

- iec 251

- Digital Signal Processing — Proakis & Manolakis

- Digital Signal Processing — A computer science perspective Jonathan

- Signal processing society  
dspguide.com.

## \* Signals

• Analog  $\rightarrow$  Digital

    ↳ Sampling  $\rightarrow$  Quantization

• Why digital?

    ⇒ Storage } Become easier

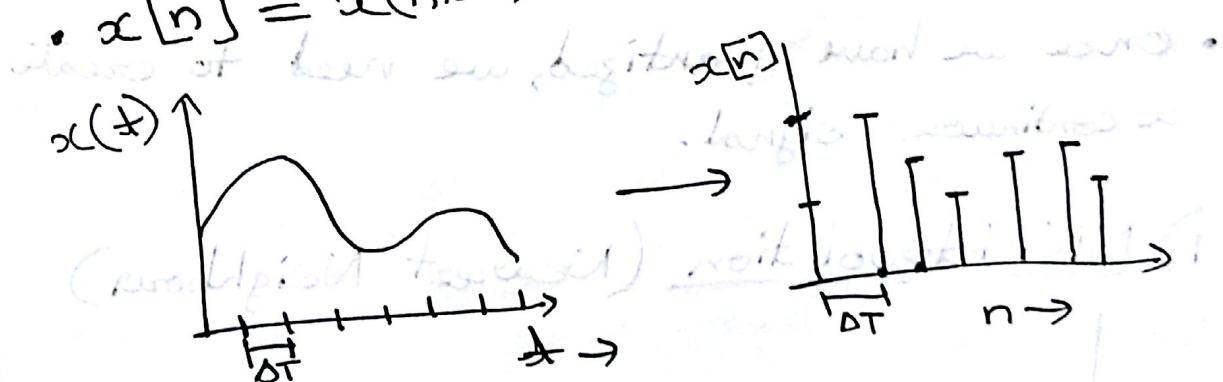
    ⇒ Reproduce signal  
(Analog data can deteriorate)

    ⇒ Modification of signal is easy.  
(Flexibility & Modularity)

    ⇒ Disadvantage is that digital signals are slow.

## \* Digital $\rightarrow$ Analog to Digital ; (Sampling)

•  $x[n] = x(n\Delta T)$



•  $f_s = \frac{1}{\Delta T}$  } Sampling Frequency

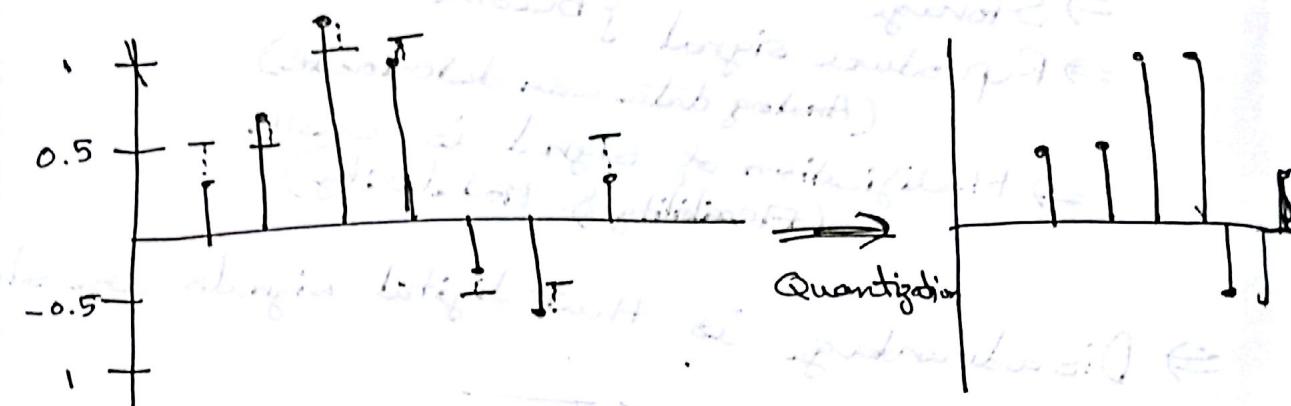
$$w_s = \frac{2\pi}{\Delta T}$$

## Shannon Frequency of Sampling; (Nyquist Frequency)

- If own max frequency of the signal is  $F_{\max}$ , then we want  $F_s \geq 2 \times F_{\max}$

## Quantization;

- We approximate the value of the signal rounding it off to the values allowed by the precision (levels) we allow.

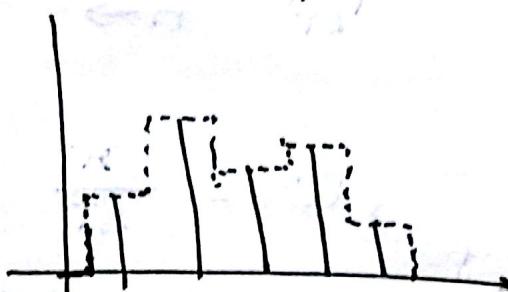


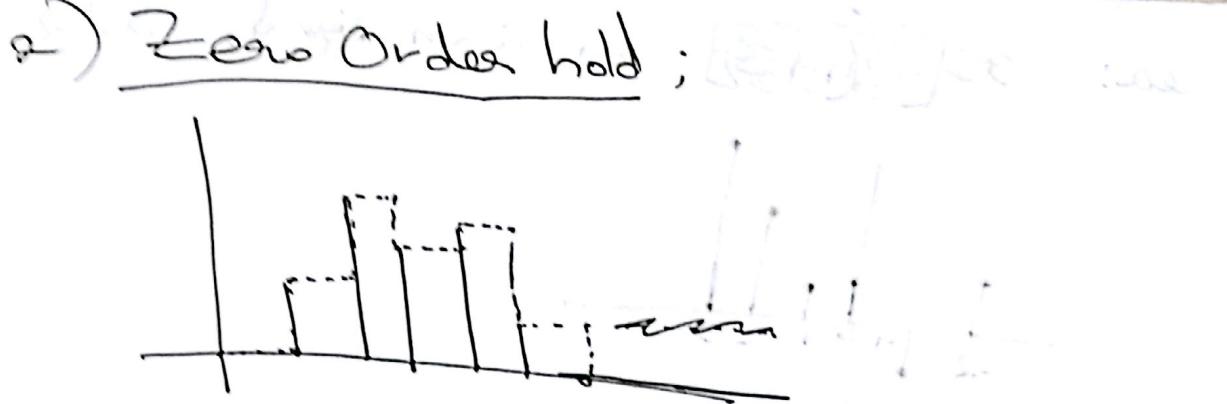
- Dividing y axis into equal intervals.

## Interpolation;

- Once we have quantized, we need to create a continuous signal.

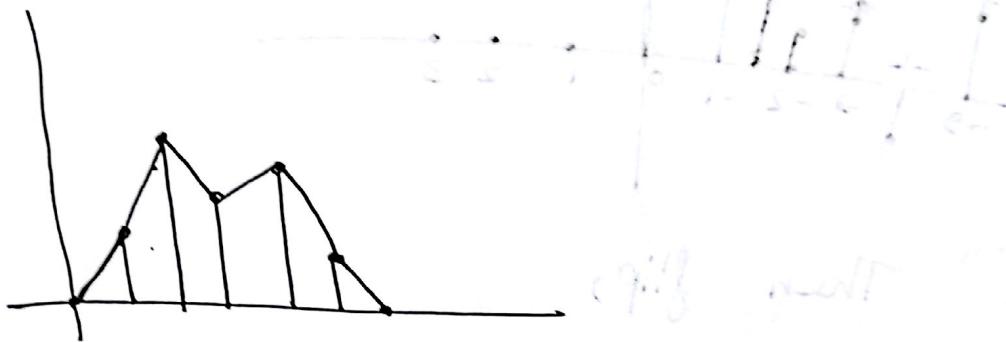
### 1) NN interpolation (Nearest Neighbour)





- Keep using the previous value until you encounter a new value.

### 3) Linear interpolation



- Linearly join our values (we use convolution functions here to get various linear interpolations).

### \*Basic Operations:

1) Flipping: ( $f(-x)$  becomes  $f(x)$ )  
 $x[n] \rightarrow x[-n]$

2) Shifting: ( $f(x) = f(x-2)$ ),  $x[n] \rightarrow x[n-2]$

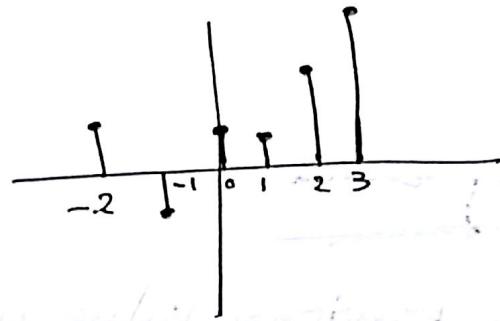
3) Scaling: ( $f(x) = f(2x)$ ),  $x[n] \rightarrow x[2n]$

↳ We will lose data if not continuous

\*\*\* Shift functions when we scale.

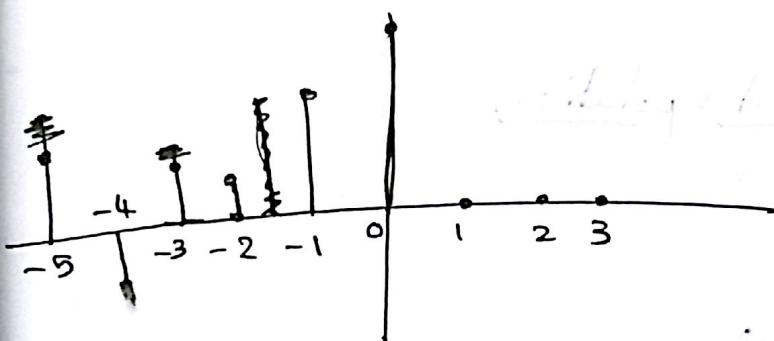
Order: ~~Shift~~ → Flip → ~~Shift~~ ~~Scale~~ } Hierarchy of operations  
(BODMAS)

ex:  ~~$x[2n+3]$~~  on given signal,  $x[-2n+3]$

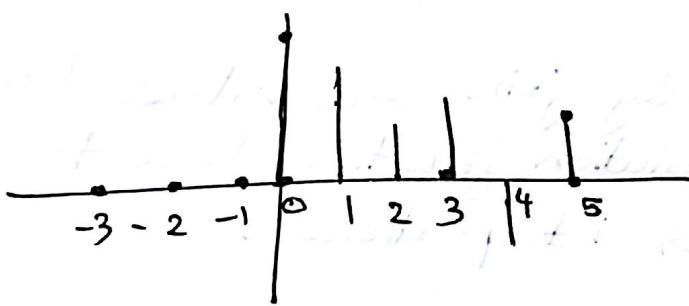


This is equivalent to the given part.

