frequency response of a filter is described by:  
 $H(\omega) = j\omega$ ,  $-\pi \leq \omega \leq \pi$ . Design the filter using rectangular window. Take  $N=7$ . (8M)

$\Rightarrow$  The desired impulse response is given as,

$$\begin{aligned} h_d(n) &= \frac{1}{2\pi} \int_{-\pi}^{\pi} H_d(\omega) e^{j\omega n} d\omega \\ &= \frac{1}{2\pi} \int_{-\pi}^{\pi} j\omega e^{j\omega n} d\omega \\ &= \begin{cases} \frac{\cos n\pi}{n} & n \neq 0 \\ 0 & \text{for } n=0 \end{cases} \end{aligned}$$

To have linear phase, we will shift  $h(n)$  to right. This shift is equal to  $\frac{N-1}{2} = \frac{7-1}{2} = 3$  samples i.e.

$$h(n) = \begin{cases} \cos(n\pi) & \text{for } n \neq 3 \\ 0 & \text{for } n=3 \end{cases}$$

Values of  $h(n)$  can be calculated as follows:

$$h(0) = \frac{\cos(-3\pi)}{-3} = \frac{1}{3}$$

$$h(1) = \frac{\cos(-2\pi)}{-2} = -\frac{1}{2}$$

$$h(2) = \frac{\cos(-\pi)}{-1} = 1$$

$$h(3) = 0$$

$$h(4) = \frac{\cos(\pi)}{1} = -1$$

$$h(5) = \frac{\cos(2\pi)}{2} = \frac{1}{2}$$

$$h(6) = \frac{\cos(3\pi)}{3} = -\frac{1}{3}$$

$$\therefore h(n) = \left\{ \frac{1}{3}, -\frac{1}{2}, 1, 0, -1, \frac{1}{2}, -\frac{1}{3} \right\}$$

9b) Design a low pass digital filter to be used in A/D - H(z) - D/A structure that will have -3dB cutoff at  $30\pi$  rad/sec and attenuation factor of 50dB at  $45\pi$  rad/sec. The filter is required to have a linear phase and system will use sampling frequency of 100 sample/sec. (8M)

$\Rightarrow$  i) Given :-

3 dB cutoff frequency of  $30\pi$  rad/sec i.e.

$$\omega_c = 30\pi \text{ rad/sec}$$

sampling frequency  $F_{SF} = 100 \text{ Hz}$ .

stopband attenuation of 50 dB at  $45\pi$  rad/sec i.e

$$A_s = 50 \text{ dB} \quad \text{for } \omega_s = 45\pi \text{ rad/sec}$$

ii) To obtain specifications of equivalent digital filter:

WKT  $\omega = 2\pi f$ , &  $f = \frac{F}{F_{SF}}$  hence we can write.

$$\omega = 2\pi \frac{F}{F_{SF}}$$

Let  $\omega_c = \omega_1 = 30\pi$  rad/sec &  $\omega_s = \omega_2 = 45\pi$  rad/sec.

$$\text{Hence } \omega_1 = \frac{\omega_c}{F_{SF}} = \frac{30\pi}{100} = 0.3\pi \text{ rad/sample}$$

$$\& \omega_2 = \frac{\omega_s}{F_{SF}} = \frac{45\pi}{100} = 0.45\pi \text{ rad/sample}$$

Thus we have specifications of equivalent digital filter as  
3 dB attenuation at  $\omega_1 = 0.3\pi$  rad/sample.

$$A_s = 50 \text{ dB} \text{ at } \omega_2 = 0.45\pi \text{ rad/sample}$$

iii) To select type of window:-

Here minimum stop band attenuation is 50dB. WKT hamming window provides -53 dB of stopband attenuation. Hence, hamming window is used to get required attenuation of 50dB.

iv) To determine order of the filter:-

For Hamming window, the width of the main lobe of window is  $\frac{8\pi}{M}$ , Hence  $k\left(\frac{2\pi}{M}\right) = \frac{8\pi}{M} \Rightarrow k=4$ .

