

Quiz 1 - 10
Quiz 2 - 10
Assn - 20
Mid - 25
Final - 35

Theory = 100

Submission - 15
Mid Exam - 10
Project/End Exam - 15

Lab = 40

Textbooks:

Signals & Systems by Oppenheim
DSP, JG Proakis

Addn: Principles of SP, BP Lathi

DSP : Sanjit K Mitra

Signal Processing

Prof. Santosh Nannuru

A > 130

F < 60

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Labs:

MATLAB Based

Github Submission

Tutorial: 10:30 - 11:30 Saturday

Signal Processing

→ Overview of NeSS :-

- Time variant - function of time
- Signals can be a function of any variable, need not be time always.
- Multi-Dimensional signals - images and videos.
- Anything that carries information is a signal.
- The course focuses more on discrete time signals, since most digital devices work in a discrete manner.
- System process signals, $x(t) \rightarrow \text{System}(H) \rightarrow y(t)$.
- LTI system - Linear and Time-Invariant Systems.
- Any circuit with only R,L,C will always form an LTI system, since R,L,C are all linear components.

Diodes and Transistors form non-linear systems.

- In any LTI system, the output $y(t)$ of a signal $x(t)$ is,

$$y(t) = \int_{-\infty}^{\infty} x(\tau) h(t-\tau) d\tau \rightarrow \underline{\text{Convolution}}$$

$h(t)$ = impulse response of the system.

- Laplace Transform :-

Representation of the signal in the s / complex frequency domain.

- Poles of the system function determine the behaviour / properties of the system.
- ROC - Region in the complex s - plane where the Laplace transform converges. Is always a vertical strip, defined by the location of poles.

- Fourier Series :-

- Any periodic signal can be represented as a sum of sines and cosines.

→ Course Topics:-

- Fourier Analysis, Z transform
- Sampling and Quantization - DSP
- Computational Aspect - fast Fourier Transform
- Manipulating Signals - filter design.

→ Fourier Analysis:-

- Periodic and Continuous-time - Fourier Series Rep.
- Aperiodic and Continuous-time - Fourier Transform
- Aperiodic and Discrete-time - Discrete Time Fourier Transform
- Finite Length and Discrete-time - DFT
 - Faster
→ FFT
- Fourier Series Representation :
 - Representation of any periodic signal as a sum of sinusoidal harmonics.

$$x(t) = b_0 + \sum_{k \in \mathbb{N}} (a_k \sin(k\omega_0 t) + b_k \cos(k\omega_0 t))$$

ω_0 - Angular frequency of the signal. ($2\pi/\tau$)

$$x(t) = \sum_{k \in \mathbb{Z}} c_k e^{jk\omega_0 t}$$

- Analysis: Computing the coefficients of the given signal

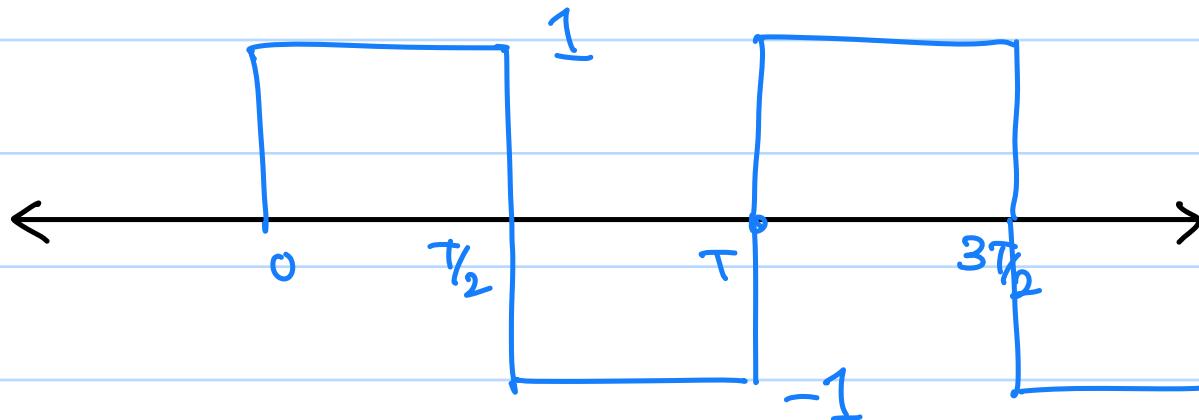
Synthesis: Given the signal, computing the coefficients.

$$a_k = \frac{2}{T} \int_T x(t) \sin(k\omega_0 t) dt, \quad k \neq 0$$

$$b_k = \frac{2}{T} \int_T x(t) \cos(k\omega_0 t) dt$$

$$c_k = \frac{1}{T} \int_T x(t) e^{-jk\omega_0 t} dt$$

Compute the FS Coeffs for a Sq. wave.



$$x(t) = \begin{cases} 1, & t \in (0, T/2) \\ -1, & t \in (T/2, T) \end{cases}$$

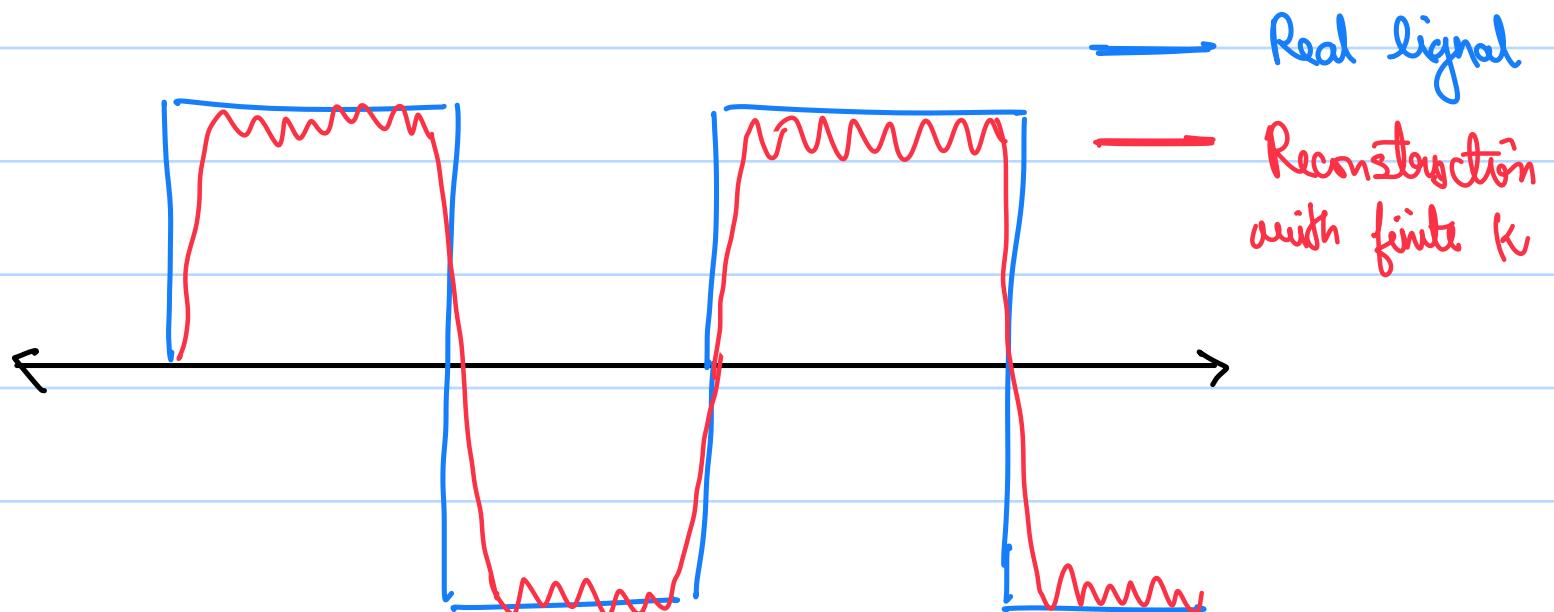
$$\begin{aligned}
c_k &= \frac{1}{T} \int_0^T x(t) e^{-jk\omega_0 t} dt \\
&= \frac{1}{T} \left[\int_0^{T/2} e^{-jk\omega_0 t} dt - \int_{T/2}^T e^{-jk\omega_0 t} dt \right] \\
&= \frac{1}{T} \left(\frac{-1}{j\omega_0} \right) \left[[e^{-jk\omega_0 t}]_{0}^{T/2} - [e^{-jk\omega_0 t}]_{T/2}^T \right] \\
&= -\frac{1}{k\pi \frac{2\pi}{T}} \left(\left[e^{-jk(\frac{2\pi}{T})(\frac{T}{2})} - e^{-jk\omega_0(0)} \right] - \left[e^{-jk(\frac{2\pi}{T})T} - e^{-jk(\frac{2\pi}{T})(\frac{T}{2})} \right] \right) \\
&= -\frac{1}{k2\pi} \left[\left(e^{-jk\pi} - 1 \right) - \left(e^{-jk2\pi} - e^{-jk\pi} \right) \right] \\
&= -\frac{1}{k2\pi} (2e^{-jk\pi} - 1 - e^{-jk2\pi}) \\
&= -\frac{1}{k2\pi} (2e^{-jk\pi} - 2) \\
&= -\frac{1}{k\pi} (e^{-jk\pi} - 1) \xrightarrow{\text{for even } k} 0 \\
&= \frac{1 - \cos k\pi}{k\pi} \quad \text{if odd } k
\end{aligned}$$

• Partial Reconstruction :-

- Not all the coefficients are known to us, so only an approximation of the signal is constructed.

- Reconstruction Error $e(t) = x(t) - \hat{x}(t)$

- Discontinuous signals will have an infinite number of non-zero coefficients.



FS Coeffs of Sq. Wave Continuation ,

$$a_k = \frac{2}{2T} \int n(t) \sin(k\omega_0 t) dt$$

$$= \frac{1}{T} \left[\int_0^T -\sin(k\omega_0 t) dt + \int_T^{2T} \sin(k\omega_0 t) dt \right]$$

$$= \frac{1}{T} \left(\frac{1}{k\omega_0} \right) \left[[\cos(k\omega_0 t)]_0^T + [-\cos(k\omega_0 t)]_T^{2T} \right]$$

$$= \frac{1}{k\pi} \left[\left(\cos \left(k \frac{2\pi}{2T} (\pi) \right) - 1 \right) + \left(\cos \left(k \left(\frac{2\pi}{2T} \right) T \right) - \cos \left(k \left(\frac{2\pi}{2T} \right) 2T \right) \right) \right]$$

$$= \frac{1}{k\pi} \left[(\cos k\pi - 1) + (\cos k\pi - 1) \right]$$

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

$$b_k = \frac{2}{2T} \int_{-T}^T x(t) \cos(k\omega_0 t) dt$$

$$= \frac{1}{T} \left[\int_0^T -\cos(k\omega_0 t) dt + \int_T^{2T} \cos(k\omega_0 t) dt \right] \left(\frac{1}{k\omega_0} \right)$$

$$= \frac{1}{T} \left[(-\sin(k\omega_0 t)) \Big|_0^T + (\sin(k\omega_0 t)) \Big|_T^{2T} \right] \left(\frac{1}{k\omega_0} \right)$$

$$= \frac{1}{T} \left[(0 - \sin(k\pi)) + (\sin k \frac{2\pi}{2T} - \sin k \frac{\pi}{2T}) \right]$$

$$= 0 \neq k$$

- Note: If $a_k \in \mathbb{R} \neq k$, then the signal is Even.