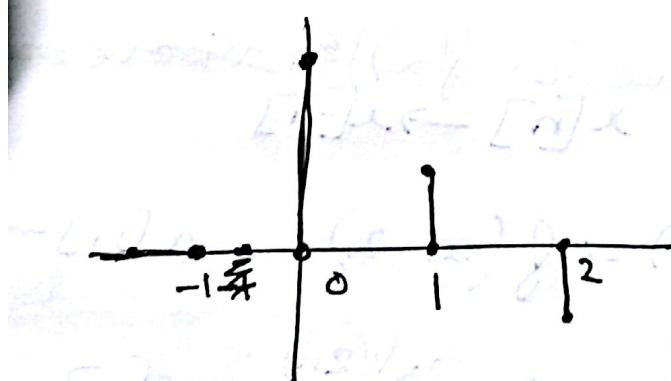
First Shift,  $\rightarrow$  observe ref



Then flip,



Then scale,



thus target what scale this will

relate to when obtained this graph

## Characteristics of Signals;

$$\begin{array}{ll} \text{1) Even} & \rightarrow x(n) = x(-n) \\ \text{Odd} & \rightarrow x(n) = -x(-n) \end{array}$$

- Any function can be represented as sum of even & odd functions.

$$* x[n] = \underbrace{\frac{1}{2}(x(n) + x(-n))}_{\text{Even part of } x[n]} + \underbrace{\frac{1}{2}(x(n) - x(-n))}_{\text{odd part of } x[n]}$$

## 2) Periodic Functions;

$$x[n] = x[n+N]$$

## \* 3) Energy | Power Signal;

$$E_x = \sum_{-\infty}^{\infty} |x[n]|^2$$

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

- If  $E_x$  is finite,  $P_x = 0$

$\overbrace{x}^{\longrightarrow}$

ex:  $u[n] = \begin{cases} 1 & \text{for } n \geq 0 \\ 0 & \text{otherwise} \end{cases}$

$$E_x = \infty$$

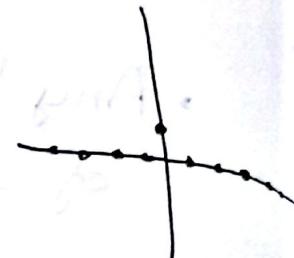
$$P_x = \lim_{N \rightarrow \infty} \frac{1}{2N+1} (N+1) = \frac{1}{2} //$$

$\overbrace{x}^{\longrightarrow}$

## \*Special Signals:

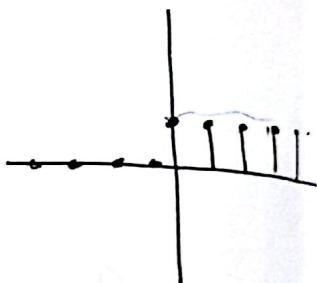
### 1) Delta:

$$S[n] = \begin{cases} 1 & \text{for } n=0 \\ 0 & \text{otherwise} \end{cases}$$



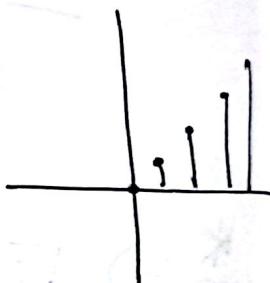
### 2) Unit Step:

$$U[n] = \begin{cases} 1 & \text{for } n \geq 0 \\ 0 & \text{otherwise} \end{cases}$$



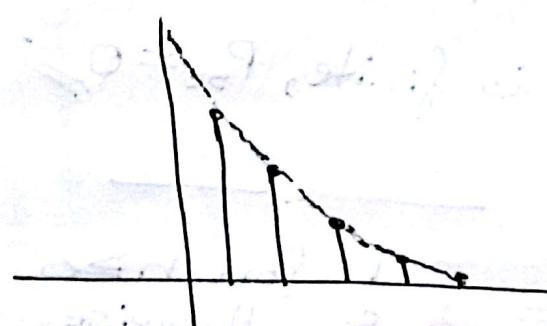
### 3) Ramp:

$$n[n] = \begin{cases} n, & n \geq 0 \\ 0, & \text{otherwise} \end{cases}$$



### 4) Exponential:

$$|a|^n u[n], 0 < |a| < 1$$



Note:  $u[n] = \sum_{k=0}^n s[n-k] = \sum_{k=-\infty}^n s[k]$

$s[n] = u[n] - u[n-1]$

( $s(t) = \frac{d}{dt} u(t) \rightarrow$  If it's continuous)

Note: Generalized function,

$x[n] = \sum_{k=-\infty}^{\infty} x[k] \cdot \delta[n-k]$

$$x[n] = a_{-2} \delta[n+2] + a_{-1} \delta[n+1] + a_0 \delta[n] + a_1 \delta[n-1] + a_2 \delta[n-2] + \dots$$

$$\Rightarrow a_k = x[k]$$

## \* System:



• Filter  $\Rightarrow y[n] = \frac{1}{P} \sum_{k=-P/2}^{P/2} x[n+k]$

$\Rightarrow$  So for every  $n$ , we approximate it to neighbours.

averaging filter ex:  $\frac{1}{5} [ \underbrace{1 \ 1 \ 1 \ 1 \ 1}_{\text{Window}} ]$

Window which is used so as to average each point.

This would reduce variations.

## \* Properties of Systems:

### 1) Static: (memoryless)

- If we don't need to store previous values,
- ex:  $y[n] = 7x[n]$   
 $\Rightarrow y[n] = 3x[n-1]$  is not a static.

### 2) Causal | Non-causal:

- Output can use values of past but not the future.
- ex:  $y[n] = 7x[n-1] + 3x[n-2]$   
 $\Rightarrow y[n] = x[n+1]$  is not causal.

### 3) Stable | Non-Stable:

- If input is bounded (finite), output is bounded (finite).  
 $\Rightarrow$  This is a stable system.

### \* 4) Linear:

- A system is linear if its additive & homogenous.  
 $x_1[n] \xrightarrow{H} y_1[n]$  &  $x_2[n] \xrightarrow{H} y_2[n]$

a) Additivity:  $x_1[n] + x_2[n] \xrightarrow{\text{then}} y_1[n] + y_2[n]$

b) Homogeneity:  $x_1[n] \xrightarrow{H} y_1[n]$   
 $\xrightarrow{x} a \cdot x_1[n] \xrightarrow{H} ay_1[n]$

ex:  $y[n] = 2x[n] + 7$

$x_1[n] \rightarrow 2x_1[n] + 7 = y_1[n]$

$x_2[n] \rightarrow 2x_2[n] + 7 = y_2[n]$

Let  $z[n] = x_1[n] + x_2[n]$

$z[n] \rightarrow \boxed{H} \rightarrow 2z[n] + 7$

$= 2x_1[n] + 2x_2[n] + 7$

$\neq y_1[n] + y_2[n]$

$\therefore$  Non linear.

\* \*

\* 5) Time invariance: (True only if coefficient of  $n$  is  $1^{\text{st}}$  power = 1 for the system)  
 ie:  $y[n] = x[n+a]$  where  $a=1$ .

If,  $x[n] \xrightarrow{H} y[n]$

Then,  $x[n-n_0] \xrightarrow{H} y[n-n_0]$

$\Rightarrow$  Then the system is time invariant

$\Rightarrow$  Then the system is time invariant

$\Rightarrow$  Then the system is time invariant

ex:  $y[n] = x[n^2] \rightarrow$  Not time invariant

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

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

Now lets consider  $z[n] = x[n-n_0]$

Lets check if output =  $y[n-n_0]$  or not.

$\Rightarrow z[n] \rightarrow \boxed{H} \rightarrow z[n^2]$

$\Rightarrow z[n^2] = x[n^2-n_0]$

From ①,  $x[n-n_0] \neq x[n^2-n_0]$

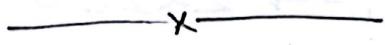
$$\text{ex: } y[n] = x[n] - 3x[n-1]$$

\*

⇒ This is time invariant

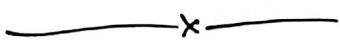
since  $a \neq b$ ,  $a = 1, b = 0$   
 $a = 0, b = -1$

∴  $a$  is 1 in both cases  
∴ It's time invariant

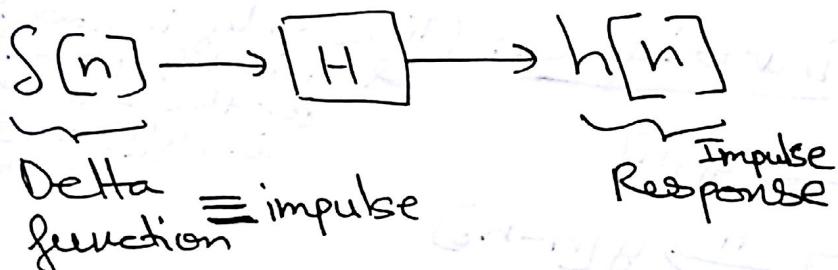


### \* LTI System:

- If a system is Linear & Time Invariant, it's called an LTI system.



### \* Impulse Response:



(For an LTI)

⇒ An impulse response, if we know it, then we can find the response of the system for any input.

$$s[n] \xrightarrow{\text{LTI}} h[n]$$

$$s[n-k] \xrightarrow{\text{LTI}} h[n-k]$$

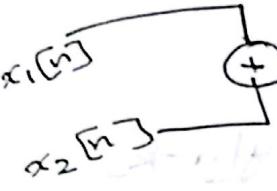
$$\alpha_x s[n-k] \xrightarrow{\text{LTI}} \alpha_x h[n-k]$$

$$x[k] s[n-k] \xrightarrow{\text{LTI}} x[k] h[n-k]$$

$$\Rightarrow \sum_{k=-\infty}^{\infty} x[k] s[n-k] \xrightarrow{\text{LTI}} \sum_{k=-\infty}^{\infty} x[k] h[n-k]$$

$\Rightarrow x[n] \xrightarrow{\text{LTI}}$

## Syntax: (Operations on signals representation)

- 

Multiply can also be done  
 $y[n] = x_1[n] + x_2[n]$
- $x[n] \xrightarrow{a} y[n] = ax[n]$   
 (Multiplication with a scalar)
- $x[n] \xrightarrow{z^{-1}} y[n] = x[n-1]$   
 (Time delay)
- $x[n] \xrightarrow{z} y[n] = x[n+1]$   
 (Time advance)

\* Note: ~~Impulse response~~ ~~Convolution sum~~  $\dots$

\*\*  $x[n] \xrightarrow{\frac{1}{2}, \frac{1}{2}, \frac{1}{4}} y[n]$

$y[n] = \frac{1}{2}x[n] + \frac{1}{2}x[n-1] + \frac{1}{4}y[n-1]$

continuing impulse response,  $\Rightarrow \therefore$  We can see, Treat like constants

$$\begin{aligned} H(x[n]) &= H\left(\sum x[k]\delta[n-k]\right) \\ &= \sum x[k] \cdot H(\delta[n-k]) \\ &= \sum x[k] \cdot h[n-k] \end{aligned}$$

$$\Rightarrow y[n] = H(x[n]) = \underbrace{\sum_{-\infty}^{\infty} x[k]h[n-k]}_{\text{Convolution sum}}$$

$$y[n] = \underbrace{x[n] * h[n]}_{\text{Convolution operator}}$$

## Convolution:

$\Rightarrow x[n] \& y[n]$

$$\Rightarrow \text{conv}(x[n], y[n]) = \sum_{k=-\infty}^{\infty} x[k] \cdot y[n-k]$$

- This is the convolution operator.