The order of filter is given by  $N = k\left(\frac{2\pi}{\omega_2 - \omega_1}\right)$

$$N = 4 \left[ \frac{2\pi}{0.45\pi - 0.3\pi} \right] = 53.33.$$

Hence we design FIR filter for odd length. Hence we select next order integer  $N = 55$ .

$\therefore$  Length of filter,  $M=55$ .

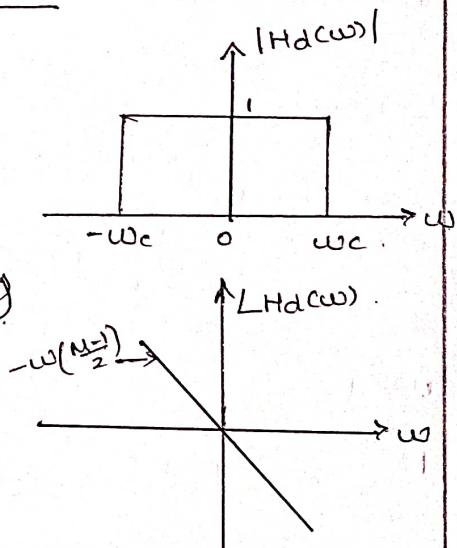
v) To obtain desired unit sample response  $h_d(n)$ :

$$H_d(\omega) = \begin{cases} e^{-j\omega\left(\frac{M-1}{2}\right)} & \text{for } -\omega_c \leq \omega \leq \omega_c \\ 0 & \text{elsewhere.} \end{cases}$$

This is ideal lowpass filter having cutoff frequency  $\omega_c$  & linear phase response of  $-\omega\left(\frac{M-1}{2}\right)$ .

The desired unit sample can be obtained by taking inverse Fourier transform of  $H_d(\omega)$  i.e.

$$h_d(n) = \frac{1}{2\pi} \int_{-\pi}^{\pi} H_d(\omega) e^{j\omega n} d\omega.$$



$$\begin{aligned} h_d(n) &= \frac{1}{2\pi} \int_{-\omega_c}^{\omega_c} e^{-j\omega\left(\frac{M-1}{2}\right)} e^{j\omega n} d\omega = \frac{1}{2\pi} \int_{-\omega_c}^{\omega_c} e^{j\omega\left(n - \frac{M-1}{2}\right)} d\omega. \rightarrow ① \\ &= \frac{1}{2\pi} \left[ \frac{e^{j\omega\left(n - \frac{M-1}{2}\right)}}{j\left(n - \frac{M-1}{2}\right)} \right]_{-\omega_c}^{\omega_c} = \frac{1}{2\pi} \left\{ \frac{e^{j\omega_c\left(n - \frac{M-1}{2}\right)} - e^{j\omega_c\left(n - \frac{M-1}{2}\right)}}{j\left(n - \frac{M-1}{2}\right)} \right\}. \\ &\approx \frac{\sin\left[\omega_c\left(n - \frac{M-1}{2}\right)\right]}{\pi\left(n - \frac{M-1}{2}\right)} \quad \text{for } n \neq \frac{M-1}{2}. \end{aligned}$$

when  $n = \frac{M-1}{2}$ , eqn ① becomes

$$h_d(n) = \frac{1}{2\pi} \int_{-\omega_c}^{\omega_c} dw = \frac{1}{2\pi} [w]_{-\omega_c}^{\omega_c} = \frac{\omega_c}{\pi}.$$

Thus we obtained  $h_d(n)$  as .

$$h_d(n) = \begin{cases} \frac{\sin[\omega_c(n - \frac{M-1}{2})]}{\pi(n - \frac{M-1}{2})} & \text{for } n \neq \frac{M-1}{2} \\ \frac{\omega_c}{\pi} & \text{for } n = \frac{M-1}{2}. \end{cases}$$

vi) To obtain  $h(n)$  by windowing:-

For hamming window ;  $w(n) = 0.54 - 0.46 \cos\left(\frac{2\pi n}{M-1}\right)$  for  $0 \leq n \leq M-1$

$h(n)$  is given as  $h(n) = h_d(n)w(n)$ .