→ Discrete-Time signals :-

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- Represented by $x[n]$, where $n \in \mathbb{Z}$, n is an index where time duration is the sampling interval.

- Unit Impulse:

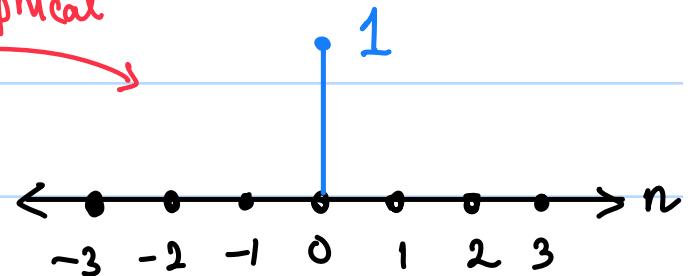
- In continuous time,

$$s(t) = 0 \neq t \neq 0, \text{ s.t. } \int_{-\infty}^{\infty} s(t) dt = 1$$

- In discrete time,

$$s[n] = \begin{cases} 1, & n = 0 \\ 0, & n \neq 0 \end{cases}$$

Graphical

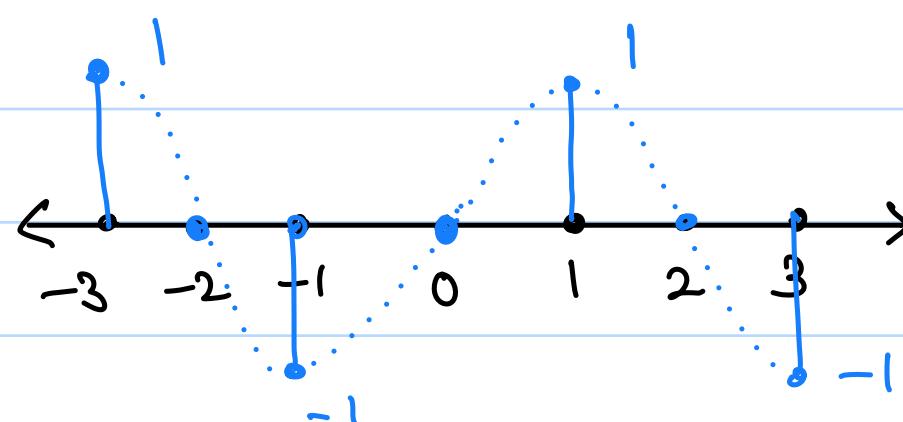


- Unit Step: $u[n] = \begin{cases} 1, & n \geq 0 \\ 0, & n < 0 \end{cases}$

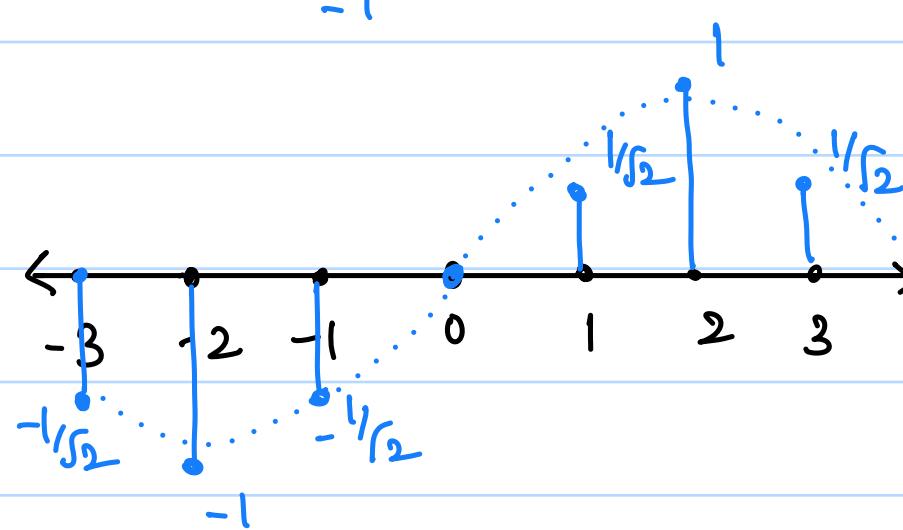
- Exponential signal: $x[n] = a^n, a \in \mathbb{R}$

Example: Plot $x[n] = \sin(\omega_0 n)$ for $\omega_0 = \frac{\pi}{2}, \frac{\pi}{4}$

1)



2)



- A signal that is periodic in continuous time may not be always be periodic in discrete time, due to the sampling frequency. ($x[n] = \sin(n)$)

- Discrete Time Periodic Signals:-

$x[n]$ is periodic if,

$$x[n+P] = x[n] \text{ for some } P \in \mathbb{Z}$$

- Complex Exponentials :-

$$x[n] = e^{sn}, s \in \mathbb{C}, \text{ ie. } s = \sigma + j\omega, \sigma, \omega \in \mathbb{R}$$

$$\Rightarrow x[n] = e^{\sigma n} (\cos(\omega n) + j \sin(\omega n))$$

Note: Sinusoids are periodic if $\omega_0 = \frac{2\pi}{P}$ for some $P \in \mathbb{Z}$

- Even and Odd:

$$\text{Even : } x[n] = x[-n] \quad \forall n \in \mathbb{Z}$$

$$\text{Odd : } x[n] = -x[-n] \quad \forall n \in \mathbb{Z}$$

- Energy of a Discrete Time signal :-

$$E = \sum_{n \in \mathbb{Z}} |x[n]|^2$$

It may not relate to any physical quantity (property of the signal).

$$\text{Power} = \lim_{N \rightarrow \infty} \frac{1}{2N+1} \sum_{n=-N}^N |x[n]|^2$$

Energy signals : signals that carry finite energy, and hence, zero power

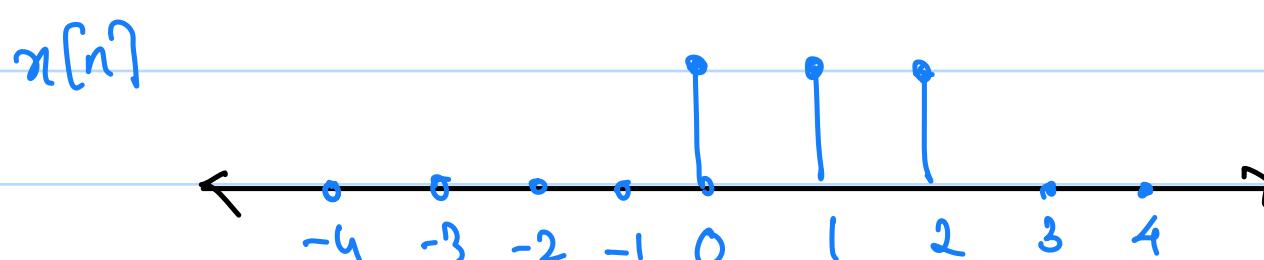
Power signals : signals that carry finite power.

→ Time axis Manipulation for Discrete time signals :-

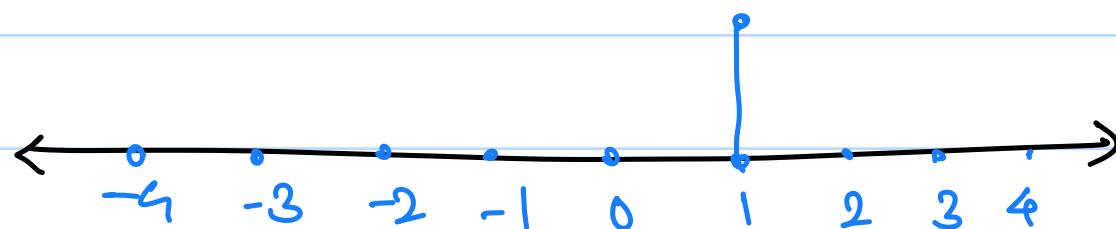
- $x[n - n_0]$ - Shift right by n_0
- $x[n + n_0]$ - Shift left by n_0
- $x[kn]$ - Only every k^{th} sample is taken

Example: $x[n] = u[n] - u[n - 3]$

$$y[n] = x[2n - 1] = ?$$



$$x[2n - 1]$$



- Also, due to the nature of discrete signals, $\omega_0 = \pi$ is the highest possible angular frequency for a sinusoid.

Beyond $\omega_0 = \pi$, the signal will be the same as that of a lower frequency, due to the properties of sinusoids.

$$\cos(\pi n) = (-1)^n$$

Oscillation b/w 1 and -1
at the freq of sampling

$$\begin{aligned}
 \sin((\pi + \phi)n) &= \sin(\pi n + \phi n) \\
 &= \sin \pi n \cos \phi n + \cos \pi n \sin \phi n \\
 &= (-1)^n \sin \phi n \quad \rightarrow \text{freq} < \pi
 \end{aligned}$$

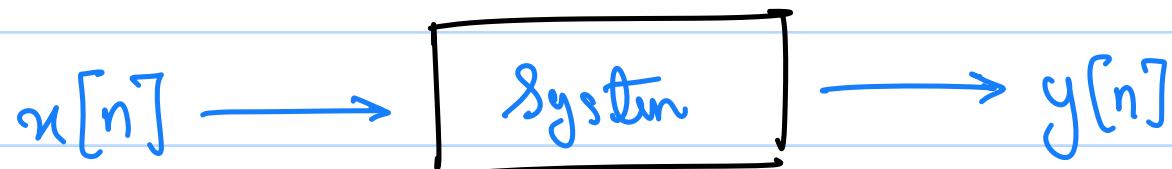
$$\begin{aligned}
 \cos((\pi + \phi)n) &= \cos(\pi n) \cos(\phi n) - \sin(\pi n) \sin(\phi n) \\
 &= (-1)^n \cos \phi n \quad \rightarrow \text{freq} < \pi
 \end{aligned}$$

- Signal Representation By Unit Impulses :-

$$x[n] = \sum_{m \in \mathbb{Z}} x[m] \delta[n-m] \quad \forall n \in \mathbb{Z}$$

Since $\delta[n-m] = 0 \quad \forall n \neq m$, all the other values of the signal are nullified at each n .

- System :-



- Delay :

$$y[n] = x[n-\Delta], \quad \Delta \in \mathbb{Z}$$

- Amplitude Scaling :

$$y[n] = \alpha x[n], \quad \alpha \in \mathbb{C}$$

- Integrator (Summation) :

$$y[n] = \sum_{m=-\infty}^n x[m]$$

o Difference:

$$y[n] = x[n] - x[n-1]$$

Q. $x(t) = \cos(4\pi t)$ is sampled with sampling frequency below.

Identify the frequency of the obtained DT signal.

$$f_s: 1\text{Hz}, 2\text{Hz}, 4\text{Hz}, 8\text{Hz}$$

$$x(t) = \cos(4\pi t)$$

In Sampling, $t \rightarrow nT$, $T = \frac{1}{f_s}$

$$\Rightarrow x[n] = \cos\left(\frac{4\pi n}{f_s}\right), \quad \text{freq} f = \frac{4\pi}{f_s}$$

$$\Rightarrow f = \frac{2}{f_s}$$

→ Properties of System :-

o Linearity: If x_1 gives o/p y_1 , x_2 gives o/p y_2 then,

$$\alpha x_1 + \beta x_2 \longrightarrow \alpha y_1 + \beta y_2. \quad \alpha, \beta \in \mathbb{C} \text{ constants}$$

o Time Invariant: If $x[n]$ gives o/p $y[n]$, then,

$$x[n-n_0] \longrightarrow y[n-n_0] \quad \forall n_0 \in \mathbb{Z}$$

Also termed as shift invariant.

