Page NO.

1. 1 to 33 – Mid Term
2. 34 to 60- Final Term

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25th BATCH

COMPUTER AND COMMUNICATION ENGINEERING

International Islamic University Chittagong

COURSE CODE: CCE-3601

COURSE TITLE: Digital Signal Processing

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Q) Explain the analog to digital signal conversion process with proper diagram and motive the answer briefly.

=> Analog to Digital Conversion (ADC) is the process of converting continuous analog signals into discrete digital signals. The main motive behind this conversion is to facilitate the processing, storage and transmission of signals by digital devices such as computers, smartphones, and digital audio players.

The diagram of ADC process:-

Analog Input → Sampling → Quantization → Encoding → Digital output

Sampling:- The first step ADC is to take samples of the continuous analog signal at fixed time intervals. The process is known as sampling. And it also involve measuring the amplitude of the signal at regular intervals.
The sampling rate is expressed in Hertz (Hz) which is the number of samples per second.

Quantization:-

Now. Assigning digital value to each sample. This process is called "Quantization", and it involves rounding the analog value to the nearest digital value. The number of digital values available is determined by the number of bits used for encoding.

For example, if we use 8 bits for encoding we can represent $2^8 = 256$ different values.

Encoding:-

This is the final step to convert the quantized values into a digital format. This process is called encoding. And it involves assigning a unique binary code to each digital value. The most common encoding scheme is binary. Here each value is sequenced of 0s & 1s.

The accuracy of ADC depends on the sampling rate, the number of bits used for encoding, and the quality of the quantization process. Because of ADC the signals can be manipulated by computers, and other digital devices.

Q.1)(ii) Analog signal $x(t) = 3\cos 50 * \pi(t)$ determine the maximum sampling rate required to avoid aliasing. Suppose that the signal is sampled at the rate $F = 25\,Hz$. What is the discrete time signal obtained after sampling and justify your answer briefly.

~~=> The maximum sampling rate required to avoid aliasing is the Nyquist - Shannon sampling~~

~~which is, $F_s = 2F_m = 2 \cdot 25 = 50\,Hz$~~

~~$\therefore$ $F_s = 2 \times 25 = 2\,50\pi\,Hz$~~

~~Here the signal $x(t) = 3\cos(50\pi t)$~~

=> The ~~frequency~~ signal of,

$$x(t) = 3\cos(50\pi t)$$

The frequency of the cosine function is 50 Hz.

According to the Nyquist sampling theorem the sampling rate should be at least twice the maximum frequency component of the signal which is 50 Hz in this case. Therefore, the minimum sampling rate required to avoid aliasing is:-

$F_s \geq 2 F_{max} = 2 \times 50 = 100$ Hz

Since the given sampling rate is $F = 25$ Hz. It is not sufficient to avoid aliasing. Aliasing will occur & the reconstructed signal will not accurately represent the original analog signal.

To determine the discrete time signal obtained after sampling, we can use formula:-

$x_n = x(nT)$ where, $T = 1/F$

$n =$ an integer

Substituting values

$$x_n = 3\cos(50\pi) \times \cancel{} nT)$$

$$= 3\cos(2\pi (25) nT) \quad [\because 50\pi = 2\pi(25)]$$

$$\therefore x(nT) = 3\cos(2\pi n) \quad [\because T = 1/F = 1/25 = 0.04s]$$

$\therefore$ The resulting discrete time signal

$$x(n) = 3\cos(2\pi n)$$

It is a discrete time cosine signal with a frequency of 0 Hz. which is the same as the DC. Component of the original analog signal. The high frequency components of the signal have been lost due to the undersampling, resulting in a lower-frequency discrete time signal that does not accurately represent the original analog signal. This demonstrates the effects of aliasing.

2(a). Determine the response of the following systems to the input signal $x(n)$.

$$x(n) = \begin{cases} |n|, & -3 \leq n \leq 3 \\ 0, & \text{otherwise} \end{cases}$$

(i) $y(n) = x(n-1)$ [unit delay]

(ii) $y(n) = x(n+1)$ [Unit Advanced System]

(iii) $y(n) = \frac{1}{3}[x(n+1) + x(n) + x(n-1)]$

[moving average filters]

Solve! -

(a) $y(n) = x(n) = ?$ [Identity system]

Solution:- Here, $x(n) = \{-3, -2, -1, \underset{\uparrow}{0}, 1, 2, 3\}$

(i) In this case output is exactly the same as the input signal. Such system is known as identity system.

(ii)(b) $y(n) = x(n-1)$ [Unit delay system]

$\Rightarrow$ It simply delay the system by one sample unit.

In this case $x(n-1)$ is obtained by shifting $x(n)$ to the right ~~sight~~ side by 1 unit.

$x(n) = \{\ldots, 0, -3, -2, -1, 0, \underset{\uparrow}{1}, 2, 3, 0, \ldots\}$

(iii)(c) $y(n) = x(n+1)$ [Unit advanced system]

$x(n) = \{\ldots 0, -3, -2, \underset{\uparrow}{-1}, 0, 1, 2, 3, 0 \ldots\}$

In this case $x(n+1)$ is obtained by shifting $x(n)$ to the left side by 1 unit.

(iv)(d) $y(n) = \frac{1}{3}[x(n+1) + x(n) + x(n-1)]$ [moving avg. filter]

2(b) (i) $y(n) = x(n) - x(n-1)$

is a casual equation because It describes a cause-and-effect relationship between the current sample $x(n)$ and the previ

(ii) $y(n) = \sum_{k=-\infty}^{n} x(k)$

is a non casual equation because It does not describe a cause and effect relationship between variables.

(iii) $y(n) = x(n^2)$ is a non casual cause the change in the n doesnot cha

(iv) $y(n) = x(2n)$

⇒ It is a non casual because it does not describe a cause and effect relationship between variables.

(v) $y(n) = x(n) + 3x(n+4)$

⇒ It is a casual equation because it describe a cause and effect relationship between the current $x(n)$ and a future sample $x(n+4)$.

7.5.1 Inverse Z-transform (easy)

Chapter - 1

Consider the Analog signal:-

$$x_a(t) = 3\cos 100\pi t$$

(a) Determine the minimum sampling rate required to avoid aliasing.

$\Rightarrow$

$$x_a(t) = 3\cos 100\pi t$$

we know, $x_a(t) = A\cos(2\pi F t + \theta)$

$$= 3\cos(2\cdot\pi\cdot 50\cdot + 0)$$

we know, $F_s = 2 * F_{max}$

$F = 50$ Hz

$\therefore F_s = 2 \times 50$ Hz

$= 100$ Hz

(b) The signal is sampled at the rate $F_s = 200$ Hz. What is the discrete time signal obtained after sampling?

$\Rightarrow$ Here,

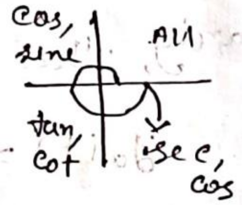
$$F_s = 200 \text{ Hz}$$

So, $$x(n) = 3 \cos \frac{100\pi}{200} \cdot n$$

$$= 3 \cos \pi/2 \cdot n$$

(c) Suppose that the signal is sampled at the rate, $F_s = 75$ Hz. What is the discrete time signal obtained after sampling?

$\Rightarrow$ Here, $F_s = 75$ Hz



$$\therefore \; x(n) = 3 \cos \frac{100\pi}{75} n$$

$$= 3 \cos \frac{4\pi}{3} n$$

$$= 3 \cos \left(2\pi - \frac{2\pi}{3} \right) n$$

$$= 3 \cos \left(2\pi/3 \right) \cdot n$$

(d) What is the frequency $0 < F < F_s/2$ of a sinusoid that yields samples identical to those obtained in part (c)?

=) The sampling rate of. $F_s = 75$ Hz

We know,

F = Sampling frequency * Sampling rate
(Frequency of sinusoid)

$= f \times F_s$

$= 75f$

From (c) we get the frequency of sinusoid $f = 1/3$

$F = 75 \times 1/3$

$= 25$ Hz

Clearly, the sinusoidal signal,

$y_a(t) = 3 \cos 2\pi F t$

$= 3 \cos 2 \cdot 25 \cdot \pi t$

$= 3 \cos 50 \pi t$

[Q2] Consider the Analog Signal,

$$x_a(t) = 3\cos 50\pi t + 10\sin 300\pi t - \cos 100\pi t$$

what is the Nyquist rate of this signal?

=) The Frequencies present in signal,

$$x_a(t) = 3\cos 50\pi t + 10\sin 300\pi t - \cos 100\pi t$$

$F_1 = 25\ Hz$, $F_2 = 150\ Hz$, $F_3 = 50\ Hz$

Here, $F_s > 2F_m = 300\ Hz$

The Nyquist rate is, $F_N = 2F_{max}$

$$\therefore F_N = 2 \times 150 = 300\ Hz$$

[Q3] Consider the Analog signal,

$$x_a(t) = 3\cos 2000\pi t + 5\sin 6000\pi t + 10\cos 12000\pi t$$

a(i) What is the Nyquist rate for this signal?

(ii) Assume now that we sample this signal using a sampling rate $F_s = 5000$ samples/s. What is the discrete time signal obtained after sampling.

=)

P.T.O

Solution

Q3(i) Here,

$$x_a(t) = 3\cos 2000\pi t + 5\sin 6000\pi t + 10\cos 12000\pi t$$

Here,

$F_1 = 1000\ Hz$ $F_2 = 3000\ Hz$ $F_3 = 6000\ Hz$

$F_{max} = 6000\ Hz$

For Nyquist rate we know,

$$F_s > 2F_{max} = 2.6000 = 12000\ Hz$$

$\therefore$ Nyquist rate $= 2.6000$

$= 12000\ Hz$

$= 12\ kHz$

Q3(ii) Here, $F_s = 5000$ samples/s $= 5000\ Hz$

~~= 5 kHz~~ We know, $x(n) = x_a(nT) = x_a(n/F_s)$

Here, the analog signal,

$$x_a(t) = 3\cos 2000\pi t + 5\sin 6000\pi t + 10\cos 12{,}000\pi t \quad \text{(i)}$$

So, sampled signal,

$$x_a(t) = 3\cos 2 \cdot \frac{1000}{5000}\pi n + 5\sin 2 \cdot \frac{3000}{5000}\pi n + 10\cos 2 \cdot \frac{6000}{5000}\pi t$$

$$= 3\cos 2 \cdot \frac{1}{5}\pi n + 5\sin 2 \cdot \frac{3}{5}\pi n + 10\cos 2 \cdot 6/5\, \pi n$$

$$= 3\cos 2\pi(2'$$

Q3(iii) what is the analog signal $y_0(t)$ that we can reconstruct from the samples if we use Ideal interpolation?