\*\*

ex:  $h[n] = [1 \ 2 \ 1 \ -1]$  } Impulse response

$x[n] = [1 \ 2 \ 3 \ 1]$  } Input signal  
↑ Indicates 0 ( $n=0$ )

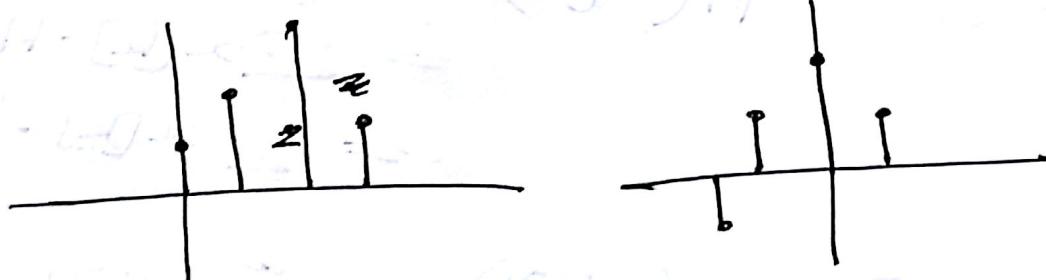
$$y[0] = \sum_{k=-\infty}^{\infty} x[k] \cdot h[0-k] = 1 + 0 + 0 + 0 = 1$$

$$y[1] = \sum_{k=-\infty}^{\infty} x[k] \cdot h[1-k] = 2 + 2 + 0 + 0 = 4$$

$$y[2] = \sum_{k=-\infty}^{\infty} x[k] \cdot h[2-k] = 1 + 2 + 3 + 1 = 7$$

Let  $y[n] = \cancel{x[n] * h[n]}$   
(Convolution)

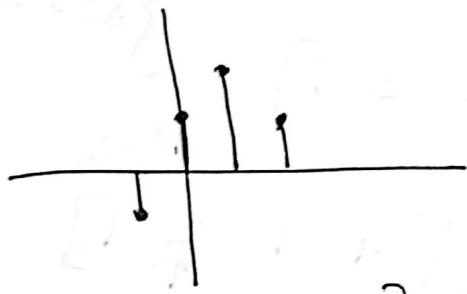
$$\Rightarrow y[0] = \sum_{k=-\infty}^{\infty} x[k] \cdot h[0-k]$$



$$(x[k], h[-k])_k = (x[k], h[-k])_{k=0} = h[-k]$$

$$y[0] = 2 + 2 = 4 //$$

$$\Rightarrow y[1] = \sum_{k=-\infty}^{\infty} x[k] \cdot h[1-k]$$



$h[1-k]$  } Shift first, then scale / flip

$$\therefore y[1] = 1 + 4 + 3 = 8$$

Similarly, find

$$\Rightarrow y[2], y[3], \dots$$

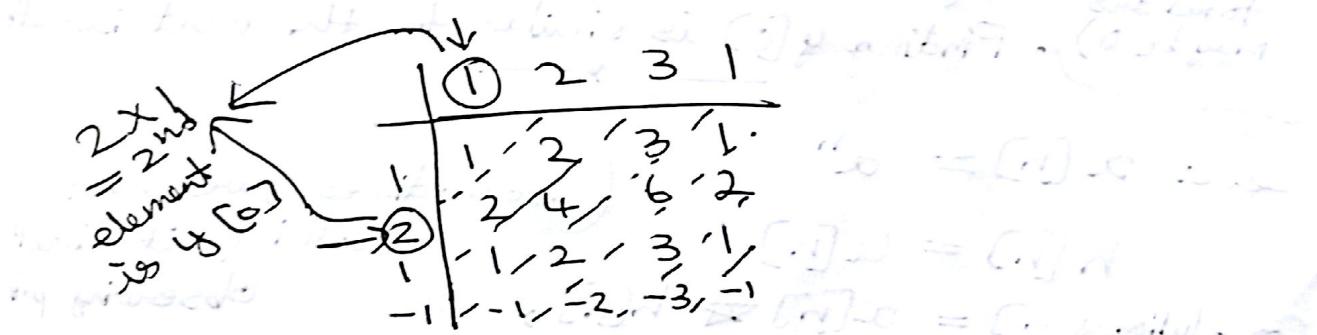


Shortcut for convolution:



$$h[n] = [1 \ 2 \ 1 \ -1]$$

$$x[n] = [1 \ 2 \ 3 \ 1]$$



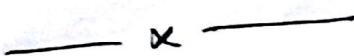
$y[1] = [1 \ 2 \ 3 \ 1] \cdot [1 \ 2 \ 3 \ 1] = 1 + 4 + 8 + 3 = 18$

$\Rightarrow$  Sum of diagonals are the outputs,

$$y[n] = [1 \ 4 \ 8 \ 8 \ 3 \ -2 \ -1]$$

$2 \times 1 = 2^{\text{nd}}$  element

$y[0]$  is this  $2^{\text{nd}}$  element



$$= 1 + 4 + 8 + 8 + 3 - 2 - 1 = 27$$

\* \* Shortcut 2 for convolution,

$$h[n] = [1 \ 2 \ 1 \ -1] \quad \left\{ \begin{array}{l} h[0] \text{ is } 2 \times 1 \\ \text{element} \\ = 2^{\text{nd}} \text{ element} \end{array} \right.$$
$$x[n] = [1 \ 2 \ 3 \ 1] \quad \cdot$$

$$\Rightarrow \text{Flip } h[n] \Rightarrow [-1 \ 1 \ 2 \ 1] \quad \left\{ \begin{array}{l} \text{Consider} \\ \text{this as} \\ \text{a filter} \end{array} \right.$$

Now consider,

$$x[n] \Rightarrow [0 \ 0 \ 0 \ 1 \ 2 \ 3 \ 1 \ 0 \ 0 \ 0]$$

Start with  
rightmost  
element of  $h[n]$  flip,  
↓  
with the 1st element  
of  $x[n]$ .

$$[-1 \ 1 \ 2 \ 1]$$

~~↓~~

$$\text{Start } \leftarrow y[n] \Rightarrow [\dots \dots \dots]$$

non zero value  
(ie at least  
one element  
is nonzero  
in  $x[n]$ ,  
total ans  
may be 0)

Since the filter is positioned  
at the 4th element.  
So, wherever the arrow points, the  
value at that  $y[n]$  is the result  
of the filter from that point.  
Finding  $y[0]$  is similar to the matrix method.

$$\text{ex: } x[n] = a^n$$

$$h[n] = u[n]$$

(Sometimes, we can try  
solving it directly)

$$\text{Convolution, } y[n] = x[n] * h[n],$$

observing pattern

$$y[0] = \sum x[k] \cdot h[-k] = 1$$

$$y[1] = \sum x[k] \cdot h[1-k] = 1+a$$

:

$$y[2] = 1+a+a^2$$

:

$$y[\infty] = \frac{1}{1-a}$$

## \* Properties of Convolution:

$$\cdot y[n] = x[n] \star h[n] = \sum_{k=-\infty}^{\infty} x[k] \cdot h[n-k]$$

### ① Commutative:

$$y[n] = x[n] \star h[n] = h[n] \star x[n]$$

Proof:

$$\text{Let, } n-k=m$$

$$k=n-m$$

$$\sum_{k=-\infty}^{\infty} x[k] \cdot h[n-k] = \sum_{m=-\infty}^{\infty} \underbrace{x[n-m]}_{x[m]} \cdot \underbrace{h[n-m]}_{h[m]}$$

∴ It's commutative

### ② Distributive:

$$x[n] \star [h_1[n] + h_2[n]] = x[n] \star h_1[n] + x[n] \star h_2[n]$$

### ③ Associative:

$$(x[n] \star h_1[n]) \star h_2[n] = x[n] \star (h_1[n] \star h_2[n])$$

## \* Correlation:

- Similar to convolution, but here,

$$y[n] = \sum_{k=-\infty}^{\infty} x[k] \cdot h[n+k]$$

- To find the correlation, use 'shortcut' for convolution (on prev page), except that you don't flip  $h[n]$  before starting to multiply.

\*Note: Auto-correlation: (Correlation with 'itself')

$$y[n] = \sum_{k=-\infty}^{\infty} x[k] \cdot x[n+k]$$

— x —

\* 2D Convolution: (Image processing)

\*\*

- Flip ~~h~~  $h_{w \times w}$  over both  $x$  &  $y$  axis.
- Then similar to the 1D method, keep going over all the points of  $x_{m \times n}$  and generate output  $y_{p \times q}$ .
- Using some point of the filter  $h$  as the center get  $y(pq)$  similar to normal conv.

Ex: An averaging filter (i.e. all 1's) would give us a blurred image. → In MATLAB, the values of the filter are all divided by  $n \times m$ .

So a filter  
of all 1's is an  
averaging filter.

Rows  
& Cols  
of filter

c: Gaussian filter:  $h[k] = A e^{-\frac{(k-\mu)^2}{2\sigma^2}}$ ,  $A=1$ ,  $\mu=0$ ,  $\sigma=1$

$$h[k] = e^{-\frac{k^2}{2}}$$

$$\Rightarrow \text{In 2D, } h[x,y] = A e^{-\left[\frac{(x-\mu_x)^2}{2\sigma_x^2} + \frac{(y-\mu_y)^2}{2\sigma_y^2}\right]}$$

— x —

\* Statistical Measures:

- Mean, standard deviation etc.

1) Mean:  $\mu_x = \frac{1}{N} \sum_{i=0}^{N-1} x[i]$

\*2) Standard deviation & Variance:

$$\sigma^2 = \frac{1}{N-1} \sum_{i=0}^{N-1} (x[i] - \mu)^2$$

Whenever

we use a

Sample out of

larger set, we divide by  $N-1$ .

$$\Rightarrow \sigma^2 = \frac{1}{N-1} \left[ \sum_{i=0}^{N-1} (x[i]^2 + \mu^2 - 2\mu x[i]) \right]$$

$$= \frac{1}{N-1} \left[ \sum (x[i]^2) + \sum \mu^2 - 2\mu \sum x[i] \right]$$

$$= \frac{1}{N-1} \left[ \sum (x[i]^2) + N\mu^2 - 2\mu N \sum x[i] \right]$$

$$= \frac{1}{N-1} \left[ \sum (x[i]^2) + NM^2 - 2N^2 \bar{x} \right]$$

$$\sigma^2 = \frac{1}{N-1} \left[ \sum (x[i]^2) - M^2 N \right]$$

$$= \frac{1}{N-1} \left[ \sum (x[i]^2) - N \left( \frac{\sum x[i]}{N} \right)^2 \right]$$

$$= \frac{1}{N-1} \left[ \sum (x[i]^2) - \frac{1}{N} (\sum x[i])^2 \right]$$

$$\sigma^2 = \frac{1}{N-1} \left[ \text{sum of squared} - \frac{(\text{sum})^2}{N} \right]$$

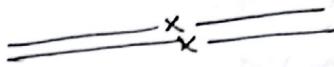
\*  $\Rightarrow$  We can see that if we add another sample to the system, we can easily get  $\sigma^2 \rightarrow$  running statistics.

\* Histogram: → Does not capture distribution!

- Variable vs Frequencies

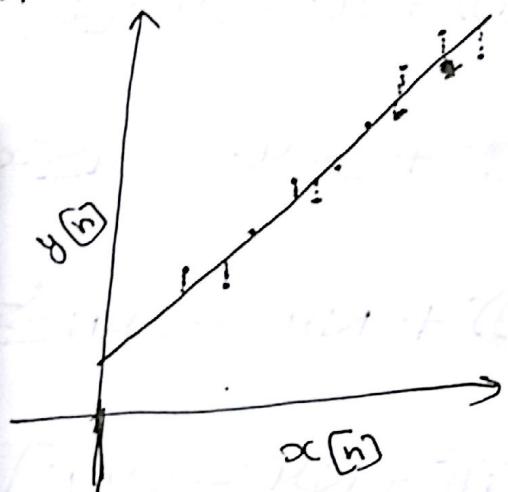
- In a normalized histogram, we plot against normalized frequency → Probability Mass Function (PMF).