Putting respective expressions in above equation

$$h(n) = \begin{cases} \frac{\sin[\omega_c(n - \frac{M-1}{2})]}{\pi(n - \frac{M-1}{2})} \cdot [0.54 - 0.46 \cos\left(\frac{2\pi n}{M-1}\right)] & \text{for } n \neq \frac{M-1}{2} \\ \frac{\omega_c}{\pi} [0.54 - 0.46 \cos\left(\frac{2\pi n}{M-1}\right)] & \text{for } n = \frac{M-1}{2}. \end{cases}$$

we have .

$$M=55, \text{ hence } \frac{M-1}{2} = 27.$$

thus  $\omega_1 = \omega_c = 0.3\pi \text{ rad/sample}$ .

$$h(n) = \begin{cases} \frac{\sin[0.3\pi(n-27)]}{\pi(n-27)} \cdot [0.54 - 0.46 \cos\left(\frac{\pi n}{27}\right)] & \text{for } n \neq 27 \\ 0.3 [0.54 - 0.46 \cos\left(\frac{\pi n}{27}\right)] & \text{for } n = 27. \end{cases}$$

. The values of unit sample response can be obtained from above equation to get required linear phase FIR filter .

→ Deduce the equation for the following frequency spectrum or rectangular window sequence defined by.

$$w_f(n) = \begin{cases} 1, & -\frac{N-1}{2} \leq n \leq \frac{N-1}{2} \\ 0, & \text{Otherwise.} \end{cases}$$

(BM)

⇒ DTFT of window function is given by

$$W(\omega) = \sum_{n=-\frac{N-1}{2}}^{\frac{N-1}{2}} w_f(n) e^{-j\omega n} = \sum_{n=-\frac{N-1}{2}}^{\frac{N-1}{2}} e^{-j\omega n}.$$

Here let us use  $\sum_{k=N_1}^{N_2} a^k = \frac{a^{N_1} - a^{N_2+1}}{1-a}$

$$W(\omega) = \frac{(e^{-j\omega})^{-\frac{N-1}{2}} - (e^{-j\omega})^{\frac{N-1}{2}+1}}{1 - e^{-j\omega}}$$

By rearranging above equation,

$$W(\omega) = \frac{e^{j\frac{\omega N}{2}} e^{j\frac{\omega}{2}} - e^{-j\frac{\omega N}{2}} e^{-j\frac{\omega}{2}}}{e^{j\frac{\omega}{2}} e^{-j\frac{\omega}{2}} - e^{-j\frac{\omega}{2}} e^{-j\frac{\omega}{2}}} = \frac{e^{j\frac{\omega N}{2}} - e^{-j\frac{\omega N}{2}}}{e^{j\frac{\omega}{2}} - e^{-j\frac{\omega}{2}}} = \frac{\sin(\frac{\omega N}{2})}{\sin(\frac{\omega}{2})}$$

Frequency Spectrum will be,  $|W(\omega)| = \frac{|\sin(\frac{\omega N}{2})|}{|\sin(\frac{\omega}{2})|}$

10b) A lowpass filter has the desired frequency response.

$$H_d(\omega) = \begin{cases} e^{-j\omega^3}, & 0 \leq \omega \leq \pi/2 \\ 0, & \frac{\pi}{2} \leq \omega \leq \pi. \end{cases}$$

Determine  $h(n)$  based on frequency sampling method. Take  $k=7$

(GM)

Given:  $H_d(\omega) = \begin{cases} e^{-j\omega^3}, & \text{for } 0 \leq \omega \leq \pi/2 \\ 0, & \text{for } \frac{\pi}{2} \leq \omega \leq \pi. \end{cases}$

i) To sample  $H_d(e^{j\omega})$ .