=) From B we get.

$$3\cos 2\pi(1/5)n + 5\sin 2\pi(3/5)n + 10\cos 2\pi(6/5)n$$

$$= 13\cos 2\pi(7/5)n + 5\sin 2\pi(3/5)n.$$

$$F_2^{\wedge} = F_2 - F_3$$

Digital to Analog Conversion:-

Digital to Analog Conversion (DAC) is the process of turning a digital signal into an analog signal. A digital signal is made up of binary digits (0's and 1's). An analog signal on the other hand, is a continuously varying signal like the sound waves produced by the music from a speaker.

To convert a digital signal into an analog signal, we need to recreate the continuously varying signal using a series of discrete value. This is done by taking the digital signal and turning each number into a corresponding analog value.

For example, if the digital signal has a value of 0, it is low voltage and if it has 1 it will be High.

After creating a sequence of analog values that correspond to the original digital signal, we can use those values to drive, such as speaker & motor.

So, we do it by sampling the original analog signal to obtain digital samples, reconstructing the analog signal from the digital samples, and filtering the reconstructed analog signal to remove unwanted high frequency.

☐ Folding Frequency $F_s/2 = 5/2 = 2.5 \text{ kHz}$

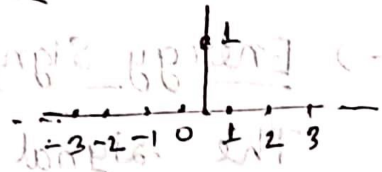
1.10

☐ A digital communication link carries binary-coded words representing samples of an input

Basic Signals in Discrete-time signals (Discrete Time Signal)

■ The **unit sample sequence** is denoted

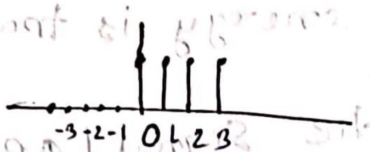
$$\delta(n) = \begin{cases} 1, & \text{For } n = 0 \\ 0, & \text{For } n \neq 0 \end{cases}$$



That means a unit sample sequence is a signal that is zero everywhere except at $n=0$ where its value is unity.

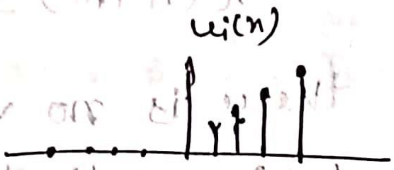
■ The unit step signal

$$u(n) = \begin{cases} 1, & \text{For } n \geq 0 \\ 0, & \text{For } n < 0 \end{cases}$$



■ The unit ramp signal,

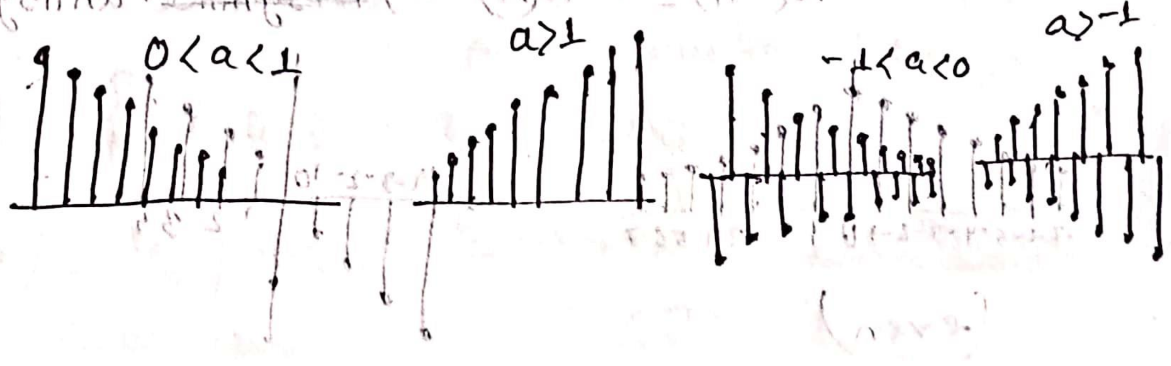
$$u_r(n) = \begin{cases} n, & \text{For } n \geq 0 \\ 0, & \text{For } n < 0 \end{cases}$$



■ The exponential signal,

$$x(n) = a^n \quad \text{For all } n$$

Here, if a is real then $x(n)$ is a real signal.



Classification of Discrete-Time Signals

Energy signals and power signals:-

Energy: Measure of the amount of work.
Power: quick work done by someone.

→ **Energy Signal:-**

The signal whose total energy is infinite but the power is finite. This means that the signal has a constant power level over time.

Power Signal:-

How quickly a work is done. or, rate at which the energy is transfered or transformed!

Periodic Signal & aperiodic

$$x(n+N) = x(n) \quad [\text{For all } n]$$

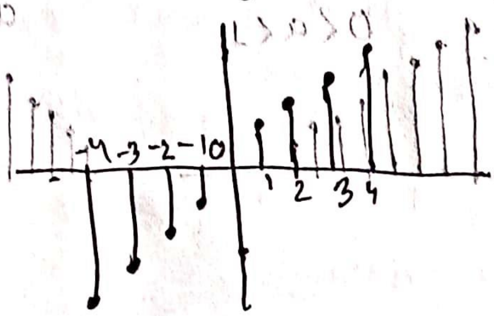
If, there is no value of N that satisfies the signal is called nonperiodic or aperiodic.

Symmetric & anti-symmetric (Odd) signals:-

A real valued signal,

$$x(-n) = x(n) \rightarrow \text{Symmetric}$$

$$x(-n) = -x(n) \rightarrow \text{~~Assymme~~ antisymmetric}$$



Example 2.2.1

Determine the response of the following systems to the input signal

$$x(n) = \begin{cases} |n|, & -3 \le n \le 3 \\ 0, & \text{otherwise} \end{cases}$$

(a) $y(n) = x(n)$ [identity system]

$$x(n) = \{\ldots 0, 3, 2, 1, \underset{\uparrow}{0}, 1, 2, 3, 0 \ldots\}$$

(b) $y(n) = x(n-1)$

$$x(n) = \{\ldots 0, 3, 2, \underset{\uparrow}{1}, 0, 1, 2, 3, 0, \ldots\}$$

(c) $y(n) = x(n+1)$ [Unit advanced system]

$$x(n) = \{\ldots 0, 3, 2, 1, 0, \underset{\uparrow}{1}, 2, 3, 0, \ldots\}$$

(d) $y(n) = \frac{1}{3}[x(n+1) + x(n) + x(n-1)]$

[moving average filter]

$\therefore y(n)$ [mean value বের করতে হবে

$y(n) = \frac{1}{3}[x(-1) + x(0) + x(1)]$]

past + present + future

$$x(n) = \{\ldots 0, 3, 2, 1, \underset{\uparrow}{0}, 1, 2, 3, 0, \ldots\}$$

$$\therefore y(0) = \frac{1}{3}[x(-1) + x(0) + x(1)]$$

$$= \frac{1}{3}\left[1+0+1\right]$$

$$= \frac{2}{3}$$

$\cancel{y(n) = \{\ldots 0, 1,}$

$$y(1) = \frac{1}{3}\left[\cancel{0} 1+0+2\right]$$

$$= 1$$

$$y(2) = \frac{1}{3}\left[1+2+3\right]$$

$$= 2$$

$$y(3) = \frac{5}{3}\left[2+3\right]$$

$$= \frac{5}{3}$$

$$y(-1) = \frac{1}{3}\left[1+2\right]$$

$$= 1$$

$$y(-2) = \cancel{\tfrac{1}{3}}\left[2\right.$$

$$y(-3) = \cancel{\tfrac{1}{3}}\left[\frac{5}{3}\right.$$

$$\therefore\ x(n) = \left\{0, 1, \frac{5}{3}, 2, 1, 0, 1, 2, \frac{5}{3}, \cancel{0}\,1, 0 \ldots\right\}$$

ঙ) $y(n) = \text{median}\,[x(n+1), x(n), x(n-1)]$ [Median Filter]

$$y(n) = \{\ldots 0, 3, 2, 1, 0, 1, 2, 3, 0 \ldots\}$$

$y(1)$

$x(n-1)\quad x(n)\quad x(n+1)$

$$y(0) = \{0, 1, 1\}$$ [ছোট হতে বড় সাজাতে হবে]

$$= 1$$

$$y(1) = \{0, 1, 2\}$$

$$= 1$$

$$y(2) = \{1, 2, 3\}$$

$$= 2$$

$$y(3) = \{\cancel{2, 3, 0}\} \quad \{0, 2, 3\}$$

$$= 2$$

~~$y(4) = \{3, 0, 1, 3\}$~~

$\therefore\ y(n) = \{0, 2, 1, 1, 1, 2, 3, 0, 0 \ldots\}$

(7) Accumulator (~~AA~~ Sum of all past value)

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

$\uparrow$ $y(0)$

$y(0) = (0+1+2+3)$

$= 6$

$y(-1) = (0+1+2)$

$= 3$

$y(-2) = (0+1)$

$= 1$

$y(-3) = 0$

$y(1) = 0+1+\cancel{3}+2+3+0$

$= 6$

$y(2) = 0+1+2+3+0+1$

$= 7$

$y(3) = 0+1+2+3+0+1+2$

$= 9$

$\therefore y(n) = \{0, 1, 3, 6, 6, 7, 9\}$

(Ans)

Block Diagram Representation of Discrete-Time Signal:-

(1) Adder:-

$$x_1(n),\ x_2(n) \rightarrow \oplus \rightarrow y(n) = x_1(n) + x_2(n)$$

(2) Multiplier:-

$$x_1(n),\ x_2(n) \rightarrow \otimes \rightarrow y(n) = x_1(n) \cdot x_2(n)$$

(3) Unit Delay element:- (requires memory, z^{-1} = unit of delay)

$$x(n) \rightarrow \boxed{z^{-1}} \rightarrow y(n) = x(n-1)$$

(4) Unit Advance element:- (z = unit of advance)

$$x(n) \rightarrow \boxed{z} \rightarrow y(n) = x(n+1)$$

Classification of Discrete-Time Systems:-

Static vs Dynamic:-

Static:- If output depends on input sample.