Example: Analyze time invariance of

1) $y[n] = x[n - \Delta]$, $\Delta \in \mathbb{Z}$

2) $y[n] = x^2[n]$

3) $y[n] = x[n] + x[n - 1]$

4) $y[n] = n x[n]$

1) $y[n] = x[n - \Delta] \rightarrow y[n - n_0] = x[n - n_0 - \Delta]$

$$x[n - n_0] \xrightarrow{\text{Sys}} x[n - n_0 - \Delta] \quad \therefore \text{Time Invariant}$$

2) $y[n] = x^2[n] \rightarrow y[n - n_0] = (x[n - n_0])^2$

$$y_2[n] = (x[n - n_0])^2 \quad \therefore \text{Time Invariant}$$

3) $y[n] = x[n] + x[n - 1] \rightarrow y[n - n_0] = x[n - n_0] + x[n - n_0 - 1]$

$$y_2[n] = x[n - n_0] + x[n - n_0 - 1] \quad \therefore \text{Time Invariant}$$

4) $y[n] = n x[n] \rightarrow y[n - n_0] = (n - n_0)x[n - n_0]$

$$y_2[n] = n x[n - n_0] \quad \therefore \text{Time Variant}$$

• Causality: A system is said to be causal if the output $y[n] \forall n \in \mathbb{Z}$ should only depend on $x[m]$ for $m \leq n$.

- Stability: If $x[n] \leq B + n \in \mathbb{Z}$ for some $B \in \mathbb{C}$, then $y[n] \leq B_0 + n \in \mathbb{Z}$, for some $B_0 \in \mathbb{C}$. Bounded Input, Bounded Output.

→ Linear - Time Systems:-

- LTI Systems are linear and time invariant.
- Impulse Response: Response of the system to the impulse signal $\delta[n]$. Denoted by $h[n]$.
- Step Response: Response of the system to the step signal $v[n]$. Denoted by $s[n]$

$$\alpha \delta[n - n_0] \xrightarrow{H} \alpha h[n - n_0]$$

- Since we can denote any signal in the form,

$$x[n] = \sum_m x[m] \delta[n - m] \quad \text{scaled and delayed impulse.}$$

$$\Rightarrow y[n] = \sum_m x[m] h[n - m]$$

Applying the above 2 results.

$$\Rightarrow y[n] = x[n] * h[n] \quad x[m] \text{ is a constant w.r.t } n.$$

Properties of Convolution :

III to multiplication

1) Commutativity : $x[n] * y[n] = y[n] * x[n]$

Proved using
change of variable

2) Distributive of Addition : $x[n] * (y[n] + z[n]) = x[n] * y[n] + x[n] * z[n]$

3) Associative : $(x[n] * y[n]) * z[n] = x[n] * (y[n] * z[n])$

- Properties of $\delta[n]$ in Convolution :

1) $x[n] * \delta[n] = x[n]$

2) $x[n] * \delta[n-\Delta] = x[n-\Delta] \quad \forall \Delta \in \mathbb{Z}$

3) $x[n] * S[n-\Delta] = \underbrace{x[\Delta] \delta[n-\Delta]}_{\text{Scaled impulse}} + \Delta \in \mathbb{Z}$

Description of a System :-

1) Working description : Describe the working of the system.

2) Mathematical description : Describe the mathematical function relating input and output.

3) Impulse Response (LTI) : Describe the impulse response of the LTI system. Output can be calculated through convolution.

o Properties of a system through Impulse Response :-

- Causal Systems: $h[n] = 0 \quad \forall n \in \mathbb{Z}^-$ for a Causal system.

$$\begin{aligned} y[n] &= x[n] * h[n] \\ &= \sum_m x[m] h[n-m] \\ &= \sum_{m=-\infty}^n x[m] h[n-m] + \sum_{m=n+1}^{\infty} x[m] h[n-m] \end{aligned}$$

should be zero

$$\Rightarrow h[n-m] = 0 \quad \forall m \geq n+1$$

$$m-n \geq 1$$

$$n-m \leq -1 \Rightarrow n-m < 0$$

$$\Rightarrow h[r] = 0 \quad \forall r < 0$$

- Stable Systems: $\sum_n |h[n]|$ should be bounded for stability

$$|y[n]| = \left| \sum_m x[m] h[n-m] \right|$$

$$\Rightarrow |y[n]| \leq \sum_m |x[m]| |h[n-m]| \quad \Delta \text{le inequality}$$

$$\Rightarrow |y[n]| \leq \sum_m B |h[n-m]| = B \sum_k |h[k]|$$

\therefore If $\sum |h[k]|$ is bounded, $|y[n]|$ becomes bounded. (Sufficient Condition)

Note:

If $A \Rightarrow B$, then A is sufficient for B

If $A \Leftarrow B$, then A is Necessary for B

$A \Leftrightarrow B$, A is Necessary and sufficient (iff)

Example: Derive the impulse response of $y[n] = x[n] + \alpha y[n-1]$

$$y[n] = x[n] + \alpha y[n-1]$$

$$= x[n] + \alpha(x[n-1] + \alpha y[n-2])$$

$$= x[n] + \alpha(x[n-1] + \alpha(x[n-2] + \dots))$$

$$= x[n] + \alpha x[n-1] + \alpha^2 x[n-2] + \alpha^3 x[n-3] + \dots$$

$$= \sum_{i=0}^n \alpha^i x[n-i]$$

[Assuming initial rest, i.e
 $y[n] = 0 \forall n < 0$]

$$h[n] = \sum_{i=0}^n \alpha^i \delta[n-i]$$

$$h[n] = \alpha^n$$

→ Linear Constant Coefficient Difference Equations :- (LCCDE)

• First order difference equation : $y[n] = x[n] - x[n-1]$

• Generic Difference Eqn : $y[n] = \sum_{p=0}^n a_p x[n-p] + \sum_{q=1}^{n_2} b_q y[n-q]$

If $x[n] = e^{j\omega n}$ is the input signal for an LTI system H ,

$$y[n] = h[n] * e^{j\omega n}$$

$$= \sum_{m=-\infty}^{\infty} h[m] e^{j\omega(n-m)}$$

$$= e^{j\omega n} \underbrace{\sum_m h[m] e^{-j\omega m}}_C$$

$$y[n] = C e^{j\omega n} = \underline{Cx[n]}$$

For any LTI System H , $e^{j\omega n}$ is an Eigensignal of H . The output waveform will be a scaled and phase-shifted input waveform.

→ Discrete Time Fourier Transform :-

The Discrete Time Fourier Transform of a discrete signal $x[n]$ is defined as,

$$X(\omega) = \sum_{-\infty}^{\infty} x[n] e^{-j\omega n}$$

→ Analysis Equation

Applying this in the Eigensignal concept,

$$y[n] = e^{j\omega n} \sum_{m=-\infty}^{\infty} h[m] e^{-j\omega_0 m}$$

$$\Rightarrow y[n] = e^{j\omega_0 n} H(\omega_0)$$

$H(\omega) \rightarrow$ FT of $h[n]$,
ie, system function

- For a discrete signal,

$$X(\omega + 2\pi) = X(\omega)$$

$X(\omega)$ is continuous in ω .

- Inverse DTFT :-

$$x[n] \xleftarrow{\text{DTFT}} X(\omega)$$

- We only need the values of $X(\omega)$ for $\omega \in [-\pi, \pi]$ or $[0, 2\pi]$, due to its periodicity.
- Inverse DTFT is done as follows,

$$\begin{aligned} X(\omega) e^{j\omega m} &= \sum_{n=-\infty}^{\infty} x[n] e^{-j\omega n} e^{j\omega m} \\ &= \int_0^{2\pi} X(\omega) e^{j\omega m} d\omega = \int_0^{2\pi} \sum_{n=-\infty}^{\infty} x[n] e^{j\omega(m-n)} d\omega \\ &= \sum_{n=-\infty}^{\infty} x[n] \int_0^{2\pi} e^{j\omega(m-n)} d\omega \end{aligned}$$

$$\Rightarrow x[m] = \frac{1}{2\pi} \int_{-\pi}^{\pi} X(\omega) e^{j\omega m} d\omega \rightarrow \text{Synthesis Equation}$$

Example: Compute DTFT of

$$i) x[n] = a^n u[n] \quad (|a| < 1)$$

$$X(\omega) = \sum x[n] e^{-j\omega n}$$

$$= \sum_{n=0}^{\infty} a^n e^{-j\omega n} = \frac{1}{1 - ae^{-j\omega}} \quad \begin{bmatrix} \text{Sum of } \infty \\ \text{GP} \end{bmatrix}$$

$$2) x[n] = \delta[n]$$

$$\begin{aligned} X(\omega) &= \sum_{n=-\infty}^{\infty} \delta[n] e^{-j\omega n} \\ &= e^{-j\omega(0)} = 1 + \omega \end{aligned}$$

$$3) x[n] = \begin{cases} 1, & -M \leq n \leq M \\ 0, & \text{otherwise} \end{cases}$$

$$\begin{aligned} X(\omega) &= \sum_{n=-\infty}^{\infty} x[n] e^{-j\omega n} = \sum_{n=-M}^{M} e^{-j\omega n} \\ &= \frac{e^{j\omega M} (1 - e^{-j\omega 2M})}{1 - e^{-j\omega}} \end{aligned}$$

$$X(\omega) = \frac{e^{j\omega M} - e^{-j\omega M}}{1 - e^{-j\omega}}$$

• If $X(\omega)$ is not periodic with period 2π , it cannot be a valid DTFT of a discrete signal.

Note: $\int_{-\infty}^{\infty} S(t) x(t) dt = n(0)$

Note 2: If one waveform in time domain gives another waveform in Fourier domain, the same waveform in Fourier domain gives the latter

waveform in time domain. , ie, the shape of waveform can be switched from time to Fourier and vice versa.

→ Properties of DTFT:-

i) Linearity :

$$x_1[n] \xleftrightarrow{\text{DTFT}} X_1(\omega)$$

$$x_2[n] \xleftrightarrow{\text{DTFT}} X_2(\omega)$$

$$\Rightarrow \alpha x_1[n] + \beta x_2[n] \xleftrightarrow{\text{DTFT}} \alpha X_1(\omega) + \beta X_2(\omega)$$

Can be proved by the linear nature of Integration .