To sample  $H_d(e^{j\omega})$ , put  $\omega = \frac{2\pi k}{N}$ ,  $k=0, 1, \dots, N-1$ .

For  $N=7$ ,  $\omega = \frac{2\pi k}{7}$ ,

$\therefore H_d(e^{j\omega})$  becomes,

$$H(k) = \begin{cases} e^{-j\frac{2\pi k}{7}} & \text{for } 0 \leq \frac{2\pi k}{7} \leq \frac{\pi}{2} \\ 0 & \text{for } \frac{\pi}{2} \leq \frac{2\pi k}{7} \leq \pi \end{cases}$$

$$H(k) = \begin{cases} e^{-j\frac{6\pi k}{7}} & \text{for } 0 \leq k \leq \frac{3}{4} \\ 0 & \text{for } \frac{3}{4} \leq k \leq \frac{7}{2} \end{cases}$$

The range of 'k' in above equation can be written in nearest integer as follows.

$$H(k) = \begin{cases} e^{-j\frac{6\pi k}{7}} & \text{for } 0 \leq k \leq 2 \\ 0 & \text{for } 2 < k \leq 4 \end{cases}$$

ii) To obtain  $h(n)$

The expression for  $h(n)$  is

$$h(n) = \frac{1}{M} \left\{ H(0) + 2 \sum_{k=1}^P \operatorname{Re} [H(k) e^{j2\pi kn/M}] \right\}$$

Here  $H(0) = 1$  for  $k=0$ ,  $M=N=7$

$$P = \frac{M-1}{2} = \frac{7-1}{2} = 3.$$

Hence above equation becomes,

$$\begin{aligned} h(n) &= \frac{1}{7} \left\{ 1 + 2 \sum_{k=1}^2 \operatorname{Re} \left[ e^{-j\frac{6\pi k}{7}} \cdot e^{j2\pi kn/7} \right] \right\} \\ &= \frac{1}{7} \left\{ 1 + 2 \sum_{k=1}^2 \operatorname{Re} \left[ e^{-j2\pi k(3-n)/7} \right] \right\} \end{aligned}$$

Here  $\operatorname{Re}[e^{j\theta}]$  is  $\cos \theta$ .

$$\text{Hence } h(n) = \frac{1}{7} \left\{ 1 + 2 \sum_{k=1}^2 \cos \left[ \frac{2\pi k(3-n)}{7} \right] \right\}, n=0, 1, 2, \dots, 6.$$

10c) Realize the linear phase FIR filter having the following impulse response:

$$h(n) = \delta(n) + \frac{1}{4}\delta(n-1) - \frac{1}{8}\delta(n-2) + \frac{1}{4}\delta(n-3) + \delta(n-4) \quad (4M)$$

$\Rightarrow$  Given:-

$$h(n) = \delta(n) + \frac{1}{4}\delta(n-1) - \frac{1}{8}\delta(n-2) + \frac{1}{4}\delta(n-3) + \delta(n-4).$$

It can be written as.

$$h(n) = \left\{ 1, \frac{1}{4}, -\frac{1}{8}, \frac{1}{4}, 1 \right\}$$

Here  $h(0) = 1$ ,  $h(1) = \frac{1}{4}$ ,  $h(2) = -\frac{1}{8}$ ,  $h(3) = \frac{1}{4}$ ,  $h(4) = 1$ .

Here  $M=5$ , the impulse response is symmetric. i.e  $h(n)$  satisfy the following condition

$$h(n) = h(M-1-n)$$

i.e  $h(0) = h(4)$ ,  $h(1) = h(3)$ .

As ' $M$ ' is odd, we have to use linear phase structure.