Dynamic:- Have memory. Can be determined by the input samples in the interval form.

② Time Invariant vs Time Variant:

→ Time Invariant don't change with time

→ Time variant changes with time

③ Linear vs Non-Linear:

→ Linear: ~~when a system is~~

$$T[a_1 x_1(n) + a_2 x_2(n)] = a_1 T[x_1(n)] + a_2 T[x_2(n)]$$

| Math-problem |

(a) $y(n) = n x(n)$ linear or non linear?

$\Rightarrow y_1(n) = n x_1(n)$

$y_2(n) = n x_2(n)$

$T[a_1 x_1(n) + a_2 x_2(n)] = \cancel{T(a_1 x_1(n)) + T}$

$= n[a_1 x_1(n) + a_2 x_2(n)]$

$= n \cdot a_1 x_1(n) + n \cdot a_2 \cdot x_2 \cdot (n)$

On the otherhand,

~~$n \cdot \{a_1 x_1(n)\} + n \cdot a_2 \cdot x_2(n)$~~ =

$a_1 y_1(n) + a_2 y_2 \cdot (n) = a_1 n \cdot x_1(n) + a_2 n \cdot x_2(n)$

(b) $y(n) = x(n^2)$

Here,

$y_1(n) = x_1(n^2)$, $y_2(n) = x_2(n^2)$

The output of the system to a linear comb. of $x_1(n)$ & $x_2(n)$

So, $y_3(n) = T[a_1 x_1(n) + a_2 x_2(n)]$

$= a_1 x_1(n^2) + a_2 x_2(n^2)$

The linear combination of the two outputs,

$a_1 y_1(n) + a_2 y_2(n) = a_1 x_1(n^2) + a_2 x_2(n^2)$

Casual & Non-casual:-

Casual:- n এর value বসানোর পর Input এ যদি output এর $y(n)$ এর n এর value হতে বাম বা সমান থাকে হয় তবেই Casual। Mathematics or DSP তে এদের Future value হিসেবে বলা হচ্ছে।

Example:-

$y(n) = x(n) - x(n-1)$

যদি, $n=1$ $\therefore$ $y(1) = x(1) - x(1-1)$

$= x(1) - x(0)$

এখানে $(x(1)$ ও $x(0) \rightarrow y(1)$ এর n মান এর সমান বা কম।

<u>Non-casual</u> :- যদি বেশি হয় তবে।

<u>Example</u> :-

① $y(n) = x(n) + 3x(n+4)$

$n = 1$

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

$= x(1) + 3x(5)$

$y(1)$ এর n হচ্ছে $x(n)$ এর n এর value বেশি, অর্থাৎ, $y(n)$ future value এর উপর dependent.

② $y(n) = x(-n)$

$n = 1$

$x(n) = x(-1)$ → যা কম

কিন্তু $n = -1$, $x(n) = x(1)$ → যা future value

তাই এটি ও Non-casual.

① Determine if the recursive system defined by the difference eqn,

$y(n) = ay(n-1) + x(n)$

$\Rightarrow$ Let, $y(n) = y_{zi}(n) + y_{zs}(n)$ ——— (i)

First check if $x(n) = c_1 x_1(n) + c_2 x_2(n)$ ——— (ii)

~~using~~ (i),

$y_{zs} = \sum_{k=0}^{n} a^k [c_1 x_1(n-k) + c_2 x_2(n-k)]$

$= c_1 y_{zs}^{(1)}(n) + c_2 y_{zs}^{(2)}(n)$

Now, it is zero-state linear

Let assume $y(-1) = c_1 y_1(-1) + c_2 y_2(-1)$

$$\therefore y_{zi}(n) = a^{(n+1)} \left[c_1 y_1(-1) + c_2 y_2(-1) \right]$$

$$= c_1 a^{n+1} y_1(-1) + c_2 y_2 a^{n+1}(-1)$$

$$= c_1 y_{zi}^{(1)}(n) + c_2 y_{zi}^{(2)}(n)$$

$\therefore$ It's zero-input linear

Chapter-3

Z-Transformation and its application to the Analysis of the LTI System

The Inverse Z-Transform:-

$$x(z) = \sum_{k=-\infty}^{\infty} x(k) \cdot z^{-k}$$

$$x(n) \xleftrightarrow{z} X(z)$$

$$x(n-K) \longleftrightarrow z^{-K} X(z)$$

$$x_1(n) = \{ \underset{\uparrow}{1}, 2, 5, 7, 0, 1 \}$$

$$x_2(n) = \{ 1, 2, \underset{\uparrow}{5}, 7, 0, 1 \}$$

$$x_3(n) = \{ \underset{\uparrow}{0}, 0, 1, 2, 5, 7, 0, 1 \}$$

VVIm 3.2.9

Compute the Convolution $x(n)$ of the signals:-

$$x_1(n) = \{ 1, -2, 1 \}$$

$$x_2(n) = \begin{cases} 1, & 0 \le n \le 5 \\ 0, & \text{elsewhere} \end{cases}$$

Q) Determine inverse Z-transform of:-

$$x(z) = \frac{1}{1 - 1.5z^{-1} + 0.5z^{-2}}$$

when, (a) ROC :- $|z| < 1$

(b) ROC :- $|z| < 0.5$

(a) $$X(z) = \frac{1}{1 - \frac{3}{2}z^{-1} + \frac{1}{2}z^{-2}}$$

$$= 1 + \frac{3}{2}z^{-1} + \frac{7}{4}z^{-2} + \frac{15}{8}z^{-3} + \frac{31}{6}z^{-4} + \ldots$$

By Comparing this relation with,

$$x(n) = \{\underset{\uparrow}{1}, \tfrac{3}{2}, \tfrac{7}{4}, \tfrac{15}{8}, \ldots\}$$

$3z^2$ $\frac{1}{2z^2}$

(b) $$\tfrac{1}{2}z^{-2} - \tfrac{3}{2}z^{-1} + 1 \overline{\big)\ 1\ } \left(2z^2 + 6z^2\right.$$

$$\underline{1 - 3z + 2z^2}$$

$$3z - 2z^2$$

$$\underline{3z - 9z^2 + 6z^3}$$

$$7z^2 - 6z^3$$

$$- \quad -$$

Chapter-6

(Frequency -Domain Analysis of LTI Systems)

Chapter-6

[Sampling and reconstruction of sampling]

Chapter-2

Convolution method :-

Question:- Determine the system's response $y(n)$ graphically and analytically for the following signals using the convolution method.

$$h(n) = \{1, 2, 1, -3\}$$

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

Where $h(n)$ is the impulse response of LTI system $x(n)$ is the input signal. Justify the answer.

Solution:-

Determine z-transform & their ROC of the following discrete time signals.

(a) $x(n) = \{ \underset{\uparrow}{3}, 2, 5, 7 \}$

So,

$x(0) = 3$

$x(1) = 2$

$x(2) = 5$

$x(3) = 7$; and $x(n) = 0$ for $n < 0$ & for $n > 3$.

By definition of Z-transform,

$$Z\{x(n)\} = X(z) = \sum x(n) z^{-n}$$

$$\therefore X(z) = x(0) \cdot z^{0} + x(1) \cdot z^{-1} + x(2) \cdot z^{-2} + x(3) \cdot z^{-3}$$

$$= 3 + \frac{2}{z} + \frac{5}{z^2} + \frac{7}{z^3}$$

In $X(z)$, if $z = 0$, except the first one all other will be infinity.

(2) Determine Z-transform of

$x(n) = \{ \underset{\uparrow}{0}, 0, 1, 2, 5, 7, 0, 1 \}$

$x(0) = 0,$
$x(1) = 0$
$x(2) = 1$
$x(3) = 2$
$x(4) = 5$
$x(5) = 7$
$x(6) = 0$
$x(7) = 1$

We know, $X(z) = \sum x(n) \cdot z^{-n}$

So, $X(z) = x(0) \cdot z^{-0} + x(1) \cdot z^{1} + x(2) \cdot z^{-2} + x(3) \cdot z^{-3}$
$+ x(4) \cdot z^{-4} + x(5) \cdot z^{-5} + x(6) \cdot z^{-6} + x(7) \cdot z^{-7}$

$= 0 + 0 + 1 \cdot \frac{1}{z^2} + \frac{2}{z^3} + \frac{5}{z^3} + \frac{7}{z^3} + 0 + \frac{1}{z^7}$

Except first two if $z = 0$ then all other terms will become Infinity.

(ii) $x(n) = \{ 2, 4, \underset{\uparrow}{5}, 7, 3 \}$

$x(0) = 5$
$x(1) = 7$
$x(2) = 3$
$x(-1) = 4$
$x(-2) = 2$

we Know,

$$x(z) = \sum x(n) \cdot z^{-n}$$

~~$x(z) = x(0) \cdot z^0 + x(1) z^{-1} + x(2)$~~

$$x(z) = x(-2) \cdot z^{2} + x(-1) \cdot z^1 + x(0) z^0 + x(1) z^{-1} + x(2) \cdot z^{-2}$$

$$= 2 \cdot z^2 + 4 \cdot z + 5 \cdot 1 + 7/z + 3/z^2$$

If $z = \alpha$ except the third the rest will be infinity.