⇒ Histograms <sup>of images</sup> only store values, not location of occurrence of intensity of pixels etc, so data is 1D.



\* Linear Regression: (Ordinary least squares)

approximated.



$$\hat{y}[n] = A + Bx[n]$$

↑  
Intercept      ↑  
Slope

- $e[k] = [y[k] - (A + Bx[k])]$

↑  
error

$$\Rightarrow E = \sum_{k=0}^{N-1} e[k]^2 = \sum_{k=0}^{N-1} (y[k] - (A + Bx[k]))^2$$

⇒ We try to minimize this error  $E$ .

$$\Rightarrow \frac{\partial E}{\partial A} = 0 \quad \& \quad \frac{\partial E}{\partial B} = 0$$

$$\Rightarrow \frac{\partial E}{\partial A} = -2 \sum (y[k] - A - Bx[k]) = 0$$

$$\Rightarrow \sum y[k] - NA - B \sum x[k] = 0$$

$$\Rightarrow A = \frac{\sum y[k]}{N} - \frac{B \sum x[k]}{N}$$

$$A = \mu_y - B \mu_x \rightarrow ②$$

$$\text{Now, } \frac{dE}{dB} = 0 = -2 \sum (y[k] - A - Bx[k]) \cdot x[k]$$

$$\Rightarrow \sum y[k]x[k] - \sum Ax[k] - B \sum x^2[k] = 0$$

Using A from ①, we get

$$\Rightarrow \sum y[k]x[k] - \left( \frac{\sum y[k]}{N} - \frac{B \sum x[k]}{N} \right) \sum x[k]$$

$$- B \sum x^2[k] = 0$$

$$\Rightarrow \sum x[k]y[k] - \frac{\sum y[k]}{N} \sum x[k] - B \sum x^2[k] + B \left( \frac{\sum x[k]^2}{N} \right)$$

$$\Rightarrow B = \frac{\sum x[k]y[k] - \frac{\sum x[k]y[k]}{N}}{\sum x^2[k] - \frac{(\sum x[k])^2}{N}}$$

- So we can see that B can be calculated easily if a new value is added, so greenning values of A & B can be calculated.

Note: The line we get from linear regression always pass through the mean  $x$ , mean  $y$ .

so we get about 10 voltage  $y$  for 10 different  $x$  values, so we get one  $y$  for one  $x$ .

## \* Sinusoids & Complex Exponentials:

Note:  $\cos(\omega n)$  is periodic if we can find an integer  $N = 2\pi k/\omega$ , where  $k$  is an integer.  $\cos(\omega n) = \cos(\omega(n+N))$ .

## \* Vector Spaces: (V)

- 1) A set is a vector space if  $\forall u, v \in V$ ,  $u+v \in V$
- 2)  $\forall u \in V \& a \in \mathbb{R}$ ,  $au \in V$

## : Spanning Set: (S)

- A set  $S = \{v_1, v_2, \dots, v_n\}$  is said to span a vector space  $V$  if for all  $v \in V$ ,

$$v = \alpha_1 v_1 + \alpha_2 v_2 + \dots + \alpha_n v_n$$

$\alpha_1 \in \mathbb{R}, \alpha_2 \in \mathbb{R}, \dots, \alpha_n \in \mathbb{R}$ .

Linear Spanning Set

## Linear Independence:

- A set  $S = \{v_1, v_2, \dots, v_n\}$  is said to be linearly independent if
$$\alpha_1 v_1 + \alpha_2 v_2 + \dots + \alpha_n v_n = 0 \rightarrow (1)$$
then  $\alpha_1 = 0$  and  $\alpha_2 = 0 \dots$  and  $\alpha_n = 0$ .
- If ~~the~~ equation (1) holds for any one or more non-zero  $\alpha$ , then its linearly dependent.

ex: Is  $S = \left\{ \begin{bmatrix} 1 \\ -1 \\ 2 \end{bmatrix}, \begin{bmatrix} 2 \\ 1 \\ 3 \end{bmatrix}, \begin{bmatrix} -1 \\ 0 \\ 2 \end{bmatrix} \right\}$  a spanning set?

Well if it's a basis

$$\Rightarrow x_1 + 2x_2 - x_3 = a$$

$$-x_1 + x_2 = b$$

$$2x_1 + 3x_2 + 2x_3 = c$$

On solving,

$$\Rightarrow x_2 = \frac{2a + 4b + c}{11}$$

$$x_3 = 3x_2 - a - b \quad \left. \begin{array}{l} \text{So they are all} \\ \text{only dependent} \\ \text{on } a, b, c \end{array} \right\}$$

$$x_1 = x_2 - b$$

$\therefore$  They form  
a spanning  
set.

and  $\Rightarrow$  Substituting  $x_2$  and  $x_3$

$$\underline{x} = \underline{x}$$

### \* Inner Product:

• Generalization of dot product.

$$\bullet \langle u, v \rangle = \sum_{k=0}^{N-1} u[k] \cdot \overline{v[k]}$$

Complex  
conjugate.

with  $\underline{x}$  and  $\underline{y}$ .

### \* Orthogonality:

• If  $\langle u, v \rangle = 0$ , they are orthogonal vectors.

$(u, v)$

### \* Norm:

•  $\|u\| = \sqrt{\left( \sum_{k=0}^{N-1} u[k]^2 \right)}$ , this is the norm of a vector.

$$\bullet \|u\|^2 = \langle u, u \rangle$$

$$\underline{x} = \underline{x}$$

\*Note: Vectors  $u$  &  $v$ , if

$\langle u, v \rangle = 0$  &  $\|u\| = 1$  &  $\|v\| = 1$   
then  $u$  &  $v$  are orthonormal.

\*  
Theorem: If a set  $S = \{v_1, v_2, \dots, v_n\}$  is non zero orthogonal, it is linearly independent.

$$\Rightarrow \alpha_1 v_1 + \alpha_2 v_2 + \dots + \alpha_n v_n = 0, \text{ For some values of } \alpha_i$$

⇒ Take inner product with  $v_m$ ,

$$\langle v_m, \alpha_1 v_1 + \alpha_2 v_2 + \dots + \alpha_n v_n \rangle = \langle v_m, 0 \rangle \\ = 0$$

→ We know  $\langle v_i, v_j \rangle = 0$  if  $i \neq j$  {Orthogonal}   
⇒ So if LHS = RHS, then

$$\alpha_m \langle v_m, v_m \rangle = 0, \text{ then } \alpha_m = 0$$

Considering all  $m$  from 1 to  $n$ ,  
 $\therefore \forall m \in 1, n, \alpha_m = 0$

∴ If a set is orthogonal, then  
for  $\alpha_1 v_1 + \alpha_2 v_2 + \dots + \alpha_n v_n = 0$ , it  
must be linearly independent.

\*Basis : (Minimal span)

- A linearly independent set  $S = \{v_1, \dots, v_n\}$  which spans the vector space  $V$  is called the basis of  $V$ .

\* Theorem of orthogonal decomposition:

\*  $v = \alpha_0 v_0 + \alpha_1 v_1 + \dots + \alpha_{n-1} v_{n-1}$ , for any  $v \in V$   
 then,  $\alpha_k = \frac{\langle v, v_k \rangle}{\langle v_k, v_k \rangle}$  if  $\{v_0, v_1, \dots, v_{n-1}\}$  are an orthogonal set

Proof:

$$\langle v, v_k \rangle = \langle (\alpha_0 v_0 + \alpha_1 v_1 + \dots + \alpha_{n-1} v_{n-1}), v_k \rangle$$

$$\langle v, v_k \rangle = \alpha_k \langle v_k, v_k \rangle,$$

Since all others are 0,  
from orthogonality.

$$\Rightarrow \alpha_k = \frac{\langle v, v_k \rangle}{\langle v_k, v_k \rangle}$$

ex:  $v_1 = \begin{bmatrix} \frac{1}{3\sqrt{2}} \\ \frac{1}{3\sqrt{2}} \\ -\frac{4}{3\sqrt{2}} \end{bmatrix}, v_2 = \begin{bmatrix} 2/3 \\ 2/3 \\ 1/3 \end{bmatrix}, v_3 = \begin{bmatrix} 1/\sqrt{2} \\ -1/\sqrt{2} \\ 0 \end{bmatrix}$

- This is an orthogonal set of vectors
- $\langle v_1, v_2 \rangle = \langle v_2, v_3 \rangle = \langle v_1, v_3 \rangle = 0$

$\Rightarrow v = \begin{bmatrix} ? \\ ? \\ ? \end{bmatrix}$ , let us consider this,

$$\alpha_1 = \frac{-2}{\sqrt{18+18+18}} = \frac{-2}{3\sqrt{2}} = -\frac{\sqrt{2}}{3}$$

$$\alpha_2 = \frac{5/3}{(4/9+4/9+1/9)}, \alpha_3 = 0 //$$

\* Note: Orthonormal set of  $n$  vectors always form a basis, But not all bases are orthonormal

# \* Discrete Fourier Transform:

## \* Fourier Basis

$$V_k[n] = e^{j \frac{2\pi k n}{N}}$$

$n \in [0 \dots N-1]$

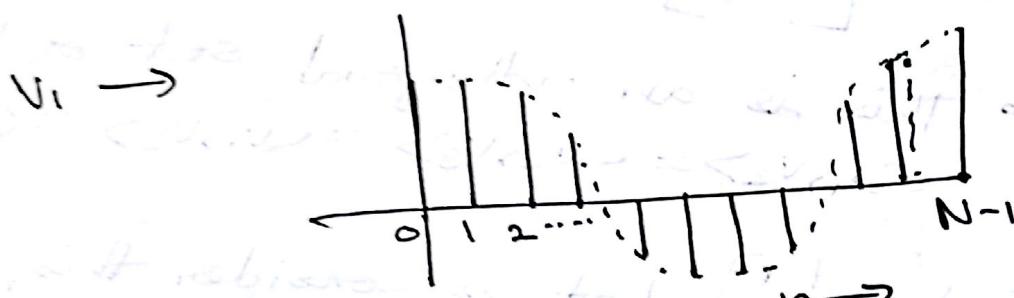
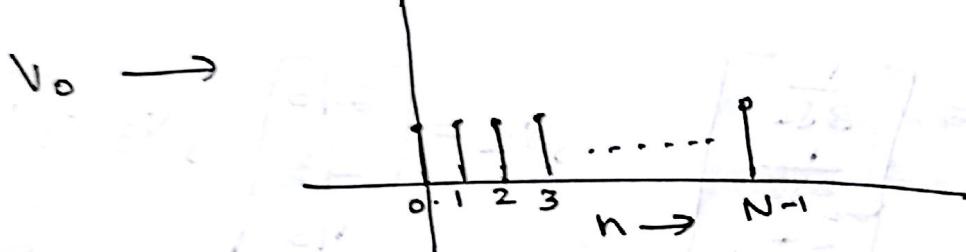
$k \in [0 \dots N-1]$

⇒ We have a 2D matrix.