2) Time Shift :-

$$x[n] \xleftrightarrow{\text{DTFT}} X(\omega)$$

$$\Rightarrow x[n-n_0] \xleftrightarrow{\text{DTFT}} X(\omega)e^{-j\omega n_0}$$

Proof :

$$X'(w) = \sum_n x[n-n_0] e^{-j\omega n}$$

$$\text{Take } n - n_0 = m$$

$$\Rightarrow X'(w) = \sum_m x[m] e^{-j\omega(m+n_0)}$$

$$= \sum_m x[m] e^{-j\omega m} \cdot e^{-j\omega n_0}$$

$$\Rightarrow \underline{X'(w) = X(w) e^{-j\omega n_0}}$$

Since $|e^{-j\omega n_0}| = 1$, the magnitude spectrum is unaffected . Only the

phase spectrum changes.

3) Frequency Shift :-

$$X(\omega) \xleftarrow{\text{DTFT}} x[n] .$$

$$\Rightarrow X(\omega - \omega_0) \xleftarrow{\text{DTFT}} x[n] e^{-j\omega_0 n}$$

4) Symmetry:

$$x[n] \xleftarrow{\text{DTFT}} X(\omega)$$

meal

meal and every

read and odd

$$X(\omega) = X^*(-\omega)$$

real and even

imag. and odd.

Example: Is frequency shifting an LTI system?

$$Y(\omega) = X(\omega - \omega_0)$$

o/p delay :

$$y[n-n_0] = x[n-n_0] e^{j\omega_0(n-n_0)}$$

ilp delay:

$$y_2[n] = x[n - n_0] e^{j\omega_0 n}$$

$$\Rightarrow y[n - n_0] \neq y_2[n]$$

System is not time-invariant.

5) Differentiation in Frequency :-

$$x[n] \xleftrightarrow{\text{DTFT}} X(\omega)$$

$$-jnx[n] \xleftrightarrow{\text{DTFT}} \frac{d}{d\omega} X(\omega)$$

$$X(\omega) = \sum_n x[n] e^{-j\omega n}$$

$$\frac{d}{d\omega} X(\omega) = \sum_n \frac{d}{d\omega} x[n] e^{-j\omega n} = \sum_n x[n] \frac{d}{d\omega} e^{-j\omega n}$$

$$= \sum_n x[n] (-jn) e^{-j\omega n}$$

$$= \sum_n (-jnx[n]) e^{-j\omega n}$$

6) Convolution :-

$$x[n] \xleftrightarrow{\text{DTFT}} X(\omega) \quad y[n] \xleftrightarrow{\text{DTFT}} Y(\omega)$$

$$x[n]*y[n] \xleftrightarrow{\text{DTFT}} X(\omega) \cdot Y(\omega)$$

$$F(x[n]*y[n]) = \sum_n \left(\sum_m x[m] y[n-m] \right) e^{-j\omega n}$$

$$= \sum_n \left(\sum_m x[m] y[n-m] e^{-j\omega(m+n-m)} \right)$$

$$= \sum_n \left(\sum_m x[m] y[n-m] e^{-j\omega m} e^{-j\omega(n-m)} \right)$$

$$= \sum_n \left(\sum_m x[m] e^{-j\omega m} y[n-m] e^{-j\omega(n-m)} \right)$$

$$\begin{aligned}
 &= \sum_m x[m] e^{-j\omega m} \left(\sum_n y(n-m) e^{-j\omega(n-m)} \right) \\
 &= \left(\sum_m x[m] e^{-j\omega m} \right) Y(\omega) \\
 &= \underline{x(\omega) \cdot Y(\omega)}
 \end{aligned}$$

7) Parseval's Relation :-

Energy is conserved across time and frequency domain, ie,

$$\sum_n |x[n]|^2 = \frac{1}{2\pi} \int_{-2\pi}^{2\pi} |X(\omega)|^2 d\omega$$

8) Multiplication in Time Domain :-

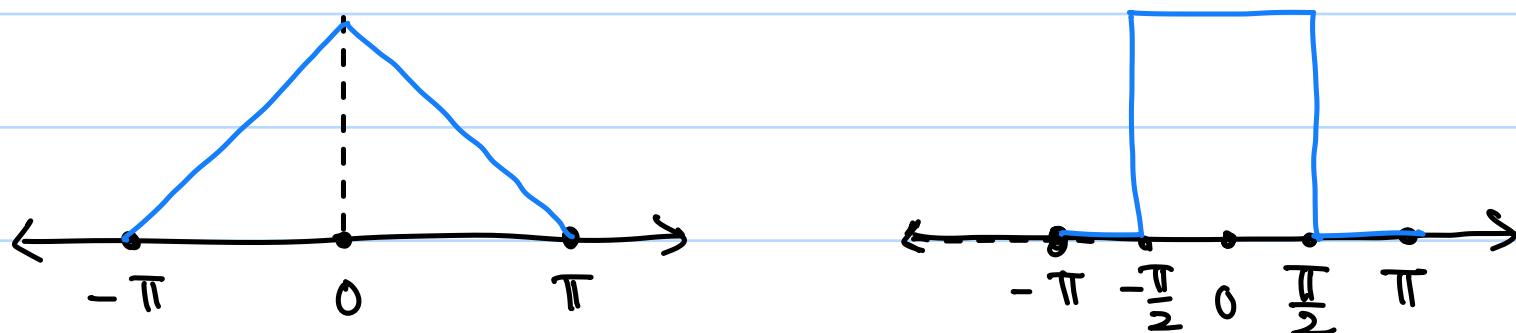
$$x[n] \xrightarrow{\text{DTFT}} X(\omega), \quad y[n] \xrightarrow{\text{DTFT}} Y(\omega)$$

$$x[n] \cdot y[n] \xrightarrow{\text{DTFT}} \frac{1}{2\pi} \int_{-2\pi}^{2\pi} X(\alpha)Y(\omega-\alpha) d\alpha$$

↳ Periodic Convolution

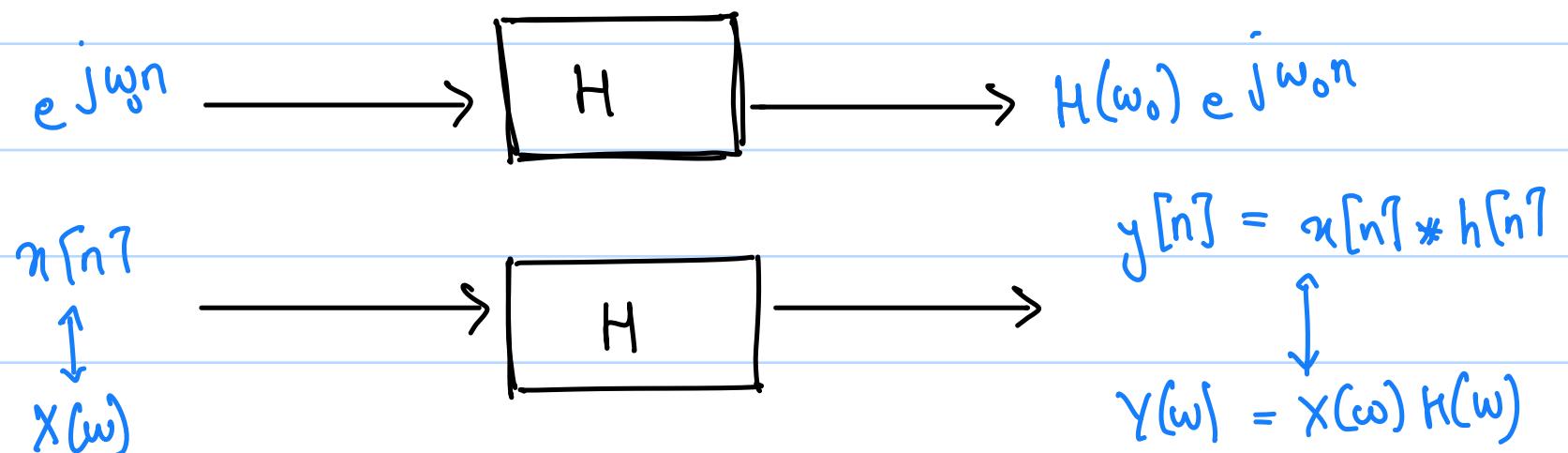
Example:

Perform periodic convolution of



→ Convolution and Frequency Selectivity :-

- We know that,



$$\therefore \text{For all LTI system } Y(w) = X(w) H(w)$$

Because of the above relation, $Y(w)$ cannot contain frequencies that $X(w)$ does not have, ie, $H(w)$ is like a gate/filter.

$$\text{If } X(\omega_0) = 0, \text{ then } Y(\omega_0) = X(\omega_0) H(\omega_0) = \underline{\underline{0}}.$$

$H(w)$ - Frequency Response

$|H(w)|$ - Magnitude Response

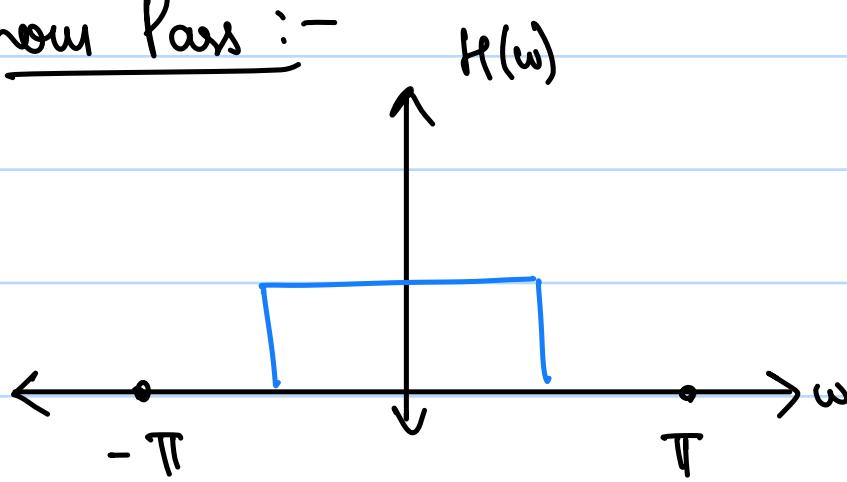
$\angle H(w)$ - Phase Response

The above mentioned multiplication consequence is termed as **frequency selectivity**.

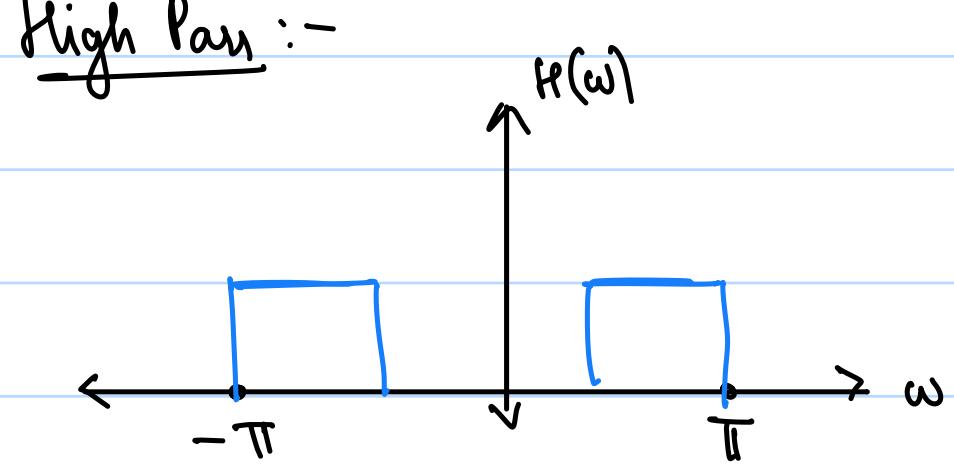
- The behaviour of $H(w)$ can be modified to make filters.
 - 1) Low Pass & High Pass Filter
 - 2) Band stop & Band pass Filter

3) Notch & All pass filter.