$\boxed{\text{properties of z-transform}}$

Linearity: - If $x_1(n)$ & $x_2(n)$ are two input signal and a_1, a_2 = Constants

then,

$$z\{a_1 x_1(n) + a_2 x_2(n)\} = a_1 X_1(z) + a_2 X_2(z)$$

Time Shifting :- If $x(n)$ is a signal, the shifting by k samples in time time domain,

$$z\{x(n-m)\} = z^{m} \cdot X(z)$$

$$z\{x(n+m)\} = z^m X(z)$$

~~Scaling~~ :- Time reversal:- z-transform of $x(-n)$ is equal to the complex conjugate of the z-transform.

if, $z\{x(n)\} = X(z)$

$\therefore z\{x(-n)\} = X(z^{-1})$

Convolution:-

$$Z\{x_1(n) * x_2(n)\} = X_1(z) \cdot X_2(z)$$

Initial value Theorem:-

Signal x(n) evaluated at z=1 is

$$Z\{x(n)\} = X(z)$$

$$x(0) = \lim_{z \to \infty} X(z)$$

Final value Theorem:-

$$x(\infty) = \lim_{z \to 1} (1 - z^{-1}) X(z)$$

$$= \lim_{z \to 1} \left(\frac{z-1}{z}\right) X(z)$$

Scaling:-

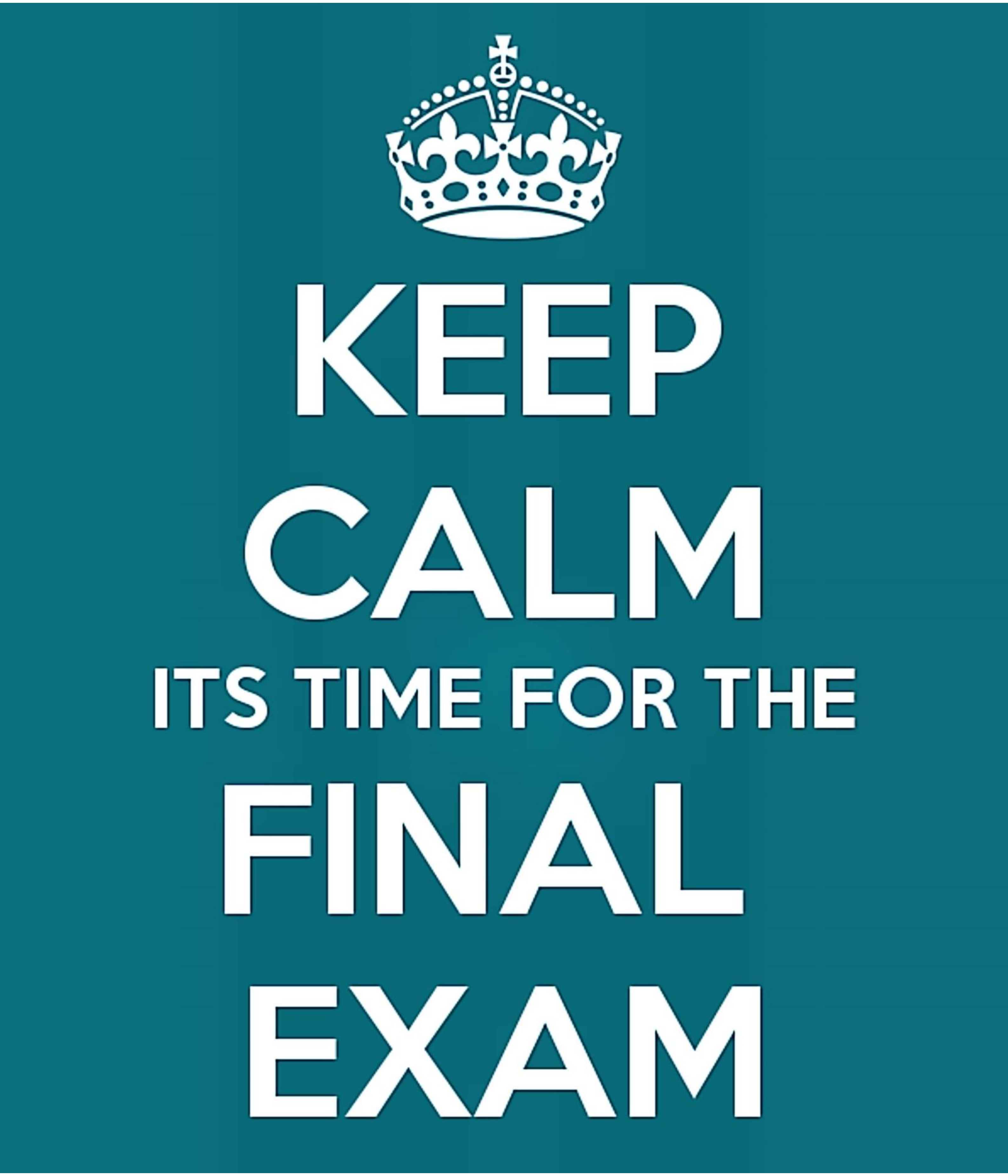
$$x(n) = \{\underset{\uparrow}{1}, 2, 3, 4\}$$

$$x(z) = 1 + 2 \cdot z^{-1} + 3 \cdot z^{-2} + 4 \cdot z^{-3}$$

Scale the signal by 2,

$$y(z) = 2x(z) = 2 + 4z^{-1} + 6z^{-2} + 8z^{-3}$$

$$\begin{array}{r} 32 \\ \times 31 \\ \hline 32 \\ 960 \\ \hline 992 \end{array}$$


KEEP
CALM
ITS TIME FOR THE
FINAL
EXAM

Final

SAKUR_E201037

Lecture-01

Introduction to Analog Filter:-

Passive Analog Filters (Ideal): (Made up of passive components. Registors, Capacitors, inductors)

(i) Losspass Filter:-

Low Frequency signal pass হচ্ছে এই filter দ্বারা যেখানে High Frequency কে কমানো হচ্ছে।

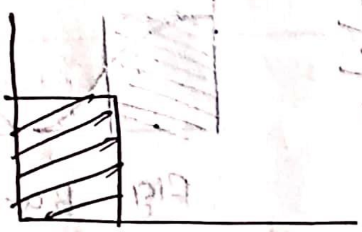


Fig: Lowpass Filter

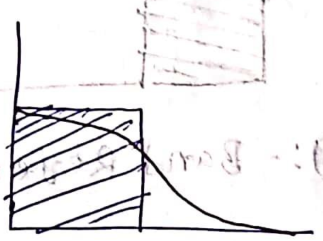


Fig: Low pass (Realistic)

(ii) Highpass Filter:-

High-pass Frequency signal কে allow করে, Low Frequency এর signal কে কমায়।

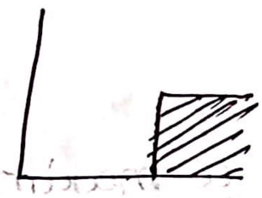


Fig:- Highpass Filter.

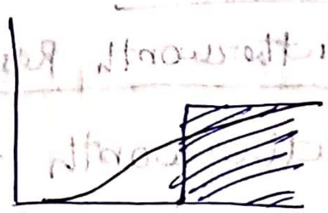


Fig:- Highpass (Realistic)

(iii) Band-pass Filter:-

Specific Range Frequency pass করে, Outside Range Frequency বাদ দেয়।

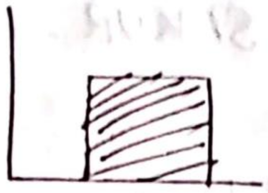


Fig:- Band pass

Fig:- Bandpass (Realistic)

Band-reject/stop filter:-

specific range frequency বাদ, অন্যান্য Range এর Frequency pass করে।

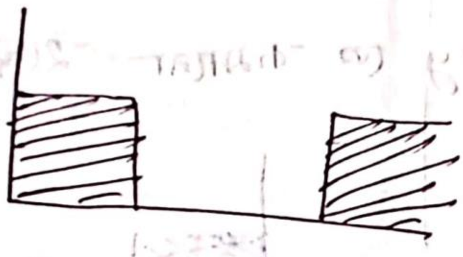


Fig:- Band Reject

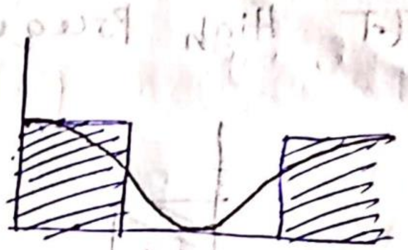


Fig:- Band Reject (Realistic

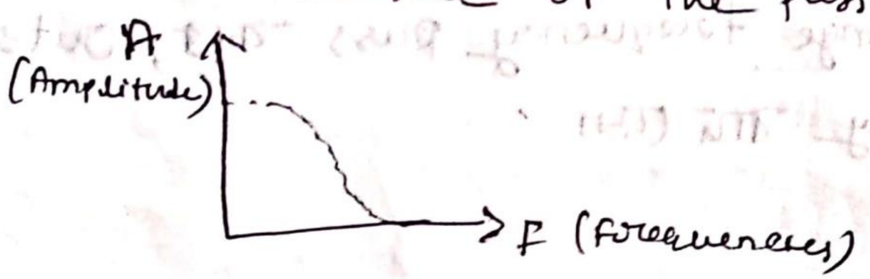
Filter response characteristic:-

A graph that shows the amplitude of a filters output signal as a function of its input function.

Three types:-

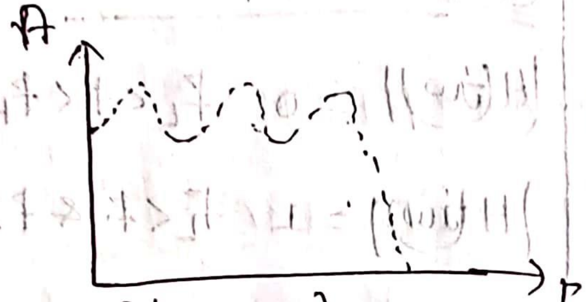
(i) Butterworth Response:-

Butterworth filters have a maximally flat passband. They take long time to attenuate frequencies outside of the pass band.

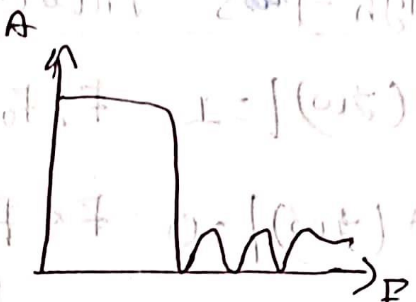


Chebyshev filter: (i) It is steeper roll-off than butterworth. That means এটা outside frequencies কেও বাদ (attenuate) করে more quickly.

(ii) It has ripples in the passband as a result the amplitude of the output signal is not the same for all frequencies in the passband.