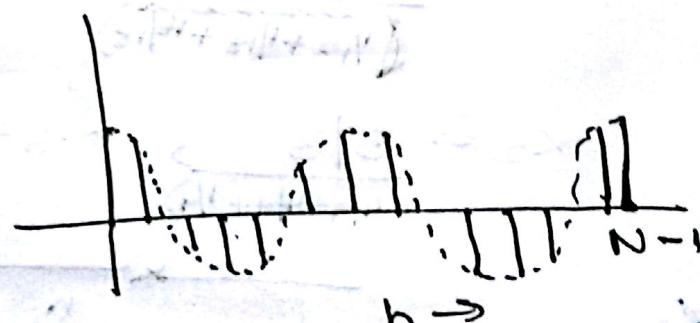
$$V_0 = e^{j \frac{2\pi \times 0 \times n}{N}} = [1 \ 1 \ 1 \ 1 \ \dots \ ]_N$$

$$V_1 = e^{j \frac{2\pi \times 1 \times n}{N}} = \left[ e^{j \frac{2\pi n}{N}} \ e^{j \frac{4\pi n}{N}} \ e^{j \frac{6\pi n}{N}} \ \dots \right]_N$$

\* Plotting the real parts,



$V_2 \rightarrow$  Similar to  $V_1$ , but we will form twice as many oscillations as before.



Now consider, any  $v_k, v_l$   
 $\Rightarrow$  we show  $\langle v_k, v_l \rangle = 0$  or  $\langle v_k, v_l \rangle = N$

$$\begin{aligned} \langle v_k, v_l \rangle &= \sum_{n=0}^{N-1} v_k[n] \cdot \overline{v_l[n]} \\ &= \sum_{n=0}^{N-1} e^{j \frac{2\pi k n}{N}} \cdot e^{-j \frac{2\pi l n}{N}} \\ &= \sum_{n=0}^{N-1} e^{j \frac{2\pi (k-l) n}{N}} \end{aligned}$$

Case ①,

$$k=l, \quad \langle v_k, v_l \rangle = N$$



Case ②,

$$k \neq l$$

$$1 + a + a^2 + \dots + a^{N-1}$$

$$\text{where } a = e^{j \frac{2\pi (k-l)}{N}}$$

$$\begin{aligned} \therefore \text{G.P sum} \Rightarrow \frac{1-a^N}{1-a} &= \frac{1-e^{j \frac{2\pi (k-l)}{N}}}{1-e^{j \frac{2\pi (k-l)}{N}}} \\ &= 0 // \end{aligned}$$

$\because k-l$  is an integer.

$\therefore$  We have found a basis of  $N$  vectors for the vector space of  $N$ -dimensional vectors. These vectors may also contain complex values.

\*  $\Rightarrow$  So any signal (vector of  $N$  dimensions) can be expressed as a linear combination of this basis.

From the theorem of orthogonal decomposition,

$$x = \sum_{k=0}^{N-1} X_k \cdot v_k, \quad X_k = \frac{\langle x, v_k \rangle}{\langle v_k, v_k \rangle}$$

$\Rightarrow$

$$x[n] = \sum_{k=0}^{N-1} X_k \cdot e^{j \frac{2\pi k n}{N}}$$

$$X_k = \frac{1}{N} \sum_{n=0}^{N-1} x[n] \cdot e^{-j \frac{2\pi k n}{N}}$$

Inverse discrete Fourier transform

Discrete Fourier transform

In some books this normalization may change.

- Using DFT, we can get the weightages (weights) of the basis signals / frequencies.

## DFT as Matrix Product:

$$\begin{bmatrix} X_0 \\ X_1 \\ \vdots \\ X_{N-1} \end{bmatrix} = \frac{1}{N} \begin{bmatrix} 1 & 1 & 1 & \cdots & 1 \\ 1 & \omega & \omega^2 & \cdots & \omega^{N-1} \\ 1 & \omega^2 & \omega^4 & \cdots & \omega^{2(N-1)} \\ \vdots & \vdots & \vdots & \ddots & \vdots \\ 1 & \omega^{(N-1)} & \omega^{2(N-1)} & \cdots & \omega^{(N-1)^2} \end{bmatrix} \begin{bmatrix} x[0] \\ x[1] \\ \vdots \\ x[N-1] \end{bmatrix}$$

Fourier transform  
of signal

$$e^{-j \frac{2\pi k n}{N}} = (\omega)^{kn}$$

$$\omega = e^{-j \frac{2\pi}{N}}$$

$\omega_N$

Note: Fourier transform:  $X = \omega \cdot x$

\*  $x(t) = 5 + 2\cos(2\pi t + \frac{\pi}{2}) + 3\cos(4\pi t)$

\*  $\xrightarrow{\text{frequency of terms increases}}$

(Now  $\omega t = 2\pi t$ ,  $\omega t = 4\pi t$ ) as we go to the right

$\Rightarrow \omega = 2\pi$ ,  $\omega = 4\pi$

$f = 1$ ,  $f = \frac{3}{2}$  (Max frequency  $\times 2$ )

$\Rightarrow$  So we choose 4 samples in a second

$t=0, t=\frac{1}{4}, t=\frac{1}{2}, t=\frac{3}{4}$

$x[0] = 8$   $\xrightarrow{\text{At } t=0}$

At  $t=\frac{1}{4} \left\{ x[1] = 5 + 2 - 3 = 4 \right.$

$x[2] = 5 + 2 \times 0 + 3 = 8$

$x[3] = 0$

The first 4 entries in our sample.

$$\begin{bmatrix} X_0 \\ X_1 \\ X_2 \\ X_3 \end{bmatrix} = \frac{1}{4} \begin{bmatrix} 1 & 1 & 1 & 1 \\ 1 & -j & -1 & j \\ 1 & -1 & 1 & -1 \\ 1 & j & -1 & -j \end{bmatrix} \begin{bmatrix} 8 \\ 4 \\ 8 \\ 0 \end{bmatrix} = \frac{1}{4} \begin{bmatrix} 20 \\ -4j \\ 12 \\ 4j \end{bmatrix} = \begin{bmatrix} 5 \\ -j \\ 3 \\ j \end{bmatrix}$$

$\Rightarrow X_K = X_{N-K}^*$   $\left\{ \text{For } K=1,2,\dots,N-1 \right.$

$\downarrow$

Proof not needed

$\therefore$  If  $K=0$ ,  $X_N$  does not exist as of now

\* Note: Each of the terms in our given signal corresponds to  $X_K$  &  $X_{N-K}$ . (order according to frequency)

- So in the above example, 5 is given by  $|X_0|$ , 2 is given by  $|X_1|$  &  $|X_3|$ , 3 is given by  $|X_2|$ . (These are the amplitudes)
- Now the phase for each part is given by the first occurrence phase i.e.  $X_K$ 's phase.

## \* Fourier Transform:

$$X(f) = \int_{-\infty}^{\infty} x(t) \cdot e^{-jx2\pi ft} \cdot dt$$

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

## \* Discrete Time Fourier Transform: (DTFT)

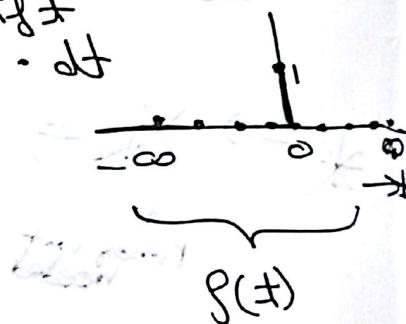
$$X[j\omega] = \sum_{n=-\infty}^{\infty} x[n] \cdot e^{-j\omega n}$$

## \* Functions:

### 1) Delta: ( $\delta(t)$ )

$$X(f) = \int_{-\infty}^{\infty} \delta(t) \cdot e^{-jx2\pi ft} \cdot dt$$

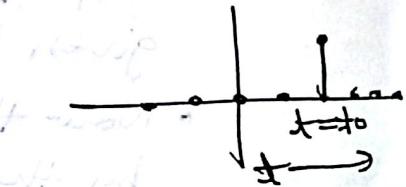
$$X(f) = 1$$



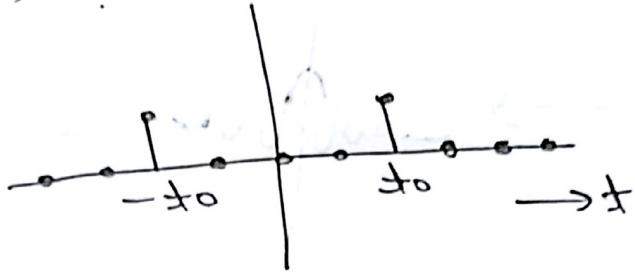
### 2) Delta Shift ( $\delta(t-t_0)$ )

$$X(f) = \int_{-\infty}^{\infty} \delta(t-t_0) \cdot e^{-jx2\pi ft} \cdot dt$$

$$X(f) = e^{-jx2\pi f t_0}$$



### 3) Function with 2 impulses:

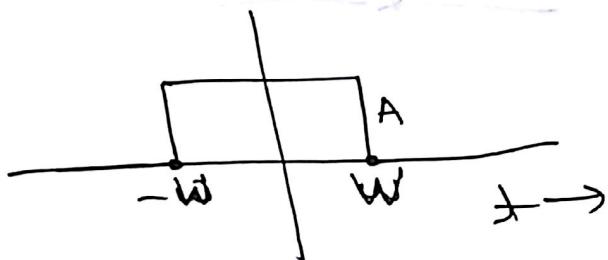


$$X(\delta) = e^{-j\omega t_0} + e^{j\omega t_0} = 2 \cos(\omega t_0)$$

$\downarrow$   
 $2\pi f$

(We can generate a  
cos function by using  
Fourier transforms on  
symmetric impulses)

\* 4)