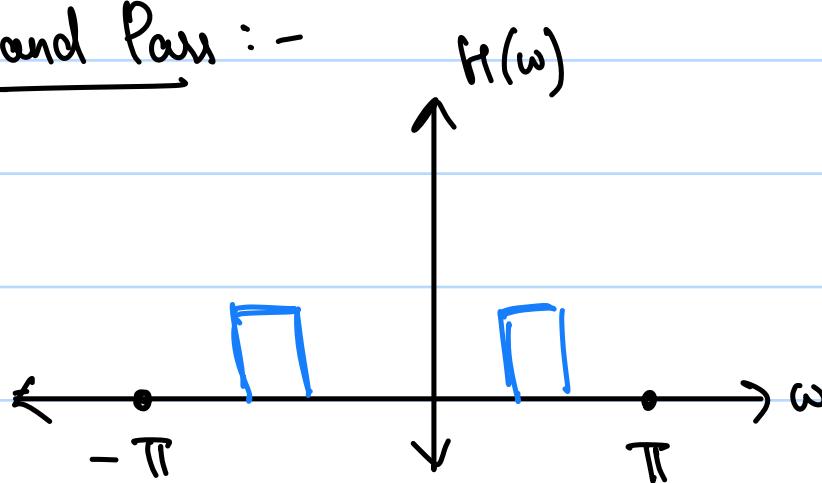
Low Pass :-



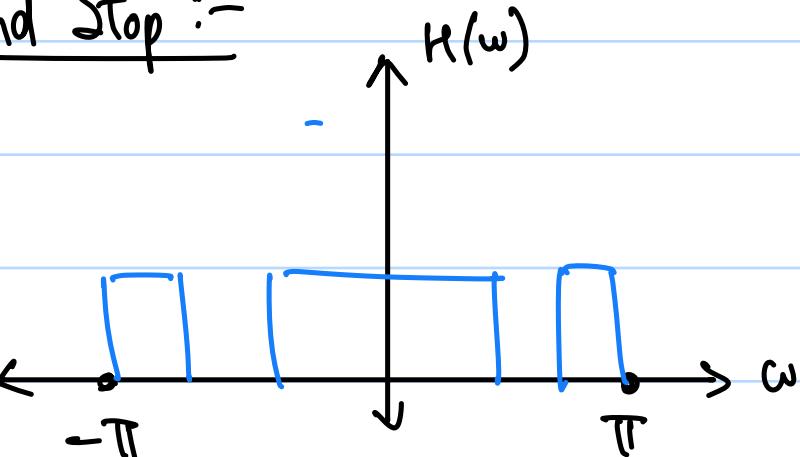
High Pass :-



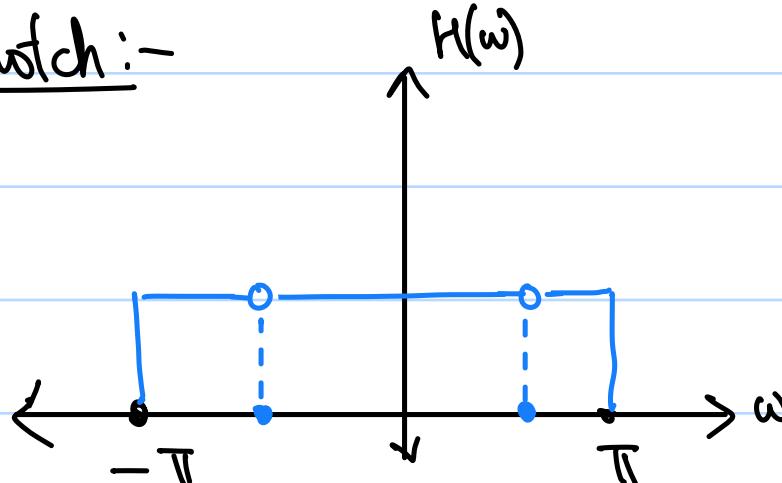
Band Pass :-



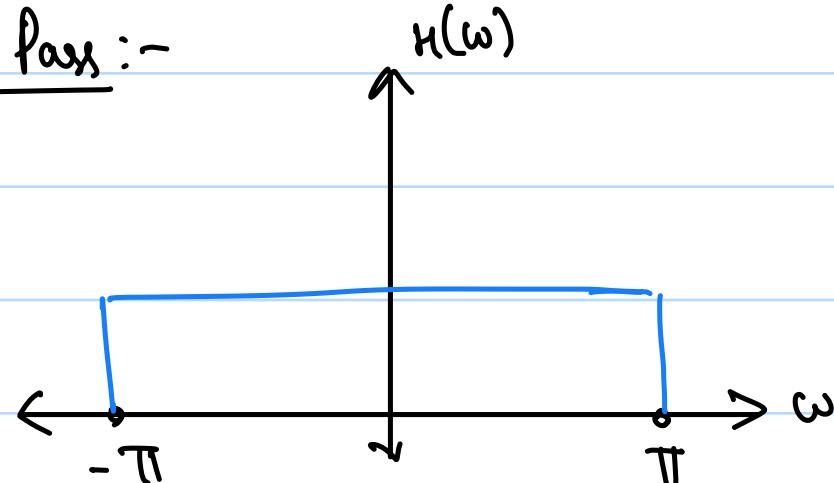
Band Stop :-



Notch :-



All Pass :-



$H(\omega)$ is symmetric in all the filters since $h[n]$ is a real valued impulse response.

All-pass filters are used in delay systems, since phase can be non-zero.

• Constraints for implementing Ideal filters :-

- 1) They are not BIBO stable. (Bounded input \neq Bounded output)
- 2) They have infinite impulse responses.
- 3) They are not causal.

Example: Identify the filter type of the following LTI system.

$$a) y[n] = \frac{1}{2}(x[n] + x[n-1])$$

$$\Rightarrow Y(s) = \frac{1}{2}(X(s) + X(s)e^{-j\omega}) \Rightarrow$$

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

$$= \frac{1}{2}(1 + \cos(\omega) - j\sin(\omega))$$

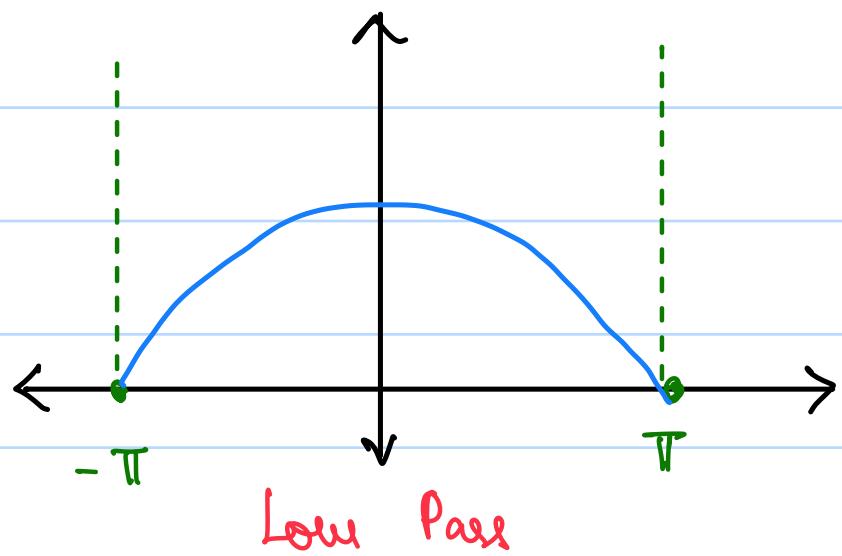
$$= \frac{1}{2} + \frac{1}{2}\cos(\omega) - \frac{1}{2}j\sin(\omega)$$

$$|H(s)| = \sqrt{\frac{1}{4}(1 + \cos\omega)^2 + \frac{1}{4}\sin^2\omega}$$

$$= \sqrt{\frac{1}{4} + \frac{1}{2}\cos\omega + \frac{1}{4}\cos^2\omega + \frac{1}{4}\sin^2\omega}$$

$$= \sqrt{\frac{1}{2} + \frac{1}{2}\cos\omega}$$

$$= \frac{1}{\sqrt{2}} \sqrt{1 + \cos(\omega)}$$



$$b) y[n] = \frac{1}{2}(x[n] - x[n-1])$$

$$\Rightarrow Y(s) = \frac{x(s)}{2} (1 - e^{-j\omega})$$

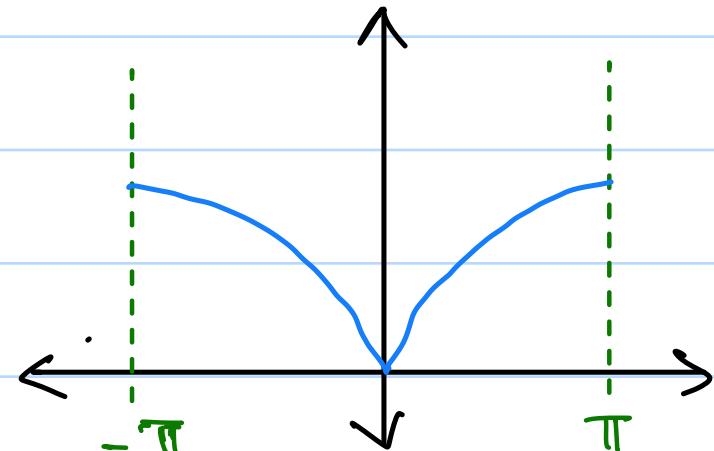
$$\Rightarrow H(s) = \frac{1 - e^{-j\omega}}{2}$$

$$|H(s)| = \frac{1}{2} \sqrt{(1 - e^{-j\omega})(1 - e^{j\omega})}$$

$$= \frac{1}{2} \sqrt{1 - e^{-j\omega} - e^{j\omega} + e^{-j\omega + j\omega}},$$

$$= \frac{1}{2} \sqrt{2 - (2\cos(\omega))}$$

$$= \frac{1}{\sqrt{2}} \sqrt{1 - \cos(\omega)}$$



$$c) h[n] = a^n u[n]$$

$$H(\omega) = \sum_n a^n u[n] e^{-j\omega n} = \sum_{n=0}^{\infty} a^n e^{-j\omega n}$$

$$= \frac{1}{1 - ae^{-j\omega}} \times \frac{(1 - ae^{j\omega})}{(1 - ae^{j\omega})}$$

$$= \frac{1 - ae^{j\omega}}{1 - ae^{j\omega} - ae^{-j\omega} + a} = \frac{1 - ae^{j\omega}}{1 + a - a(2\cos\omega)}$$