(type-1)



type-2

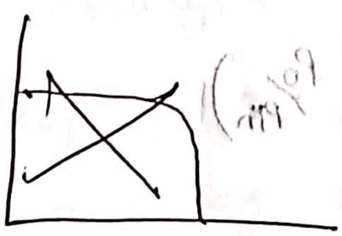
Bessel Filter:

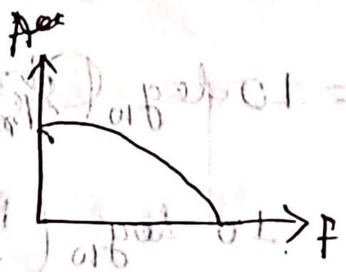
→ Constant phase delay থাকে।

→ সব গুলো frequency passband এ same amount এর delay হয়।

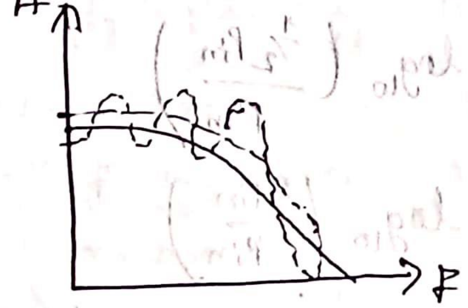
→ Chebyshev এর ~~থেকে~~ হতে slow roll-off

→ But don't have any ripple





All together:



Frequency transfer function of filter $H(j\omega)$:-

① Low-pass filter:-

$|H(j\omega)| = 1 \quad f < f_0$

$|H(j\omega)| = 0 \quad f > f_0$

② High-pass filter:-

$|H(j\omega)| = 1 \quad f > f_0$

$|H(j\omega)| = 0 \quad f < f_0$

③ Band pass filter:-

$|H(j\omega)| = 0 \quad f < f_L \ \& \ f > f_H$

$|H(j\omega)| = 1 \quad f_L < f < f_H$

④ Band-stop filter:-

$|H(j\omega)| = 0 \quad f_L < f < f_H$

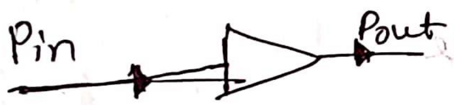
$|H(j\omega)| = 1 \quad f_L < f \ \& \ f > f_H$

⑤ All pass or phase shift filter:-

$H(j\omega) = 1$

For all f has a specific phase response.

Power gain in dB:-



$$A_p(dB) = 10 \log_{10}\left(\frac{P_o}{P_{in}}\right) \ \left(P_o/P_{in}\right)$$

$$\Rightarrow 0\,dB = 10 \log_{10}\left(\frac{P_{in}}{P_{in}}\right)$$

$$\Rightarrow -3dB = 10 \log_{10}\left(\frac{\frac{1}{2}P_{in}}{P_{in}}\right)$$

$$\Rightarrow +3dB = 10 \log_{10}\left(\frac{2P_{in}}{P_{in}}\right)$$

Of voltage gain in dB:- $P = \left(\frac{V^2}{R}\right)$

~~Act~~

$$A_v(dB) = 20\log_{10}\left(\frac{V_o}{V_{in}}\right)$$

$$\Rightarrow 0\,dB = 20\log_{10}\left(\frac{V_{in}}{V_{in}}\right)$$

$$\Rightarrow -6\,dB = 20\log_{10}\left(\frac{\frac{1}{2}V_{in}}{V_{in}}\right)$$

$$\Rightarrow +6\,dB = 20\log_{10}\left(\frac{2V_{in}}{V_{in}}\right)$$

Lecture-03 Book: A Nagoor Kani

DISCRETE FOURIER Transform (DFT)
Fast Fourier Transform (FFT)

DFT:- ~~DFT of a discrete~~

DFT is developed to convert a continuous function of ω to a discrete function of ω. So that, frequency analysis of discrete time signals can be performed on a digital system.

It is obtained by sampling the DTFT of the signal at uniform frequency intervals & the number of samples should be sufficient to avoid aliasing of frequency spectrum. DFT is a sequence of complex number.

$X(k)$, $k = 0, 1, 2, 3$, etc...

- The plot of magnitude vs k is called Magnitude spectrum
- The plot of phase vs k is called phase spectrum.
- In general these plots are called Frequency spectrum.

Fast Fourier Transform (FFT):-

DFT কে compute করতে গেলে, a large number of calculation প্রয়োজন হয়। ফলে DFT computation যদি বড় হয় তবে calculation এর পরিমান ও বাড়ে। যা প্রচুর সময় লাগে। আর এ থেকে পরিত্রাণ পেতেই FFT এর আবিষ্কার।

Definition of DFT:-

$x(n)$ = Discrete time signal of Length (L)

$X(K)$ = DFT of $x(n)$

- N-point DFT of $x(n)$ can be expressed

$$= \text{DFT}\{x(n)\}$$

$$\therefore \text{DFT}\{x(n)\} = X(K) = \sum_{n=0}^{N-1} x(n) \cdot e^{\frac{-j2\pi kn}{N}} \quad ; \text{ for } K = 0, 1, 2, \ldots N-1$$

The DFT of $x(n)$ can be expressed,

$$X(K) = \{X(0), X(1), X(2) \ldots X(N-1)\}$$

$$x(2) = \frac{1}{3}\left[1 + \cos\frac{2\pi}{4} - j\sin\frac{2\pi}{4} + \cos\frac{2\pi}{2} - j\sin\frac{2\pi}{2}\right]$$

$$= \frac{1}{3}\left[1 + \cos\frac{\pi}{2} - j\sin\frac{\pi}{2} + \cos\pi - j\sin\pi\right]$$

$$= \frac{1}{3}\left[1 + 0 - j1 - 1 - 0\right]$$

$$= \frac{1}{3}\{-j\}$$

$$= -j\,0.333$$

$$= 0.333 \angle -\pi/2$$

$$= 0.333 \angle -0.5\pi$$

Lecture - 04

Analog Filter Type Summary:-

① Butterworth Filter:-

Maximally flat passband and a minimally smooth stopband. They are characterized by a constant gain in the passband and monotonically decreasing in the stopband.

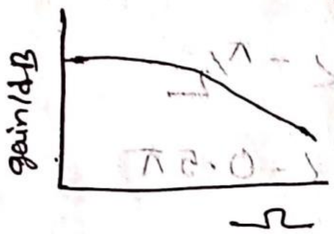


Fig: Butterworth

② Chebyshev Filter:-

Lecture Sheet-03

Math Design a Butterworth filter with 1 dB cutoff at 1 KHz & a minimum attenuation of 40 dB at 5 KHz

Ans:- Assume peak passband gain = 1

Minimum passband gain $= \dfrac{1}{\sqrt{1+\varepsilon^2}}$

ripple $\alpha_{max} = 20\log_{10}\sqrt{1+\varepsilon^2}$ dB

Minimum stopband attenuation

$$\alpha_s = -20\log_{10} \frac{1}{A}$$

$$= 20\log_{10} A \text{ dB}$$

Design equation:-

$$N \geq \frac{1}{2}\,\frac{\log_{10}\left(\frac{A^2-1}{\varepsilon^2}\right)}{\log_{10}\left(\frac{\Omega_s}{\Omega_p}\right)}$$

$$\frac{1}{\sqrt{1+\varepsilon^2}} = 0 - 1$$

$$-\sqrt{1+\varepsilon^2}$$

$$-1 = \ldots \quad -2 = \varepsilon^2$$

$$-1\,dB = 20\log_{10}\frac{1}{\sqrt{1+\varepsilon^2}}$$

$$\Rightarrow \cancel{\sqrt{(1+\varepsilon^2)} = \frac{20\log_{10}}{-1}}$$

$$= 0$$

$$\Rightarrow \cancel{\varepsilon^2 = 1}$$

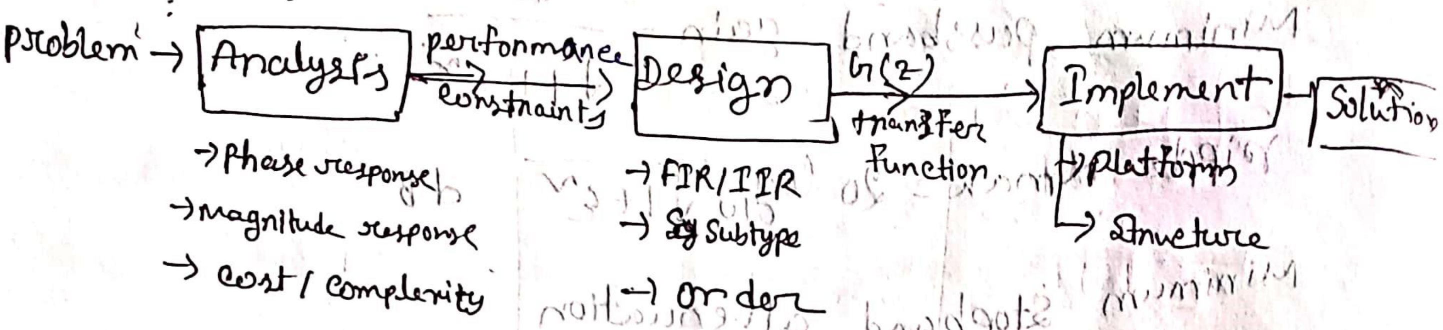
$$\Rightarrow -1 = 20\log_{10}(1) - 20\log_{10}(1+\varepsilon^2)$$

$$\Rightarrow -1 = 0 - 20\log_{10}(1) + 20\log_{10}\varepsilon^2$$

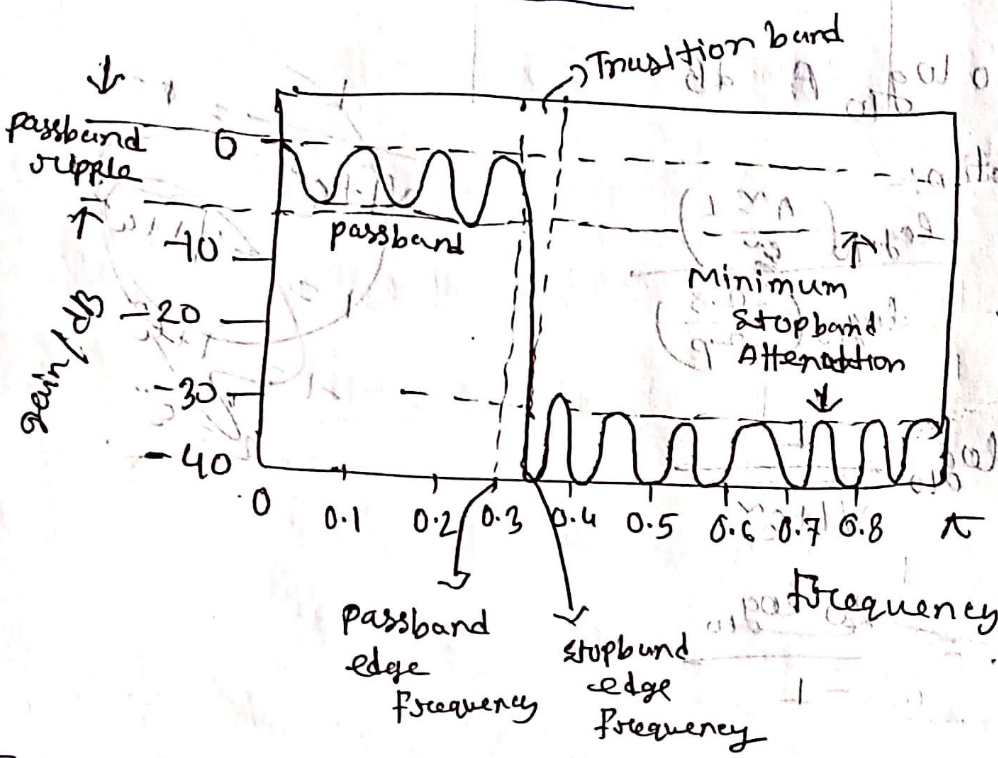
$$\Rightarrow -1 = 20\log_{10}\varepsilon^2$$