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

$$= \int_{-w}^{w} A e^{-j\omega \underline{t}} \cdot dt$$

$$= A \int_{-w}^{w} e^{-j\omega t} \cdot dt$$

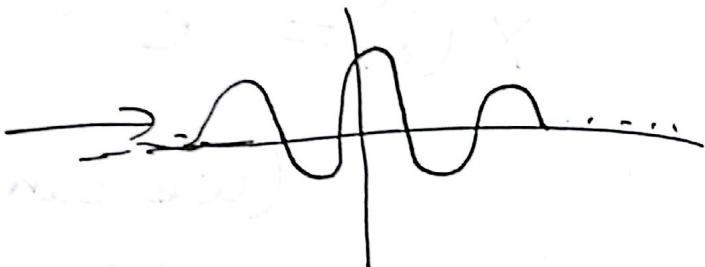
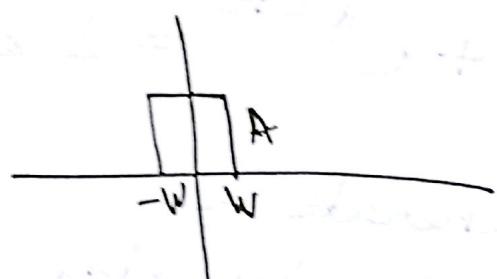
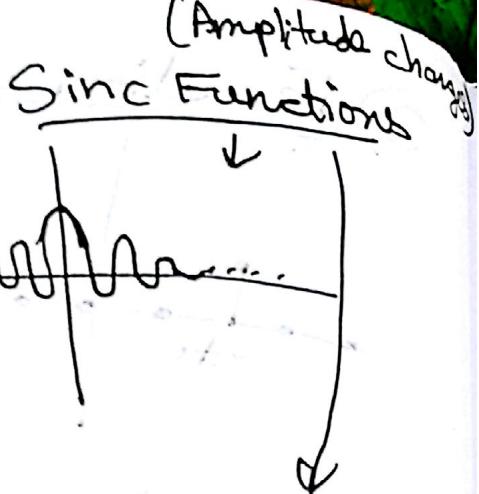
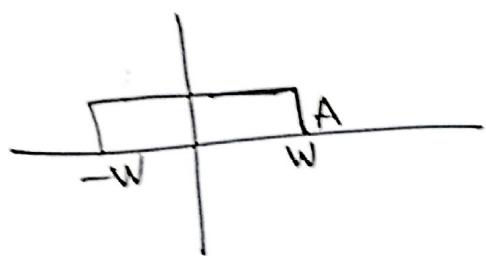
$$(\omega = 2\pi f)$$

$$= -\frac{A}{j\omega} [e^{-j\omega w} - e^{+j\omega w}]$$

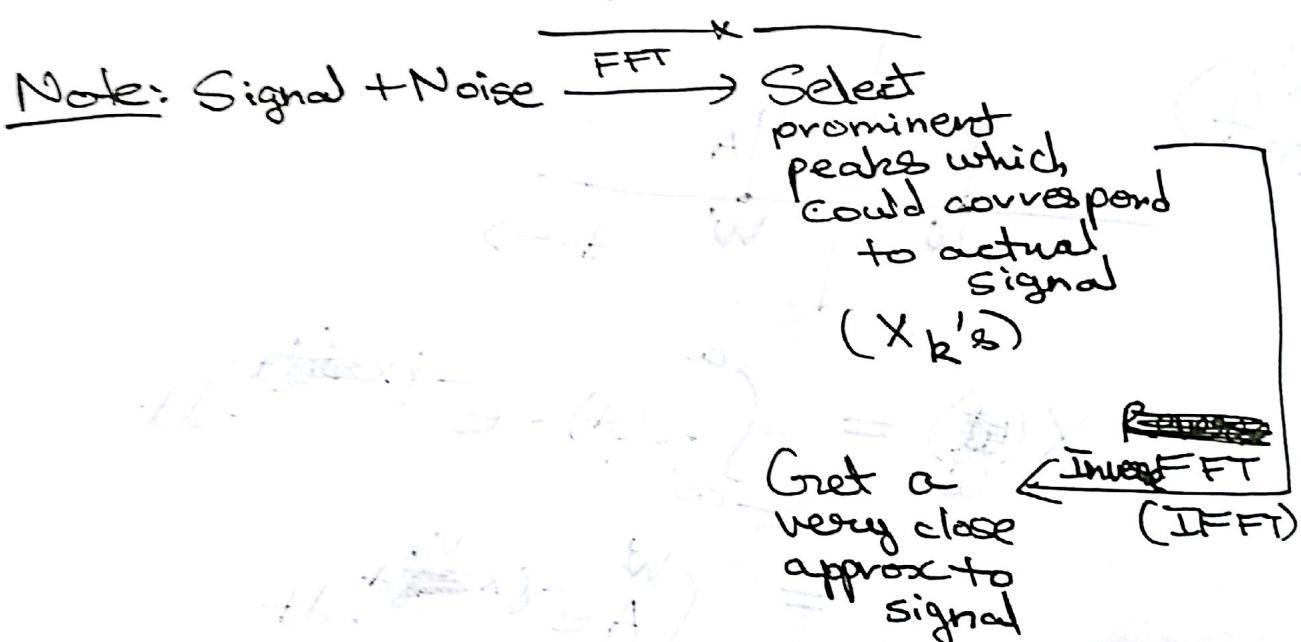
$$= \frac{A}{j\omega} [2j \sin(\underline{\omega w})]$$

$$\therefore X(\delta) = \frac{A \times j}{2\pi f} (\sin(\omega \times 2\pi f)) = \frac{A \sin(\omega \times 2\pi f)}{\pi f}$$

So we can see,



(Depending on  $A$  &  $W$ , we can see how our Fourier transform varies)



# \* Fast Fourier Transform: (FFT)

- DFT takes  $N^2$  operations to calculate values.
- FFT will only take  $N \log N$  steps.

$$X_k = \sum_{n=0}^{N-1} x[n] \cdot e^{-\frac{2\pi j k x_n}{N}},$$

i) Put  $K = \frac{N}{2}k_1 + k_0$

$$0 \leq k_0 \leq \frac{N}{2} - 1,$$

$$k_1 = 0 \text{ or } 1$$

$$\Rightarrow X_k = \sum_{n=0}^{N-1} x[n] \cdot e^{-\frac{2\pi j (\frac{N}{2}k_1 + k_0)n}{N}}$$

$$= \sum_{n=0}^{N-1} x[n] \cdot e^{-\frac{2\pi j \times \frac{N}{2} k_1 n}{N}} \cdot e^{\frac{-2\pi j k_0 n}{N}}$$

$$= \sum_{n=0}^{N-1} x[n] \cdot e^{-\pi k_1 n j} \cdot e^{\frac{-2\pi k_0 n j}{N}}$$

$$= \sum_{n=0}^{N-1} x[n] \cdot (-1)^{k_1 n} \cdot e^{\frac{-2\pi k_0 n j}{N}}$$

ii) Now, split n into even & odd indices

$$(2n), (2n+1)$$

$$\therefore X_k = \sum_{n=0}^{\frac{N}{2}-1} x[2n] (-1)^{k_1 \times 2n} \cdot e^{\frac{-2\pi j k_0 (2n)}{N}} \\ + \sum_{n=0}^{\frac{N}{2}-1} x[2n+1] (-1)^{k_1 (2n+1)} \cdot e^{\frac{-2\pi j k_0 (2n+1)}{N}}$$

# \* Fast Fourier Transform (FFT)

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$$X_k = \sum_{n=0}^{N-1} x[n] \cdot e^{-\frac{2\pi j k x n}{N}}$$

i) Put  $k = \frac{N}{2}k_1 + k_0$

$$0 \leq k_0 \leq \frac{N}{2} - 1,$$

$$k_1 = 0 \text{ or } 1$$

$$\Rightarrow X_k = \sum_{n=0}^{N-1} x[n] \cdot e^{-\frac{2\pi j (\frac{N}{2}k_1 + k_0)n}{N}}$$

$$= \sum_{n=0}^{N-1} x[n] \cdot e^{-\frac{2\pi j \times \frac{N}{2} k_1 n}{N}} \cdot e^{-\frac{2\pi j k_0 n}{N}}$$

$$= \sum_{n=0}^{N-1} x[n] \cdot e^{-\pi j k_1 n} \cdot e^{-\frac{2\pi j k_0 n}{N}}$$

$$= \sum_{n=0}^{N-1} x[n] \cdot (-1)^{k_1 n} \cdot e^{-\frac{2\pi j k_0 n}{N}}$$

ii) Now, split n into even & odd indices  
 $(2n), (2n+1)$

$$\therefore X_k = \sum_{n=0}^{\frac{N}{2}-1} x[2n] (-1)^{k_1 \times 2n} \cdot e^{-\frac{2\pi j k_0 (2n)}{N}} + \sum_{n=0}^{\frac{N}{2}-1} x[2n+1] (-1)^{k_1 (2n+1)} \cdot e^{-\frac{2\pi j k_0 (2n+1)}{N}}$$

$$x_k = \sum_{n=0}^{N/2-1} x[2n] \cdot e^{-\frac{2\pi j kn}{N/2}}$$

$$+ (-1)^{k_1} \sum_{n=0}^{N/2-1} x[2n+1] \cdot e^{\frac{-2\pi j kn}{N/2}} \cdot e^{\frac{-2\pi j k_1 n}{N/2}}$$

$$x_k = \sum_{n=0}^{N/2-1} x[2n] \cdot e^{-\frac{2\pi j kn}{N/2}}$$

$$+ (-1)^{k_1} \cdot e^{\frac{-2\pi j k_0}{N}} \sum_{n=0}^{N/2-1} x[2n+1] \cdot e^{\frac{-2\pi j k_1 n}{N/2}}$$

~~$x_{k_0} = F_0 + QF_1$~~

Now,  ~~$x_{k_0}$~~  i.e.  $x_{(k_0)} = F_0 + QF_1$   $\xrightarrow{\text{FFT of } N/2 \text{ elements}}$   $\xrightarrow{\text{FFT of } N/2 \text{ elements}}$

\* \* \*  $x_{(N/2+k_0)} = F_0 - QF_1$

where  $Q = e^{\frac{-2\pi j k_0}{N}}$

( $F_0$  = DFT of even terms,  
a total of  $N/2$  terms)

( $F_1$  = DFT of odd terms,  
a total of  $N/2$  terms)

To calculate DFT of  $N$ , we can recursively break it down into parts of size  $N/2$  each time.

i.e.  $T_N = 2T_{N/2} + 2N$

$\therefore T_N = N \log N$

$\uparrow$   
 $N/2$  multiplications  
 $N/2$  additions  
for  $k_1 = 0$  or  
 $k_1 = 1$

# \* 2D DFT:

$$\hat{A} = \begin{bmatrix} W_M \\ & M \times M \end{bmatrix} \begin{bmatrix} A \\ & M \times N \end{bmatrix} \begin{bmatrix} W_N \\ & N \times N \end{bmatrix}$$

$$* \hat{A}_{k,l} = \sum_{m=0}^{M-1} \sum_{n=0}^{N-1} A[m,n] e^{-j \times 2\pi \left( \frac{kn}{N} + \frac{lm}{M} \right)}$$

$\equiv \underline{\underline{x}} \underline{\underline{}}$

\* Convolution is multiplication of Fourier Transforms:

\* Let  $y[n] = x[n] \star h[n]$ , let's prove,

$$Y[\omega] = X[\omega] \cdot H[\omega] \rightarrow \text{DTFT's}$$

(For  $\infty$  length signals)

Proof:

$$X_\omega = \sum_{n=-\infty}^{\infty} x[n] \cdot e^{-j\omega n}$$

$$Y_\omega = \sum_{n=-\infty}^{\infty} (x[n] \star h[n]) \cdot e^{-j\omega n}$$

$$\text{but } x[n] \star h[n] = \sum_{k=-\infty}^{\infty} x[k] \cdot h[n-k]$$

$$\Rightarrow Y_\omega = \sum_{n=-\infty}^{\infty} \left( \sum_{k=-\infty}^{\infty} x[k] \cdot h[n-k] \right) \cdot e^{-j\omega n}$$

$$\text{Let } m = n - k, n = m + k$$

$$\Rightarrow Y_\omega = \sum_{m=-\infty}^{\infty} \left( \sum_{k=-\infty}^{\infty} x[k] \cdot h[m+k] \right) \cdot e^{-j\omega(m+k)}$$

$$= \sum_{m=-\infty}^{\infty} h[m] \cdot e^{-j\omega m} \cdot \sum_{k=-\infty}^{\infty} x[k] \cdot e^{-j\omega k}$$

$$\therefore y(w) = H(w) \cdot \underline{\underline{x(w)}}$$

## \* Circular Convolution: (★)

- ~~$x[n]$~~  circular conv  $H[n]$  unlike before  
then  
 $\Rightarrow$  we ~~don't~~ need to flip  $H[n]$ , we start as it is, then we circularly rotate  $H[n]$  to get  $y[n]$

ex:  $x_1[n] = [2 \ 1 \ 2 \ 1]$   ~~$x_1[n]$~~   
 $h[n] = [1 \ 2 \ 3 \ 4] \Rightarrow h[n] = [4 \ 1 \ 2 \ 3]$

 $y[n] = [14 \ 16 \ 14 \ 16]$   $h[n] = [3 \ 4 \ 1 \ 2]$   
 $y[n] = x_1[n] \star h[n]$   $\downarrow$   
 $\dots$

\*  $\Rightarrow y[n] = x_1[n] \star x_2[n]$  From prev proof  
 $= \text{IDFT}(\text{DFT}(x_1[n]) \cdot \text{DFT}(x_2[n]))$  Inner product (without sum)

\*  $\Rightarrow x_1[n] \star x_2[n] = \text{IDFT}(\text{DFT}(x_1[n]) \cdot \text{DFT}(x_2[n]))$   
(Proof not needed)  
 $\underline{\underline{x}}$  Inner product (without sum)

Note: This is because DFT assumes our signal repeats over & over infinitely (from the formulas), so it pretty much acts as a circular shift.