$$= \frac{1 - ae^{j\omega}}{1 + a - 2a\cos\omega}$$

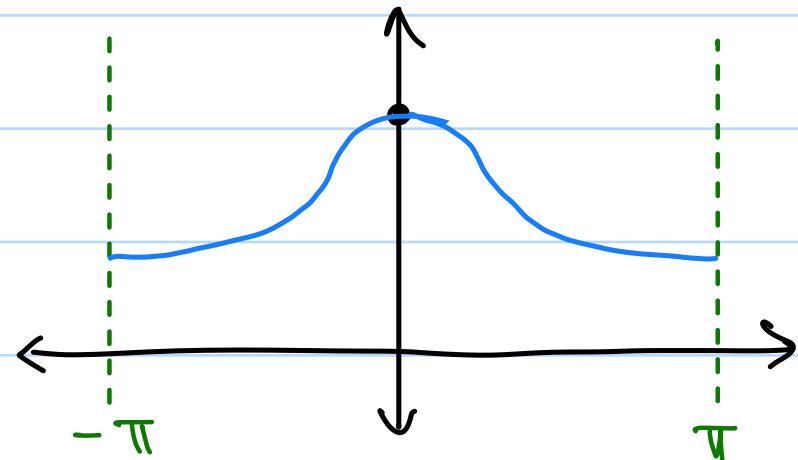
High Pass

$$|H(\omega)| = \frac{1}{1+a-2a\cos(\omega)} \sqrt{(1-a e^{j\omega})(1-a e^{-j\omega})}$$

$$= \frac{1}{1+a-2a\cos\omega} \sqrt{1+a-a e^{j\omega}-a e^{-j\omega}}$$

$$= \frac{1}{1+a-2a\cos\omega} \sqrt{1+a-2a\cos\omega}$$

$$= \frac{1}{\sqrt{1+a-2a\cos\omega}}$$



Low Pass

→ Continuous Time Fourier Transform :-

$$x(t) \xleftrightarrow{\text{CTFT}} X(\omega)$$

- An extension of FSR to non-periodic signals.

$$X(\omega) = \int_{-\infty}^{\infty} x(t) e^{-j\omega t} dt$$

$$x(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} X(\omega) e^{j\omega t} d\omega$$

$$e^{j\omega_0 t} \longrightarrow [LTI] \longrightarrow H(\omega_0) e^{j\omega_0 t}$$

i.e., $e^{j\omega_0 t}$ is an Eigen signal of continuous LTI system.

Important DTFT pairs :-

$$1) \quad a^n u[n] \xleftrightarrow{\text{DTFT}} \frac{1}{1-a e^{-j\omega}} \quad \underline{\# |a| < 1}$$

$$2) \quad -a^n u[-n-1] \xleftrightarrow{\text{DTFT}} \frac{1}{1-a e^{-j\omega}} \quad \underline{\# |a| > 1}$$

Important CTFDT pairs :-

$$1) \quad \delta(t) \xleftrightarrow{\text{CTFT}} 1$$

$$2) \quad \delta(t-b) \xleftrightarrow{\text{CTFT}} e^{-jb\omega}$$

$$3) \quad \begin{array}{c} 1 \\ \hline -T \qquad T \end{array} \xleftrightarrow{\text{CTFT}} \text{sinc}\left(\frac{2\pi}{T}\omega\right)$$

4) If $x(t)$ is periodic,

$$\text{wkt } x(t) = \sum_{k \in \mathbb{Z}} a_k e^{jk\omega_0 t} \quad [\omega_0 \text{ is the angular freq. of } x(t)]$$

$$\Rightarrow X(\omega) = \int_{-\infty}^{\infty} \sum_{k \in \mathbb{Z}} a_k e^{jk\omega_0 t} e^{-j\omega t} dt$$

$$= \sum_{k \in \mathbb{Z}} a_k \int_{-\infty}^{\infty} e^{jk\omega_0 t} e^{-j\omega t} dt$$

This is why FT
exists.

By orthogonality of sinusoids, $\rightarrow \left[\# \omega \neq k\omega_0, \int_{-\infty}^{\infty} e^{jk\omega_0 t} e^{-j\omega t} dt = 0 \right]$

$$\Rightarrow X(\omega) = 2\pi \sum_{k \in \mathbb{Z}} a_k \delta(\omega - k\omega_0) \rightarrow \text{Can also be directly proved by point ⑦}$$

$$5) \quad 1 \xrightleftharpoons{\text{CTFT}} 2\pi S(\omega)$$

$$6) \quad \sum_{k=-\infty}^{\infty} s(t-kT) \xrightleftharpoons{\text{CTFT}} \frac{2\pi}{T} \sum_{k=-\infty}^{\infty} s(\omega - k\frac{2\pi}{T})$$

$$7) \quad e^{j\omega_0 t} \xrightleftharpoons{\text{CTFT}} 2\pi S(\omega - \omega_0)$$

Most of the CTFT pairs are symmetric, with a scaling factor of 2π sometimes.

- For continuous LTI systems, if if $x(t)$ is periodic, then of $y(t)$ is periodic.

Since $x(t)$ is periodic, $x(t) = \sum_{k \in \mathbb{Z}} a_k e^{jk\omega_0 t}$

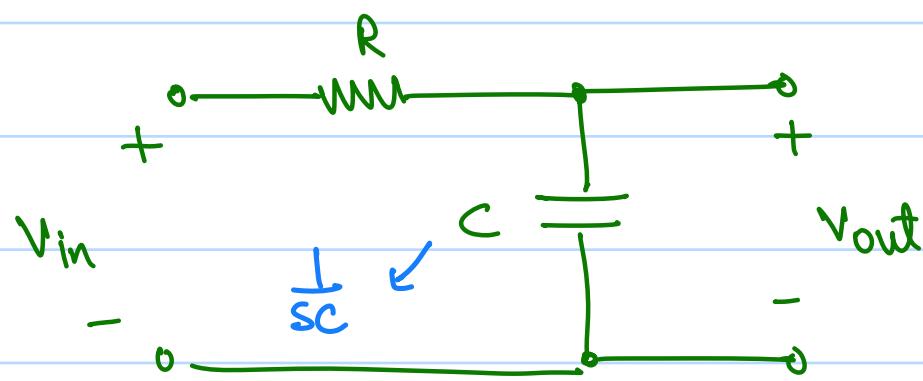
By linearity and Eigensignal nature,

$$y(t) = \sum_{k \in \mathbb{Z}} a_k h(k\omega_0) e^{jk\omega_0 t}$$

$$\Rightarrow y(t) = \sum_{k \in \mathbb{Z}} b_k e^{jk\omega_0 t}$$

$\therefore y(t)$ has FSR coefficients $a_k h(k\omega_0) + k$, making it a periodic signal.

Example: Use FT to find the VTC of the following circuit.



$$Z(\omega) = R + \frac{1}{j\omega C} = \frac{j\omega CR + 1}{j\omega C}$$

$$I(\omega) = \frac{V_{in}(\omega)}{Z(\omega)} = \frac{V_{in}(\omega)j\omega C}{j\omega CR + 1}$$

$$V_{out}(\omega) = I(\omega) \cdot \frac{1}{j\omega C} = \frac{V_{in}(\omega)}{j\omega CR + 1}$$

$$\frac{V_{out}(\omega)}{V_{in}(\omega)} = H(\omega) = \frac{1}{j\omega CR + 1}$$

→ Sampling :-

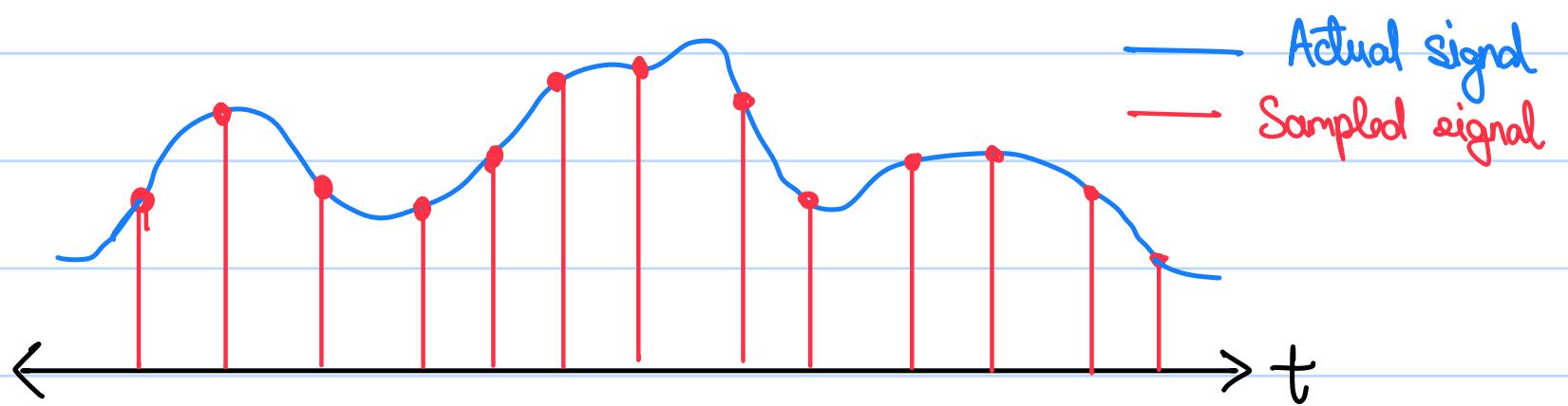
Process of converting continuous time signals into discrete time signals.

$$x[n] = x_c(nT_s) + n \in \mathbb{Z} \quad T_s - \text{Sampling period}$$

◦ Band-limited Signals :-

- Band : A range of frequencies

- Band-limited signals are those signals for which the spectrum $X(\omega) = 0 + \omega \notin [-B, B]$, where B is a finite constant.



- Clearly, shorter the sampling interval, lesser is the information lost by sampling.