Filter Design specification :-

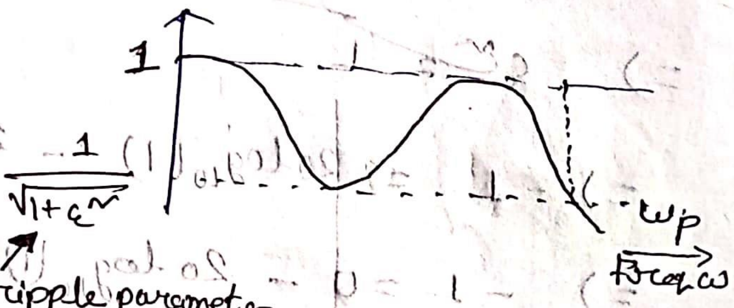
Design process :-



performance constraints :-



passBand ripple :-

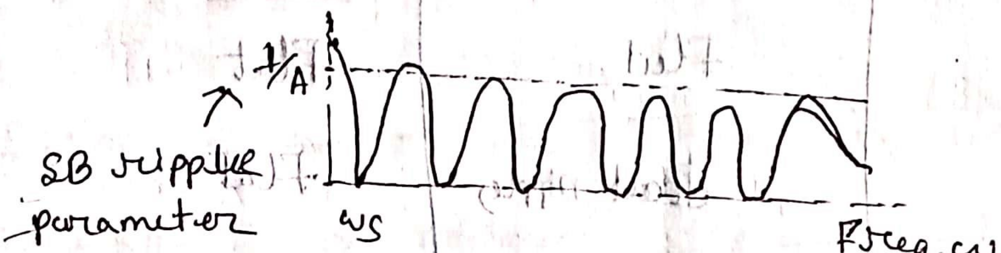


* peak passband gain = 1

∴ Minimum passband gain $= \frac{1}{\sqrt{1+\epsilon^2}}$

⇒ ripple $\alpha_{max} = 20\log_{10}\sqrt{1+\epsilon^2}$ dB

(*) Stopband ripple! –



→ peak passband gain is A × larger than peak stopband gain.

→ Minimum stopband attenuation, $\alpha_s = -20\log_{10} \frac{1}{A}$

$= 20\log_{10} A$ dB

(#) FIR VS IIR :-

FIR	IIR
(1) No feedback (just zeros)	(1) Feedback (poles & zeros)
(2) Always stable	(2) May be unstable
(3) Can be linear phase	(3) Difficult to control phase
(4) Unrelated to continuous time filtering	(4) Derive from analog prototype
(5) not Low complexity	(5) ~~not~~ Low Comp.
(6) phase ~~not~~ important use FIR	(6) Not important → use IIR

Analog Filter Design:-

Basic choices:-

More ripples → Narrow transition band

Family	PB	SB
Butterworth	Flat	Flat
Chebyshev-1	~~Flat~~ ripples	Flat
Chebyshev-2	Flat	ripples
Elliptical	ripples	ripples

Butterworth Filters:-

Maximally Flat in pass & stop band

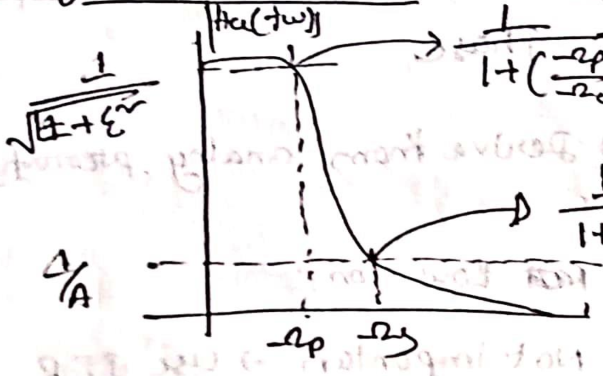
Magnitude response, $|H_a(j\Omega)|^2 = \dfrac{1}{1+\left(\frac{\Omega}{\Omega_c}\right)^{2N}}$ $\boxed{N = \text{filter order}}$

(i) $\Omega << \Omega_c$ then, $|H_a(j\Omega)|^2 \to 1$

(ii) $\Omega = \Omega_c$ then, $|H_a(j\Omega)|^2 = 1/2$

(iii) $\Omega >> \Omega_c$ the, $|H_a(j\Omega)|^2 \to \left(\Omega_c/\Omega\right)^{2N}$

Design Specification:-



$$\frac{1}{1+\left(\frac{\Omega_p}{\Omega_c}\right)^{2N}} \geq \frac{1}{1+\varepsilon^2}$$

$$\frac{1}{1+\left(\frac{\Omega_s}{\Omega_c}\right)^{2N}} \leq \frac{1}{A^2}$$

$$N \geq \frac{1}{2}\,\frac{\log_{10}\left(\frac{A^2-1}{\varepsilon^2}\right)}{\log_{10}\left(\frac{\Omega_s}{\Omega_p}\right)}$$

K_1

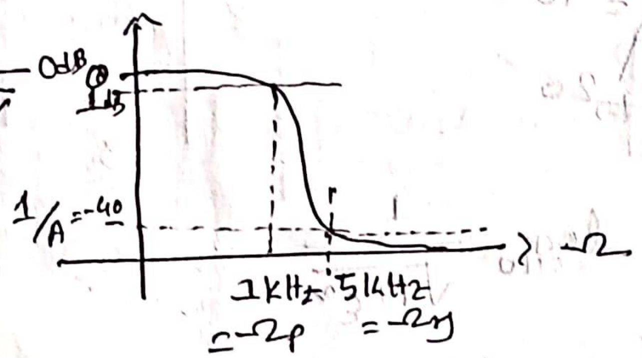
selectivity < 1

$$K = \frac{\Omega_p}{\Omega_s} = \cancel{\frac{\frac{1}{1+\epsilon^2}}{\frac{1}{A^2}}} = \cancel{\frac{1}{1+\epsilon^2} \times A^2}$$

$$k_1 = \frac{\epsilon}{\sqrt{A^2-1}}$$ discrimination << 1

Q) Design a butterworth filter with 1 dB cutoff at 1 KHz & a minimum attenuation of 40 dB at 5 KHz

=>



$-1dB = \frac{1}{\sqrt{1+\epsilon^2}} =$

$-1dB = -0.282$

$$-1dB = 20\log_{10}\frac{1}{\sqrt{1+\epsilon^2}}$$

$$-40dB = 20\log_{10}\frac{1}{A}$$

$$-\frac{40}{20} dB = \log_{10}\frac{1}{A}$$

$$\Rightarrow -2\ dB = \log_{10}\frac{1}{A}$$

$$\Rightarrow (10)^{-2}\ dB = 1/A$$

$$\Rightarrow \frac{1}{100} = 1/A \Rightarrow A = 100$$

$N = 4 \geq 3.28$

(Ans)

Chebyshev-1 Filter:-

Equiripple in passband (Flat in stopband)

→ minimize maximum error

Formulae:- $|H_a(j\Omega)|^2 = \dfrac{1}{1+\varepsilon^2 T_N^2\left(\frac{\Omega}{\Omega_p}\right)}$

Chebyshev Polynomial of order N, $T_N(\Omega) = \begin{cases} \cos(N\cos^{-1}\Omega) & |\Omega| \leq 1 \\ \cosh(N\cosh^{-1}\Omega) & |\Omega| > 1 \end{cases}$

ε = Desired passband ripple

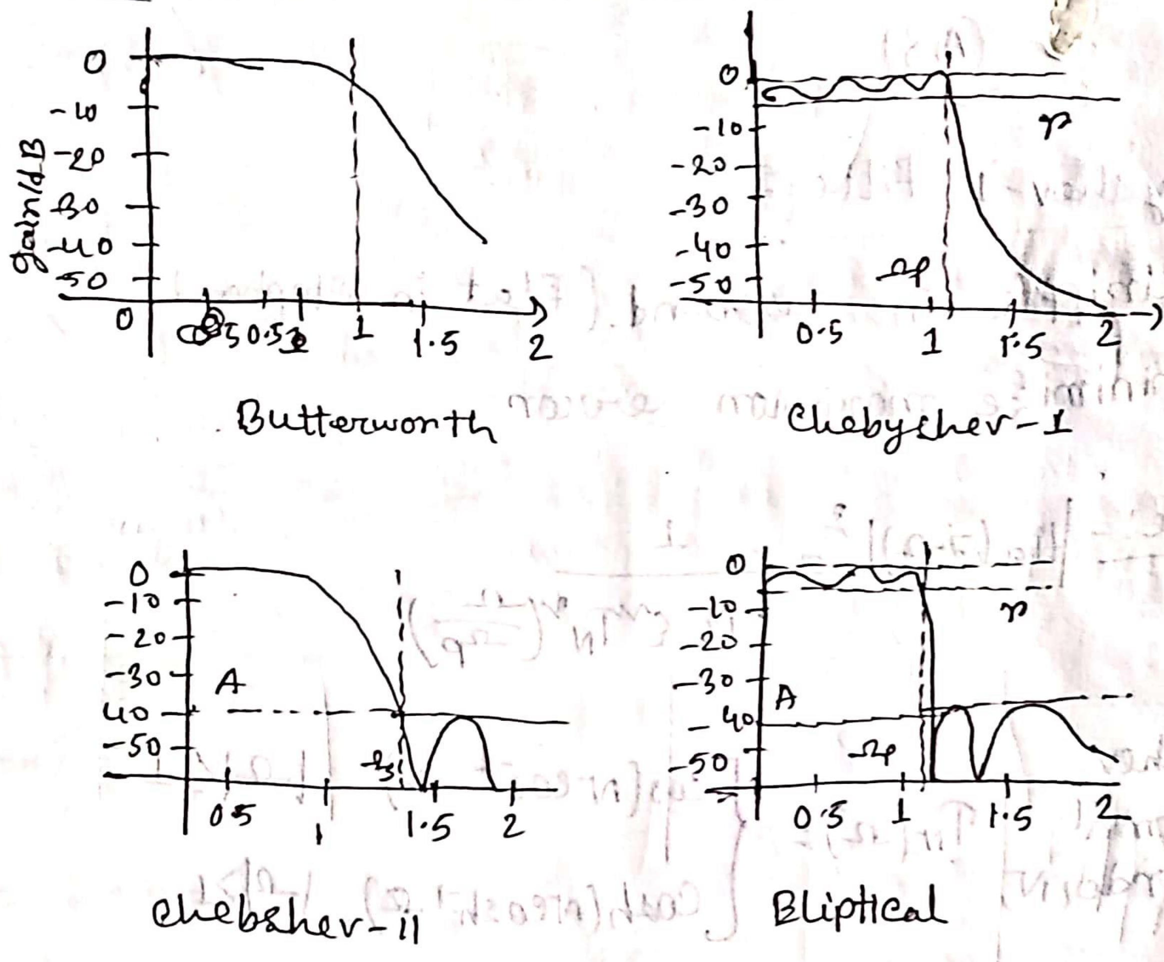
Ω_p, Ω_s = Minimum stopband attenuation

$$\frac{1}{A^2} = \frac{1}{1+\varepsilon^2 T_N^2\left(\frac{\Omega_s}{\Omega_p}\right)}$$