Note: If we pad the signals with 0's, then for a certain number of 0's, circular convolution = normal convolution.

## \* Non Locality of DFT.

- We won't have information on where the peaks of DFT were present in the original signal.

⇒ We use windowing to solve this -

- Each window has a size  $M$  and we move it by  $n$  in each iteration.  $N = \text{signal size}$ .
- \* So, we iterate from  $k=0, 1, \dots, \frac{N-M}{n}$

⇒  $[x_{kn}, \dots, x_{kn+M-1}] \rightarrow \text{Data for each window.}$

## Spectrogram: (Audio uses this (Audio matching etc))

- Time vs Frequency. Greyscale is used to show major frequency content. Each block would represent a window.

# \* Filters:

- Note: When we have DFT of our noised signal, on multiplying it with a rectangular filter & taking IDTFT
  1. is similar to convolution of signal
  2. with a ~~sinc~~ 'sinc' type of function, so we will get ripples in our output.

\*  $\Rightarrow$  So for a good filter, we use a gaussian filter instead of a rectangular filter.  $\rightarrow$  IDTFT using this is like convolution of signal with a pretty much horizontal line function, so no ripples.

Note: Generally low frequency & high amplitude signals give rise to the overall wave (the actual one without noise). Don't forget the conjugate frequencies which also contribute to the overall wave.

$X_k$  &  $X_{n-k}$   $\Rightarrow$  Contribute to actual wave without noise.

Note: Salt Noise  $\rightarrow$  Min filter (To ignore white spots)  
Pepper Noise  $\rightarrow$  Max filter (To ignore dark spots)

Note: Low pass filter  $\rightarrow$  Approx shape of signal captured.  
High pass filter  $\rightarrow$  Details of signal captured

# Leaky Integrator

\* Note: Convolution of a signal with a causal filter,  $\rightarrow$  causal signal (output)

$\Rightarrow$  An averaging filter,

$$y_M[n] = \frac{1}{M} \sum_{k=0}^{M-1} x[n-k] \quad \rightarrow \textcircled{1}$$

$$\begin{aligned} y_M[n-1] &= \frac{1}{M} \sum_{k=0}^{M-1} x[n-1-k] \\ &= \frac{1}{M} \sum_{k=1}^M x[n-k] \end{aligned}$$

Similarly,

$$y_{M-1}[n-1] = \frac{1}{M-1} \sum_{k=1}^{M-1} x[n-k] \quad \rightarrow \textcircled{2}$$

$$\Rightarrow \sum_{k=0}^{M-1} x[n-k] = x[n] + \sum_{k=1}^{M-1} x[n-k]$$

$\Rightarrow$  From  $\textcircled{1}, \textcircled{2}$

$$M \cdot y_M[n] = x[n] + (M-1) \cdot y_{M-1}[n-1]$$

$$\begin{aligned} \Rightarrow y_M[n] &= \frac{x[n]}{M} + \frac{(M-1)}{M} y_{M-1}[n-1] \\ &= (1-\lambda)x[n] + \lambda y_{M-1}[n-1] \end{aligned}$$

Consider  $\lambda = \frac{M-1}{M}$

$\therefore y_M[n] = (1-\lambda)x[n] + \lambda y_{M-1}[n-1]$

$$\therefore \boxed{y[n] = (1-\lambda)x[n] + \lambda y[n-1]} , \lambda = \frac{M-1}{N}$$

- This is a leaky integrator, we approx that m is large, so  $y_m[n] \approx y_{m-1}[n]$
- $\therefore$  we get the leaky integrator above.

(In MATLAB this is done by the 'Filter' command)

\* Note: A normal integrator  $\rightarrow y[n] = \sum_{k=0}^n x[k]$

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

- Here we use  $\lambda y[n-1]$ ,  $\lambda < 1$  so we are leaking out some part, so its a leaky Integrator.

\* Impulse Response: (Leaky Integrator)

$$\bullet h[n] = (1-\lambda)S[n] + \lambda h[n-1]$$

$\Rightarrow$  For  $n < 0$ ,  $h[n] = 0 \Rightarrow$  System is off

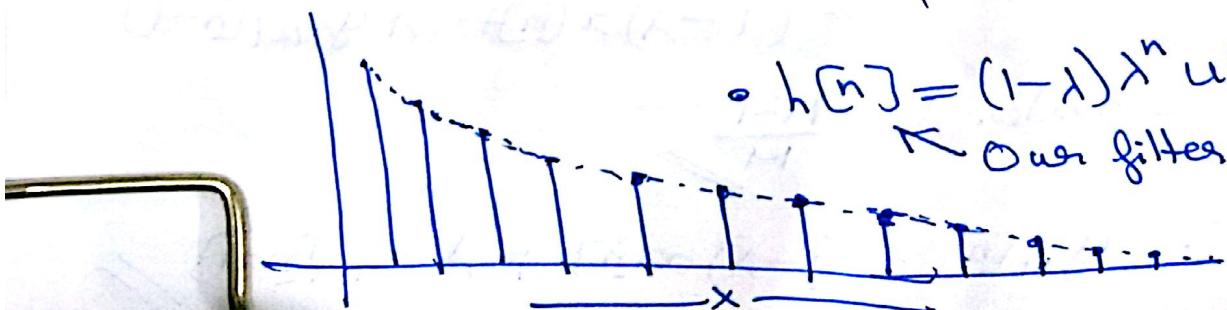
$$n=0, h[n] = 1-\lambda$$

$$n=1, h[n] = \lambda(1-\lambda)$$

$$n=2, h[n] = \lambda(\lambda(1-\lambda)) = \lambda^2(1-\lambda)$$

$$\bullet h[n] = (1-\lambda)\lambda^n u[n]$$

↖ Our filter





# \* Huffman Coding: (Variable length encoding)

- Suppose a string has

A → 8 times

! → 3 times

B → 2 times

R → 2

— → 2

K → 1

D → 1

- Prefixed free encoding for each character is generated.

- For any encoding we can define compression ratio as  $\frac{\text{actual size}}{\text{compressed size}}$

- Huffman tree with variable length encoding is formed iteratively by collecting the 2 smallest numbers & merging them at each stage.

$$\begin{aligned}
 \rightarrow A &= 0 \dots = 110 \\
 ! &= 100 \quad K = 1110 \\
 B &= 1010 \quad D = 1111 \\
 R &= 1011
 \end{aligned}$$

} Prefix Code

Now we can use this to encode.

Scanned by CamScanner

## Self Energy: (I)

- $I = \log_2 \left(\frac{1}{P}\right) = -\log_2 P$

( $P$  = Probability of occurrence)

- As  $P \uparrow$ ,  $I \downarrow$

## Entropy: (E)

- $E = - \sum_{i=0}^{L-1} p_i \log_2(p_i)$

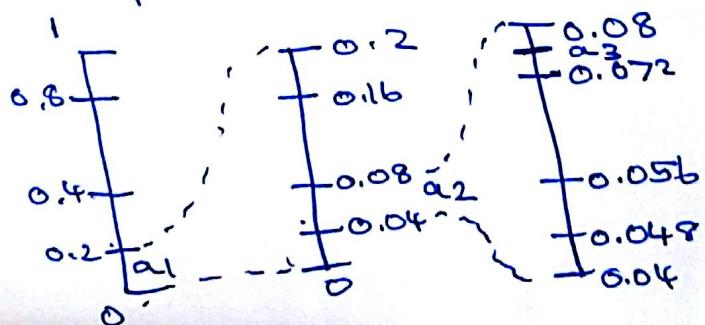
\* \* \*  $\Rightarrow$  Shannon said that the maximum amount of compression we can achieve =  $E$ . (Entropy).  $\rightarrow \frac{\text{No. of Bits}}{\text{No. of chars}}$ .

(This can be broken by some carefully crafted strings) \* \*

## \* Arithmetic Coding:-

- \* Given a string, we encode it just using a number between 0 & 1.

ex:  $S = \{a_1, a_2, a_3, a_4\}$  ? String to form is  
 $P = \{0.2, 0.2, 0.4, 0.2\}$   $a_1 a_2 a_3$



$\Rightarrow$  Now just store the midpoint,  
 $\Rightarrow \frac{0.072 + 0.08}{2}$   
 $= 0.076 //$

\* Now to decode it, we go through the same process, and at each step, check which interval our number lies in.

⇒ We get  $a_1 a_2 a_3 \dots$

\* To mark the end of the string, we can either add an EOM char & stop when we encounter it or just send the length of the string along as well, and use it to stop.

\_\_\_\_\_

\* LZW Coding: (Example in slides)  
(No need to store any dictionary)

- \* We augment another bit, and then instead of just ASCII encoding we use this new encoding where we use the extra 256 possibilities to make our own dictionary. ( $256 \rightarrow 511$ )
- \* We then send this dictionary so the decoding can be done. But this is expensive, so we follow a similar procedure while decoding where we go ahead building a similar sort of dictionary.

\* Note: While decoding, if we get current & next as  $xy$  where  $y > 255$ , so that would mean  $y$  consists of 2 chars, so we will only consider the first char of our dict[y].

We add to our new dictionary,  
 $\text{dict}[x]\text{dict}[y][0]$ .

→ However, if  $x > 255$ , then we consider  
the entire  $x$ , unlike above.

Note: At times, generally 11 to 13 bits are used where the 5 bits are used for creation of our dictionary.

\_\_\_\_\_ x \_\_\_\_\_  
\_\_\_\_\_ x \_\_\_\_\_

## \* MP3 Compression:

Without compression,  $44100 \text{ Hz} \rightarrow$  44,100 samples in a second

$$\therefore 44100 \times 16 \times 60 = 42.3 \text{ mb} \quad \left\{ \begin{array}{l} \text{In a} \\ \text{minute} \end{array} \right.$$

$2^{16}$  datapoints  
to show  
value of (frequency)  
each reading's DFT.

a) Since we humans are more sensitive to a certain range of frequencies.

i) Take DFT of signal

2) The sensitive part  $\rightarrow$  use a high level of quantization  
( $2k-5k+8$ ) bands (many bits)

\* 3) In other parts  $\rightarrow$  use less bits to quantize.  
(bands)

b) Temporal Masking:

- If we get a small sound after a large one, we can use less bits to quantize the small one since we can't hear it well anyways.