Note:

$$\text{If } x(t) \xrightarrow{\mathcal{F}} X(\omega), \quad X(t) \xrightarrow{\mathcal{F}} 2\pi x(-\omega)$$

(Duality of FT)

$$x(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} X(\omega) e^{j\omega t} d\omega$$

$$Y(\omega) = \int_{-\infty}^{\infty} x(t) e^{-j\omega t} dt$$

$$\frac{Y(\omega)}{2\pi} = \frac{1}{2\pi} \int_{-\infty}^{\infty} X(t) e^{-j\omega t} dt$$

$$\text{Let } \alpha = -t, \quad d\alpha = -dt$$

$$\Rightarrow \frac{Y(\omega)}{2\pi} = \frac{1}{2\pi} \int_{\infty}^{-\infty} X(-\alpha) e^{-j\omega(-\alpha)} (-d\alpha)$$

$$\frac{Y(\omega)}{2\pi} = \frac{1}{2\pi} \int_{-\infty}^{\infty} X(-\alpha) e^{j\omega\alpha} d\alpha$$

$$x(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} X(\omega) e^{j\omega t} d\omega \xrightarrow{t=\alpha} \frac{1}{2\pi} \int_{-\infty}^{\infty} X(\alpha) e^{j\omega\alpha} d\alpha = x(\alpha)$$

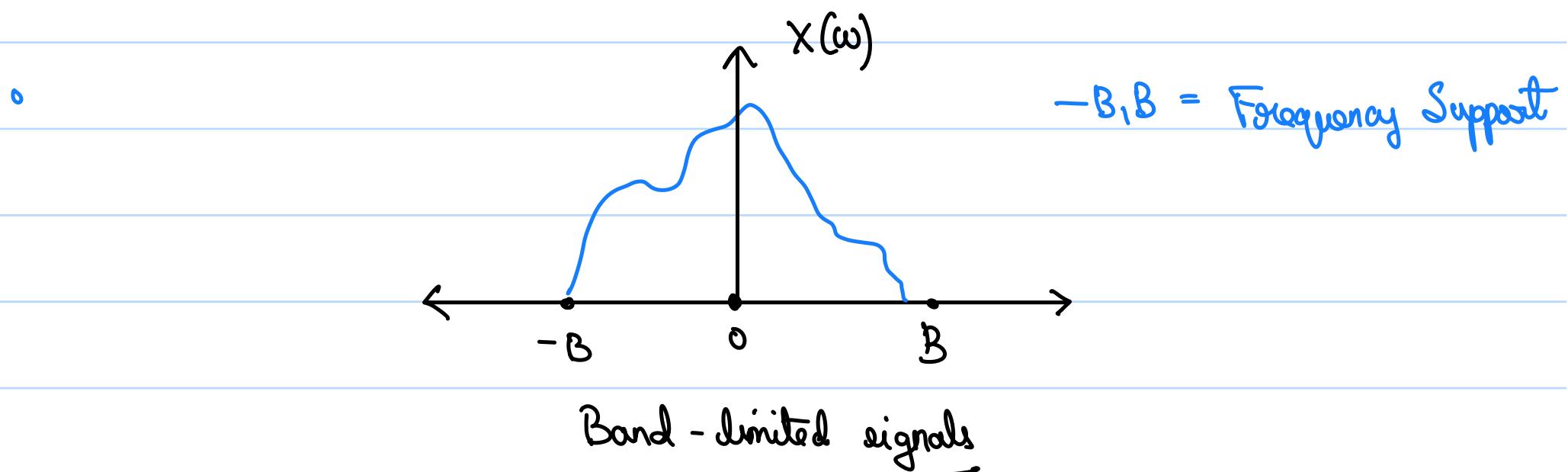
$$x(-\alpha) = \frac{1}{2\pi} \int_{-\infty}^{\infty} X(-\alpha) e^{j\omega\alpha} d\alpha$$

$$\Rightarrow \frac{Y(\omega)}{2\pi} = x(-\omega)$$

$$\Rightarrow Y(\omega) = \underline{2\pi x(-\omega)}$$

- If the sampling rate is $f_s = 1/T_s$, the sampled discrete-time signal is of the form,

$$x[n] = x_c(nT_s)$$



Examples of Band limited signals : Sine wave, $\text{sinc}(x)$.

Sampling Theorem :-

- Sampling generally causes a loss of information. But for band-limited signals, there is a certain Sampling frequency above which information is conserved.
- Information loss is due to the high frequency variation in between 2 samples.

- But, band limited signals have a hard limit on how fast those variations will be.

- Sampling Theorem: A band-limited signal with max. frequency ω_M can be perfectly recovered from its samples if the sampling frequency ω_s satisfies,

$$\omega_s > 2\omega_M$$

Proof:

$$\text{Impulse train rep. of Sampling: } p(t) = \sum_{n \in \mathbb{Z}} \delta(t - nT_s)$$

$$x_p(t) = x_c(t)p(t)$$

$$x_p(t) = \sum_{n \in \mathbb{Z}} x_c(nT_s) \delta(t - nT_s) \rightarrow \text{Sampled signal}$$

$$X_p(\omega) = \int_{-\infty}^{\infty} \sum_{n \in \mathbb{Z}} x_c(nT_s) \delta(t - nT_s) e^{-j\omega t} dt$$

$$= \sum_{n \in \mathbb{Z}} \int_{-\infty}^{\infty} x_c(nT_s) \delta(t - nT_s) e^{-j\omega t} dt$$

$$= \sum_{n \in \mathbb{Z}} x_c(nT_s) e^{-j\omega nT_s}$$

$$= \sum_{n \in \mathbb{Z}} x_c(nT_s) e^{-jk(T_s)\omega} \rightarrow \underline{\underline{\text{FSR}}}$$

$$\Rightarrow X_p(\omega) \text{ is periodic with period} = \frac{2\pi}{T_s} = \underline{\underline{\omega_s}} \text{ and FSR coefficients } a_n = x_c(nT_s)$$

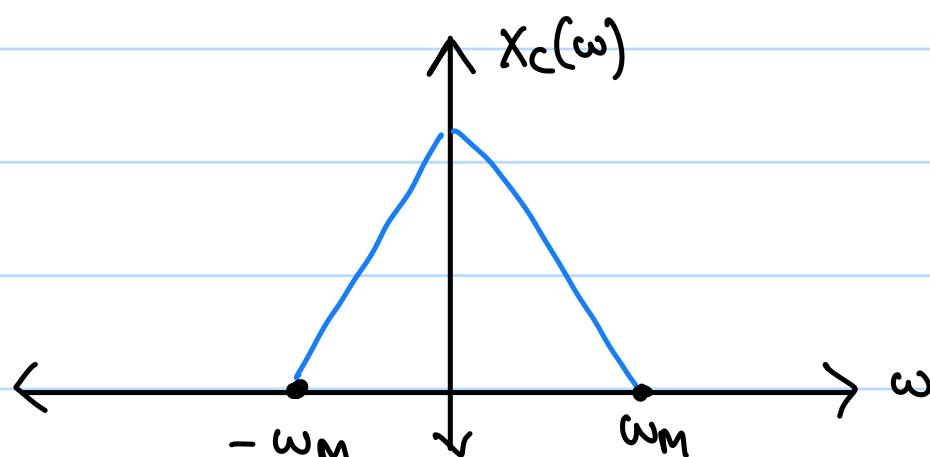
$$\text{Also, } X_p(\omega) = X_c(\omega) * P(\omega)$$

$$\begin{aligned} \Rightarrow X_p(\omega) &= \int_{-\infty}^{\infty} X_c(\alpha) \frac{2\pi}{T_s} \sum_{k \in \mathbb{Z}} S\left(\omega - \alpha + k \frac{2\pi}{T_s}\right) d\alpha \\ &= \sum_{k \in \mathbb{Z}} X_c\left(\omega - k \frac{2\pi}{T_s}\right) \\ &= \sum_{k \in \mathbb{Z}} X_c(\omega - kw_s) \end{aligned}$$

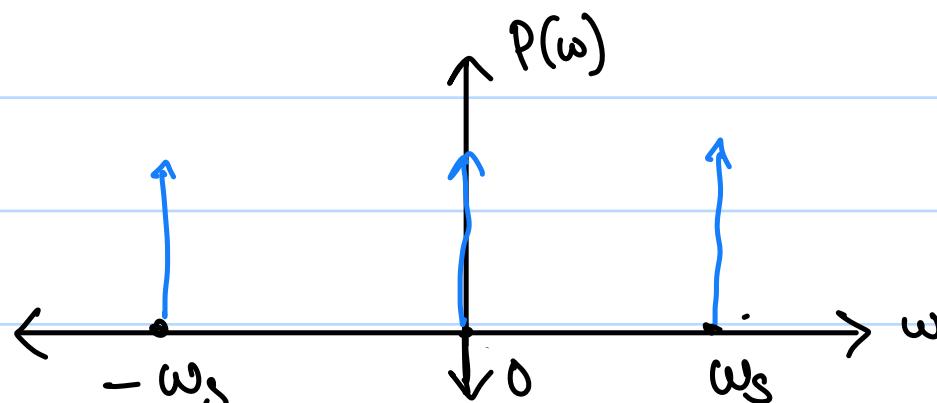
$$\Rightarrow \text{Within } (-w_s, w_s), \underline{X_p(\omega) = X_c(\omega)}$$

If $w_s > w_m$, the entire non-zero spectrum of $X_c(\omega)$ will be contained within the interval $(-w_s, w_s)$ of $X_p(\omega)$, ensuring that all the information is still present in $X_p(\omega)$, in the form of that spectrum.

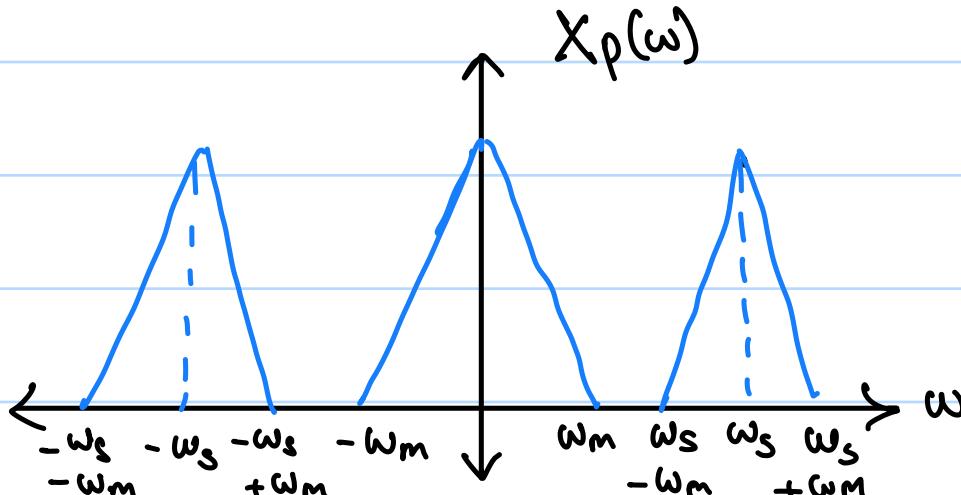
$X_c(\omega) :$



$P(\omega) :$



$X_p(\omega) :$



No overlap (loss of spectrum if $w_s > w_m$)

- To exactly reconstruct $x_c(t)$, we will pass $\alpha_p(t)$ through a low pass filter, with cutoff frequency $w_c \in (w_m, w_s - w_m)$

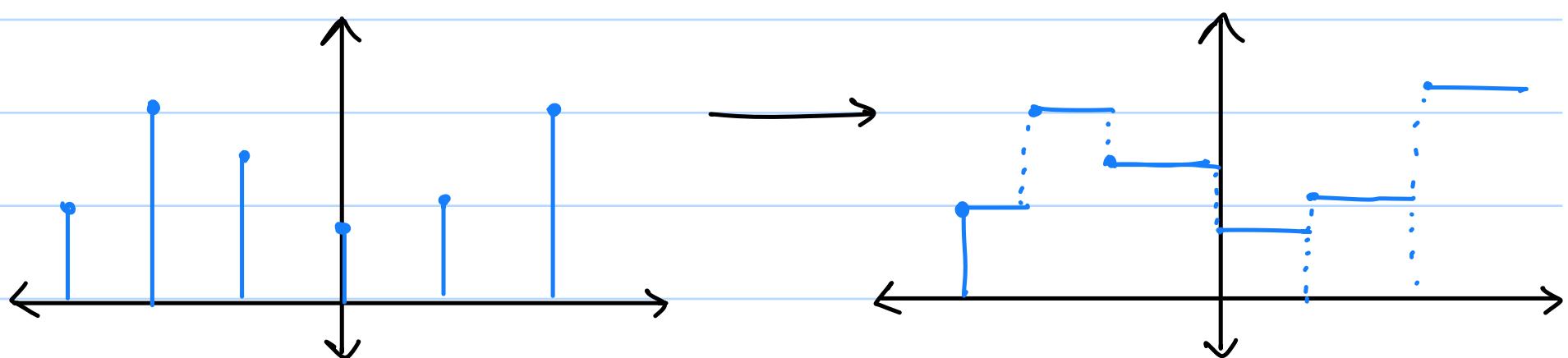
Ideal low pass filter = $\frac{\sin(\omega_0 t)}{\pi t}$

$$x_g(t) = \left[\sum_{n=-\infty}^{\infty} x_c(nT_s) \delta(t-nT_s) \right] * \frac{\sin(\omega_0 t)}{\pi t} \quad \omega_0 \in (w_m, w_s - w_m)$$

$$\Rightarrow x_g(t) = \sum_{n=-\infty}^{\infty} x_c(nT_s) \frac{\sin(\omega_0(t-nT_s))}{\pi(t-nT_s)} \rightarrow \text{Weighted sum of sinc functions}$$