$$= \frac{1}{1+\varepsilon^2\left[\cosh(N\cosh^{-1}\Omega)\right]^2} \quad [|\Omega| > 1]$$

$$\Rightarrow N \geq \frac{\cosh^{-1}\left(\frac{\sqrt{A^2-1}}{\varepsilon}\right)}{\cosh^{-1}\left(\frac{\Omega_s}{\Omega_p}\right)} \begin{array}{l} \rightarrow 1/k_1 \text{ discrimination} \\ \rightarrow 1/k \text{, selectivity} \end{array}$$

Analog filter type summary:-



Bilinear Transformation:-

Solution : $S = \dfrac{1 - z^{-1}}{1 + z^{-1}}$

inverse, $z = \dfrac{1+s}{1-s}$

Frequency axis, $s = j\Omega$

$\therefore z = \dfrac{1+j\Omega}{1-j\Omega}$

Poles, $s = \sigma + j\Omega \rightarrow z = \dfrac{(1+\sigma)+j\Omega}{(1-\sigma)-j\Omega}$

$\therefore |z|^2 = \dfrac{1+2\sigma+\sigma^2+\Omega^2}{1-2\sigma+\sigma^2+\Omega^2}$

⊛ CT ⟷ DTF Frequency relation:-

$\Rightarrow z = e^{j\omega}$

$$\Rightarrow s = \frac{1-e^{-j\omega}}{1+e^{-j\omega}} = \frac{2j\sin\omega/2}{2\cos\omega/2} = j\tan\omega/2$$

$\Omega = \tan(\omega/2)$

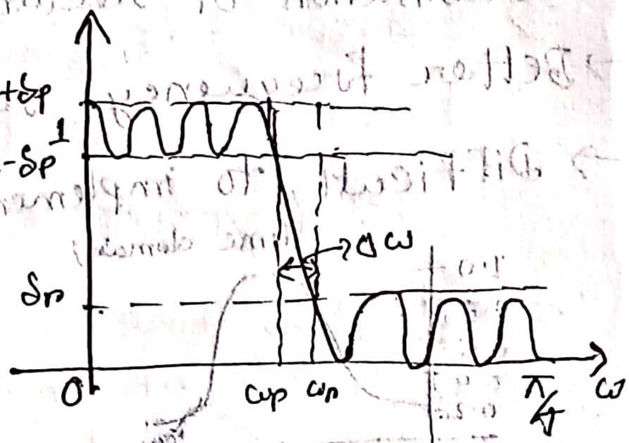
$\omega = 2\tan^{-1}\Omega$

⊛ Infinite range of CT Frequency, $-\infty < \Omega < \infty$ maps to finite DT Freq range $-\pi < \omega < \pi$

⊛ Non linear, $\frac{d}{d\omega}\Omega \to \infty$ as $\omega \to \pi$

Lecturesheet-4

⊡ Filter specifications:-



δp = peak passband deviation

δs = Stopband deviation

ωp = passband edge frequency $(2\pi F_p)$

ωs = Stopband edge frequency $(2\pi F_p)$

ωs = Sampling frequency $(\omega s = 2\pi F_s)$

⊛ FIR Coefficient calculation:-

$$y(n) = \sum_{k=0}^{N} h(k) \cdot x(n-k)$$

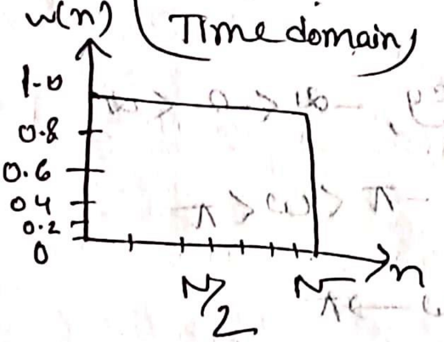
$$H(z) = \sum_{k=0}^{N} h(k) \cdot z^{-k}$$

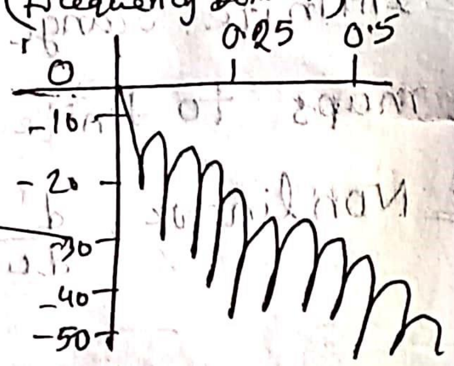
~~Hamming window~~

NVIm

Window Method

Rectangular:- Simplest type of window, Value 1 for specified range & zero = 0 for outsider range. Window has very easy to implement. Have the disadvantage of having poor frequency resolution.



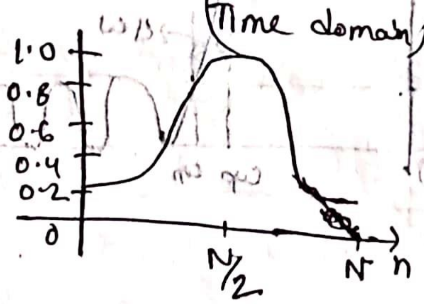


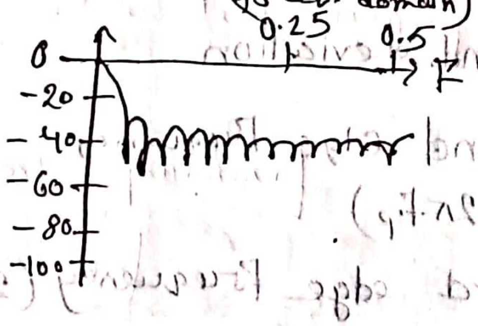
2) **Hamming window:-**

→ Combination of rectangular ~~function~~ window & cosine function

→ Better frequency resolution than Rectangular.

→ Difficult to implement.



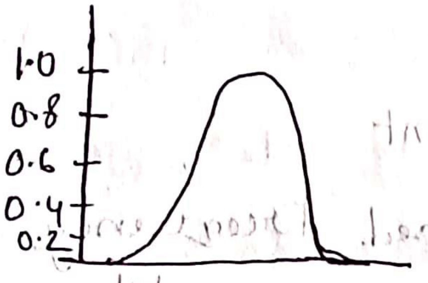


(3) **Blackman:-**

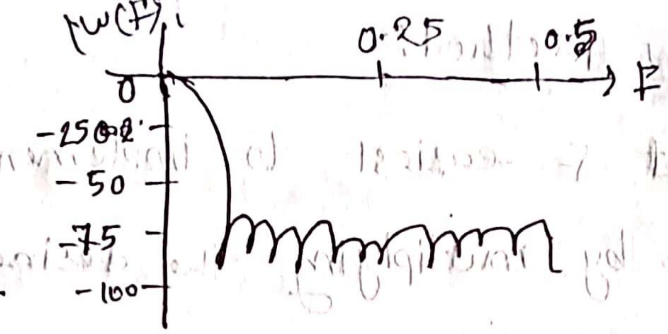
→ Used for spectral analysis.

→ Combination of rectangular & window & cosine function.

→ If frequency resolution is the most important ~~function~~ factor, then hamming / blackman good choice.



(Time domain)



(Frequency domain)

Hamming window (eqn & math)

For-1:-

$$\rightarrow w(n) = \begin{cases} 0.54 + 0.46\cos(2\pi n/N) & ; 0 \le n \le N \\ 0 & ; \text{elsewhere} \end{cases}$$

Formula-2:-

$\rightarrow \Delta F = 3.32/N$

$\rightarrow N$ = filter order

$\rightarrow \Delta F$ = Normalized transition width = $\Delta F / F_s$

ΔF → Transition width

VVSm Comparison of the window, optimum & frequency sampling methods:-

=> **Optimal Method:-**

=> FIR filter coefficients compute করতে সহায়ক।

=> N value & making filter with good amplitude response.

=> Most accurate than other three

=> Can achieve the desired frequency response with no ripple.

=> Difficult to implement

=> Linear equires computation is expensive.

(2) Window Method:-

=) Simplest & easiest to implement

=) works by multiplying the desired frequency response

=) Tapers the ~~fre~~ response at the edges in which reduce ripples in frequency response.

=) Can introduce significant ripple in the frequency if window function is not chosen carefully.

(3) Frequency sampling method:-

-) Compromise between the window & optimum Methods.

=) It is less difficult than ~~optimum~~ optimum window but more difficult than window.

=) works by sampling desired frequency at a finite number of points.

-) Coefficients are calculated using DFT.

uses:-

-) ease of implement -) $\boxed{\text{window}}$

-) Accuracy -) $\boxed{\text{Optimum}}$

-) Comprises between ease of implementation & accuracy -) $\boxed{\text{Frequency Sampling method}}$

[4]

-2, -2

$-\cancel{\pi+}\,\pi/4 \quad (\pi + \pi/4)\ \tan^{-1}$

$-\pi + \pi/4$

$\frac{-4\pi+\pi}{4} \quad -\frac{3\pi}{4}$

$\tan^{-1}\left(\frac{0}{-2}\right)$

$\rightarrow$ 8.7 - 8.13

DFT $\rightarrow$ 9.19

(*) FIR Filter Math + Solution pdf (3 (a)), 3

(*) 8. Butterworth ([illegible])

$$10 = 20 \log_{10} \frac{1}{\sqrt{1+\epsilon^2}}$$

$$\frac{1}{2} = \log_{10} \frac{1}{\sqrt{1+\epsilon^2}}$$

$$(10)^{\frac{1}{2}} = \frac{1}{\sqrt{1+\epsilon^2}}$$

$$3.1623 = \frac{1}{\sqrt{1+\epsilon^2}}$$

$$\sqrt{1+\epsilon^2} = \frac{1}{1.78}$$

$$\epsilon^2 =$$

Multirate

Multirate:-

Multirate means 'Multiple sampling rate'. DSP system uses it in case signal at one point rate has to use systems that expects a different rate.

So, the rate needs to be increased or decreased & some processing is required to do so.

Resampling:-

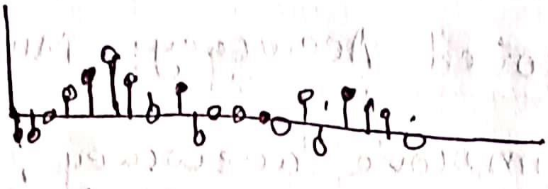
process of converting a signal from one sampling rate to another. This is often done to increase or decrease the resolution of the signal or to change the sampling rate to match the requirement of particular application.

Downsampling:-

- process of removing samples,
- without Low pass Filtering, ~~a signal is downsampled~~
- Downsampled when the signal is over sampled.
- **Decimation**:- Combined operation of both Filtering & Downsampling.
- To downsample by a factor of M, we must keep every Mth sample as it is & remove the (M-1) samples in between.

$x[n] \rightarrow \boxed{\downarrow M} \rightarrow x_d[n]$

(Downsampler)



$(M = 4)$

$$\underset{F_T}{x[n]} \rightarrow \underset{\text{LPF}}{\boxed{H_{LP}(e^{j\omega})}} \xrightarrow{u[n]} \boxed{\downarrow M} \rightarrow y[m] = u[Mm]$$

(Decimator)

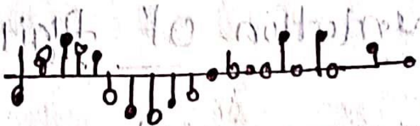
Upsampling:- Increase sampling rate of a signal.

This can be done inserting new samples between the existing samples.

$$x_u[n] = \begin{cases} x[n/L], & n = 0, \pm 1, \pm 2, \dots \\ 0 & ; \text{ otherwise} \end{cases}$$

$x[n] \rightarrow \boxed{\uparrow L} \rightarrow x_u[n]$

Symbol upsampler



$(L = 2)$

Advantage of Multirate DSP:-

(1) Reduce Computational Complexity:

By using algorithm. Algorithms can be ~~used~~ implemented using fewer operations when they are applied to signals that have been sampled at multiple rates.

(2) **Improved Accuracy:-** Multirate DSP can be used to improve accuracy. The samples that have been sampled at multiple rate can be design to exploit. ...

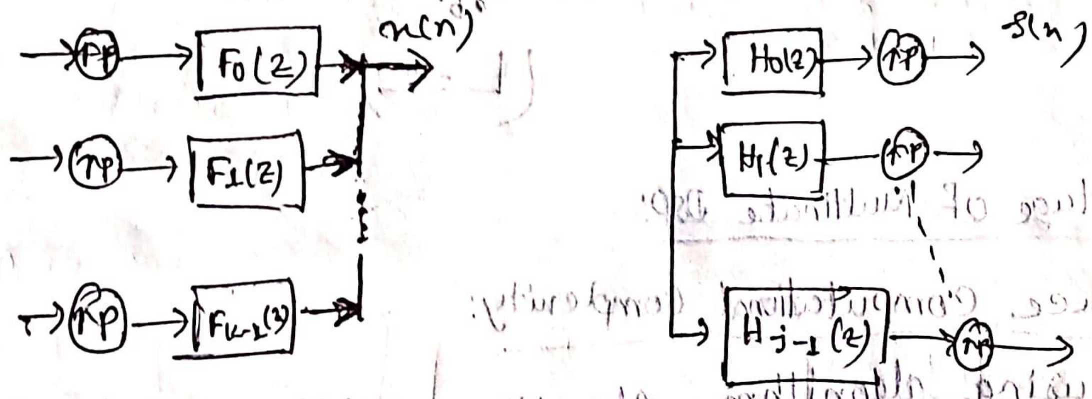
(3) **Increased Flexibility:-** Algorithms can be implemented in a variety of ways, depending on the specific requirements of the application.

(4) **Application of Multirate DSP:-**

→ used to Design for phase shifters.

→ Interfacing of digital systems with different sampling rates.

→ Implementation of Digital filter bank:-



(a) Synthesis Filter Bank

(b) Analysis Filter Bank

(4) Subband Coding of Speech Signals

A technique used to compress speech signals.

The speech signal is divided into a number of frequency bands, and each band is encoded separately. By using quantization, Differential coding this can be done.

(5) Quadrature Mirror Filters:

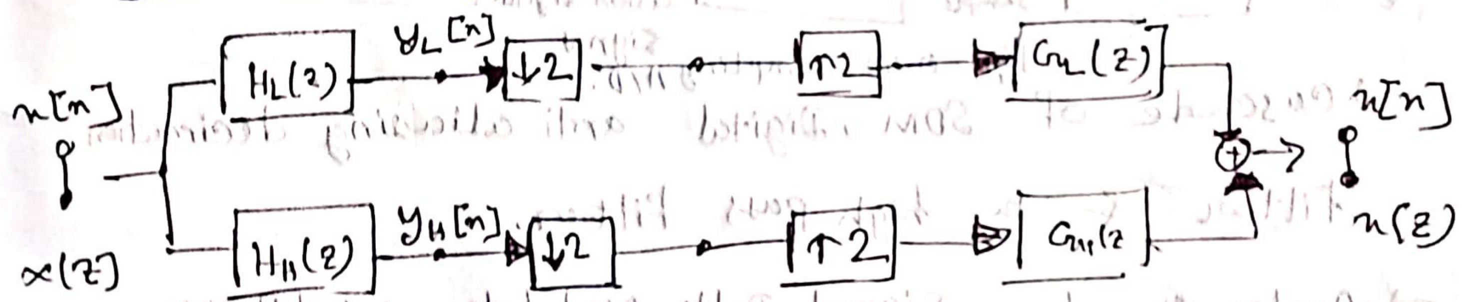


Fig: Two Channel QMF Bank

=) split into two output with bandwidth half of the original bandwidth. The output QMF bank after being processed (encoding, decoding, individual amplification etc) is recombined to a single signal using the synthesis filter bank also composed of QMFs.

(6) Transmultiplexer:-

Devices used for Converting Time Division Multiplexed (TDM) signal & Frequency Division Multiplexed (FDM)

(7) Oversampling A/D & D/A conversion:-

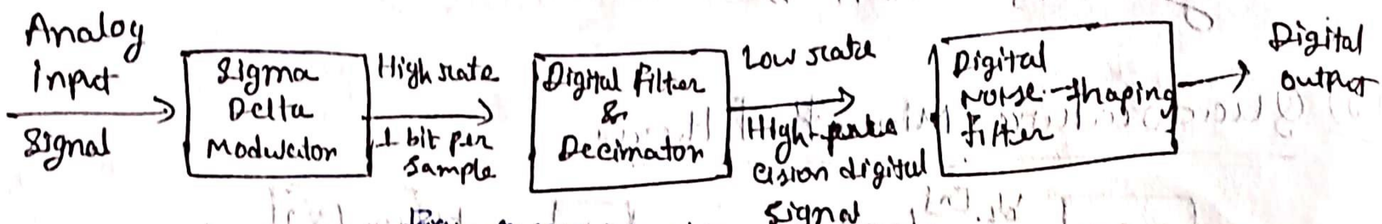


Fig:- over-sampling A/D.

⇒ Cascade of SDM, Digital anti aliasing decimation filter & a high pass filter.

⇒ Analog signal → Sigmal Delta Modulator → 1 bit per sample output at a very high sampling rate → Digital Low pass filter → provides high precision output → Lower sampling rate → DNSF to attenuate the quantization Noise at lower frequencies.

D/A Conversion:-

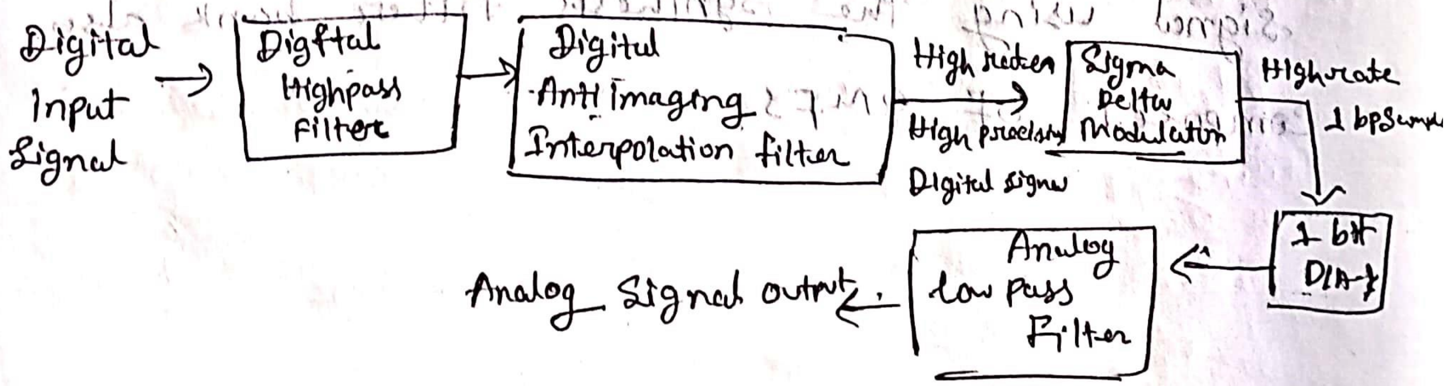


Fig: D/A Conversion.

→ Digital signal passed → Highpass Filter

→ output Fed to Digital interpolator

→ High sampling rate signal is the input of SDM.

→ SDM provides high sampling rate.

→ One bit per sample output.

→ Analog Low pass Filter makes low Frequency.

→ Analog Filters.