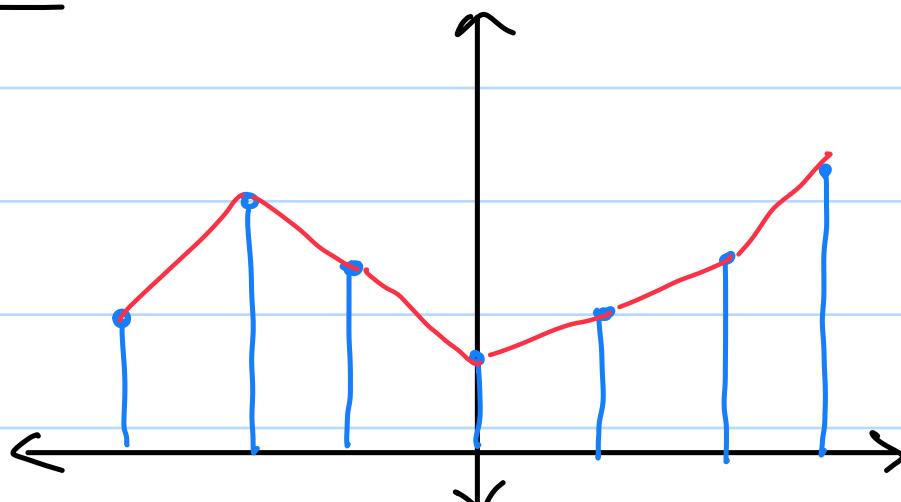
- Reconstruction :-

- Zero Order Hold / Piecewise Constant :

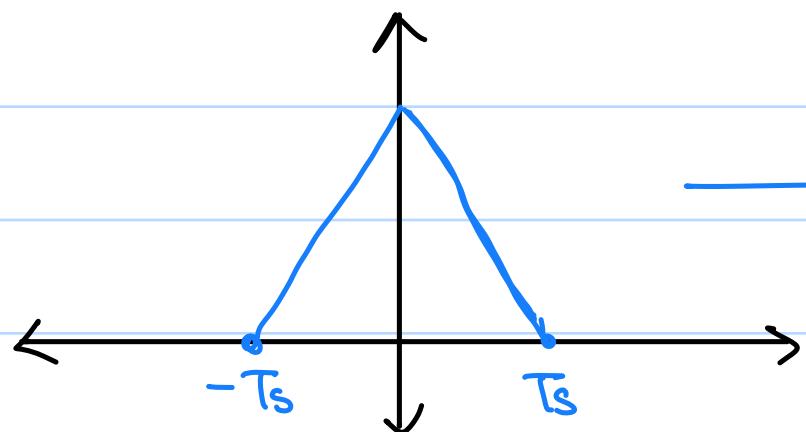


Takes the value of each sample and keeps it constant until the next sample.

- Linear Interpolation :



$h(t)$:



A non ideal low pass filter

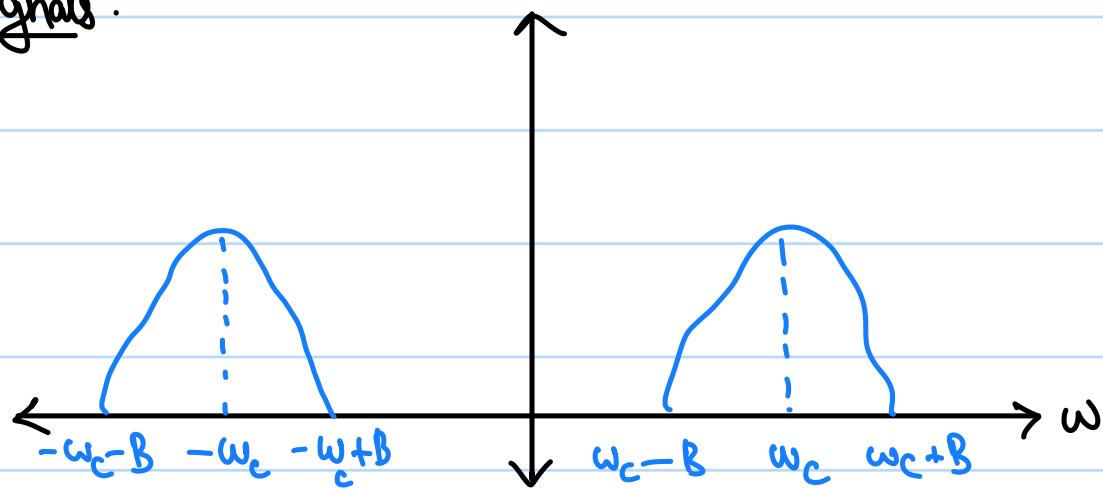
3) Sinc-Based Interpolation :-

$$x_r(t) = x_p(t) * \text{sinc}(\omega_s t)$$

This works since $\text{sinc}(t)$ acts like a non ideal low pass filter.

- Beyond Sampling Theorem :-

- i) Band Pass Signals:



Band pass signals are those signals which are non zero for intervals not centred at origin.

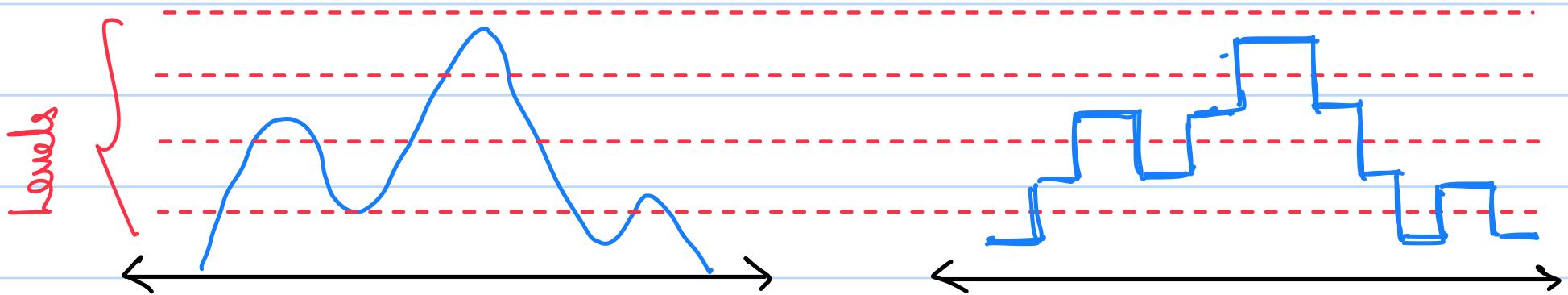
Such signals can be shifted to the origin, reducing the Nyquist frequency of the signal.

2) Frequency Compression :-

Using certain algorithms, a band limited signal can have a reduced bandwidth, albeit with the possibility of some information loss, but not as much as due to aliasing.

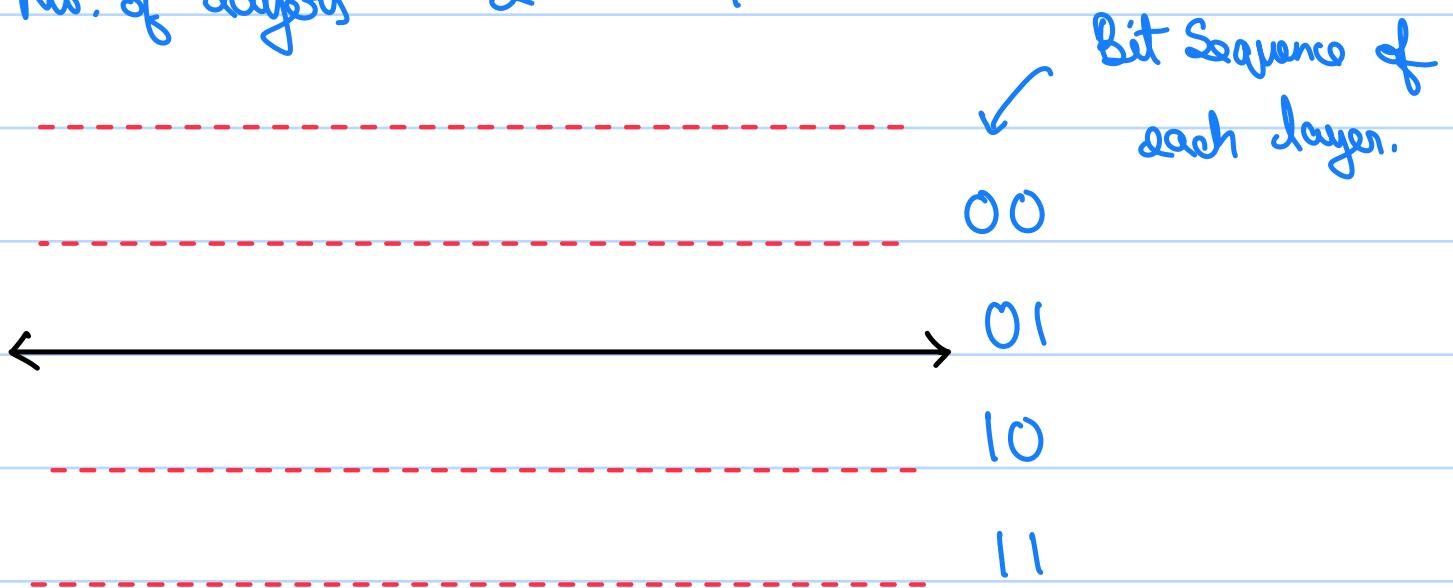
→ Quantization :-

- Sampling of the amplitude.
- Since the amplitude of a signal can be any real number, including irrational no.s, the bits needed to store a signal can become extremely large.
- Quantization divides the Amplitude interval into levels and assigns a binary sequence to each level.



- Sampling and Quantization are done in the ADC (Analog-Digital Converter)
- The no. of layers used in Quantization is 2^B , where B is the no. of bits assigned to each level.

If $B = 2$, No. of layers = $2^2 = 4$



- ° Channels: The no. of independent signals in a piece of information.

An RGB Image has 3 channels - R, G, B.

A Stereo Audio has 2 channels - Left, Right.

- ° Reconstruction of a Quantized Signal is never guaranteed to be exact, unlike in Sampling, which has the Nyquist theorem.

→ Overall Digital System :-