\* c) Frequency Masking: (It's the loudest in dB of all)

- When a certain frequency is playing, then we become less sensitive to nearby frequencies, so we can use fewer bits to quantize those nearby ones.

\* Note: All these frequency based compression are generally done using windows and DFT's on windows.

Classification → ~~for software~~ → ~~for~~ requires working  
→ ~~for~~ ~~for~~ ~~for~~

spotted side by side. There is a tiny white  
strip of soft green vine near me and  
through this I was able to get the  
spider to walk across the vine and  
over onto the leaf.

- \* c) Frequency Masking: (It's the loudest in dB as ↑)  
 When a certain frequency is playing, then we become less sensitive to nearby frequencies, so we can use fewer bits to quantize those nearby ones.

\* Note: All these frequency based compression are generally done using windows and DFT's on windows.

## \* JPEG Compression: (Lossy compression)

- 1) Type of coding
- 2) Spatial redundancy  
(Temporal redundancy → Videos)
- 3) Perceptual redundancy  
(We remove colors etc that are harder to perceive)

Ex: RGB → Y, Cb, Cr  
 (Y is like grayscale)  
 (Luminance)  
 (Cb, Cr → Color channels)

### \* I) $Y, C_b, C_r \rightarrow$ Perceptual redundancy

- Changes in  $Y^{\text{(luminance)}}$  → We can easily find out.  
 ⇒ Changes in  $C_b, C_r$  are not that noticeable. So we can compress these two channels.

We can convert an RGB image to a YCbCr image:

## \* Quality Measurement: (MSE, PSNR)

• SNR =

• PSNR =

Note: A great way to compress is by breaking the image into smaller parts since the redundancy of colours in these small parts is a lot more. So we can quantize better. (Generally each of these parts lie around one or two values of RGB & near them).

⇒ Parallelization can also be done.

\* How do we do it? 8x8 chunks

## \* JPEG - Method:

\* Input spatial or frequency domain

- Break image into  $8 \times 8$  chunks, on each ~~chunk~~ chunk subtract value by 127 (so we have a -ve to +ve range) → This is used by the DCT (Discrete cosine transform) (Implementation detail)
- We then use a DCT & then quantize.

Note: OCT,  $8 \times 8$  all these things were decided based on experimental results. These lead to the best results.  
 $(16 \times 16$  actually was best but  $8 \times 8$  was easier to implement).

2D matrix after OCT

Input

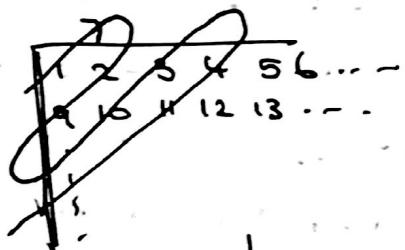
\* (This was setup by experiments asking users to rate images)

\* Quantization → There is a standard 2D matrix of  $8 \times 8$  cells which we use to quantize.

For each chunk

→ Each of our  $8 \times 8$  chunks, we divide the values with corresponding in the standard 2D matrix.

\* Encoding → Once we have quantized the image chunk, we use a zig-zag reading of the values in our matrix



→ This is done so that we get lots of repeating adjacent values (Good run length encoding can be done)

→ Once this is done, we will be left with an array (single)

.. (approx 75%) Generally a large chunk of cells are all 0's near the end, so we just represent it with EOB.

\* RLE or Huffman coding

Send message after coding it  
→ Encoding to Decoding

\* Decoding

Get back vector and reconstruct the matrix.

\* Dequantize: (We get the data loss here)

→ Multiply each cell with corresponding value in the standard 2D matrix.  
Once this is done we get back a similar matrix to the one we got after DCT-  
(But we have lost some values) by now.

Inverse DCT

→ Do an inverse DCT & then add back 127.

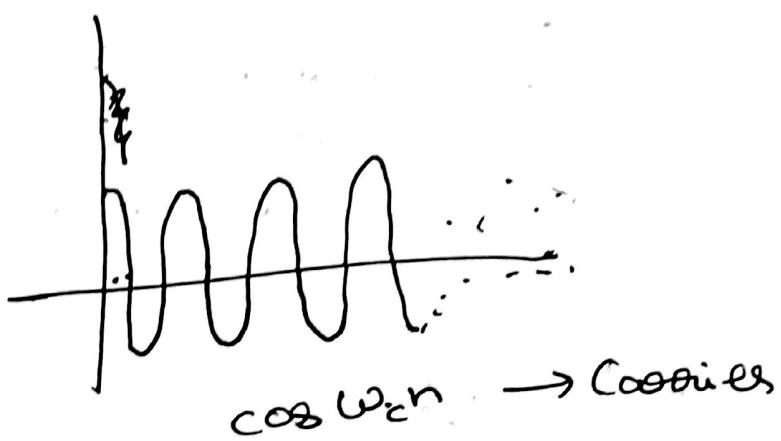
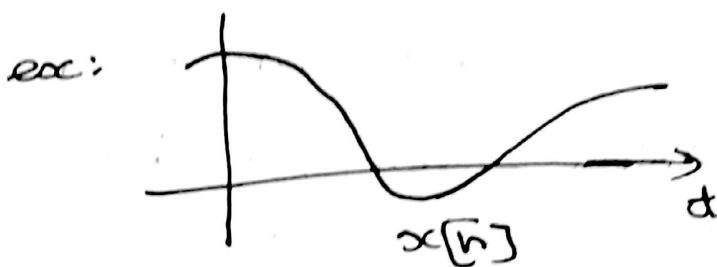
→ Then reassemble the sub parts (8x8 chunked) to get back the original image.

# Modulation

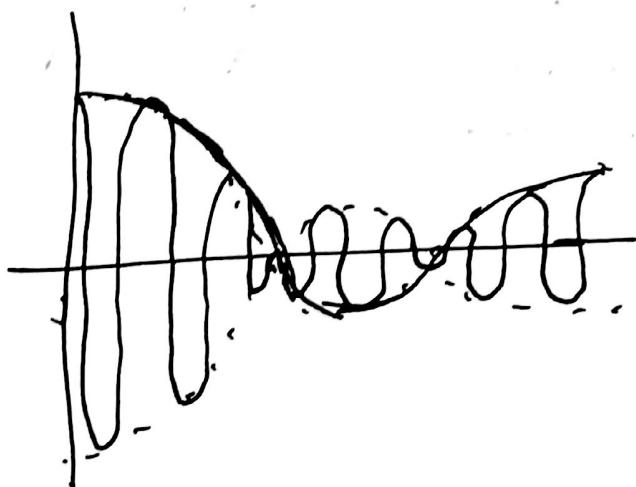
(To send signals  
to large distances)  $\rightarrow$  Radio wave  
carriers.

## \* I) Amplitude Modulation:

- We use a carrier wave by multiplying it with the signal to send it along.



$$s[n] = x[n] \cdot \cos w_c n$$



$\Rightarrow$  How do we get back the original signal.

\* 1) At receiver we have,

$$y[n] = x[n] \cos(\omega_c n)$$

We multiply it with  $\cos(\omega_c n)$

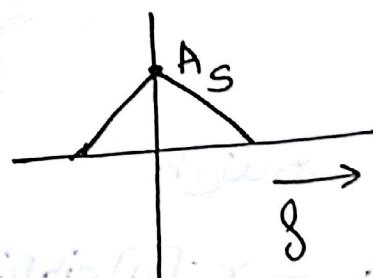
$$\rightarrow x[n] \cdot \cos^2(\omega_c n)$$

$$y[n] = x[n] \left[ \frac{1 + \cos(2\omega_c n)}{2} \right]$$

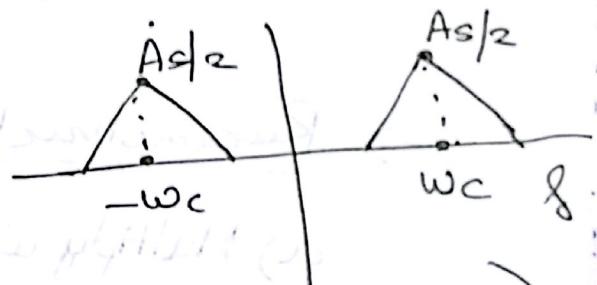
$$y[n] = \frac{x[n]}{2} + \frac{\omega_c}{2} x[n] \cos(2\omega_c n)$$

Then apply a low pass filter to neglect the 2nd term. Then multiply by 2 to get back  $x[n]$ .

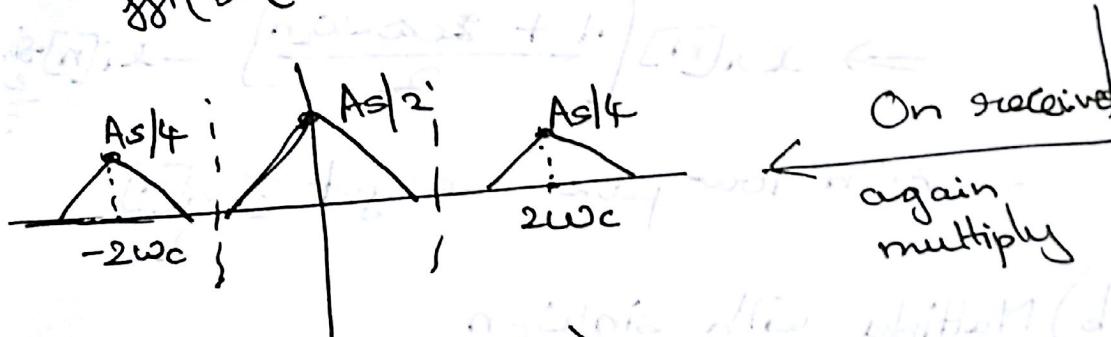
$\Rightarrow$  In frequency domain, (with fft shift)



Multiplying  
original  
signal  
with  
 $\cos(\omega_c n)$



fft( $x[n] \cos(\omega_c n)$ )



fft( $x[n] \cos^2(\omega_c n)$ )

$\Rightarrow$  Applying low pass filter & multiplying with 2, we get back the original signals fft.

fft( $x[n]$ )

2) If our signal is a complex signal

\*

⇒ We send multiplying it with  $e^{j\omega_n}$

$$\Rightarrow x[n] = x_R[n] + x_i[n] \cdot j$$

• We send  $\operatorname{Re}\{x[n] \cdot e^{j\omega_n}\}$

$$\Rightarrow \operatorname{Re}\{x_R[n] + jx_i[n]\} \cdot (\cos \omega_n + j \sin \omega_n)$$

$$y[n] = x_R[n] \cos \omega_n - x_i[n] \sin \omega_n$$

⇒ We send this over

⇒ Reconstruction:

a) Multiply with ~~cos~~  $\cos \omega_n$

$$\Rightarrow x_R[n] \cos^2 \omega_n - x_i[n] \sin \omega_n \cos \omega_n$$

$$\Rightarrow x_R[n] \left[ \frac{1 + \cos 2\omega_n}{2} \right] - x_i[n] \frac{\sin 2\omega_n}{2}$$

→ On low pass we get  $\frac{x_R[n]}{2}$

b) Multiply with  $\sin \omega_n$

$$\Rightarrow x_R[n] \sin \omega_n \cos \omega_n - x_i[n] \cdot \frac{\sin^2 \omega_n}{2}$$

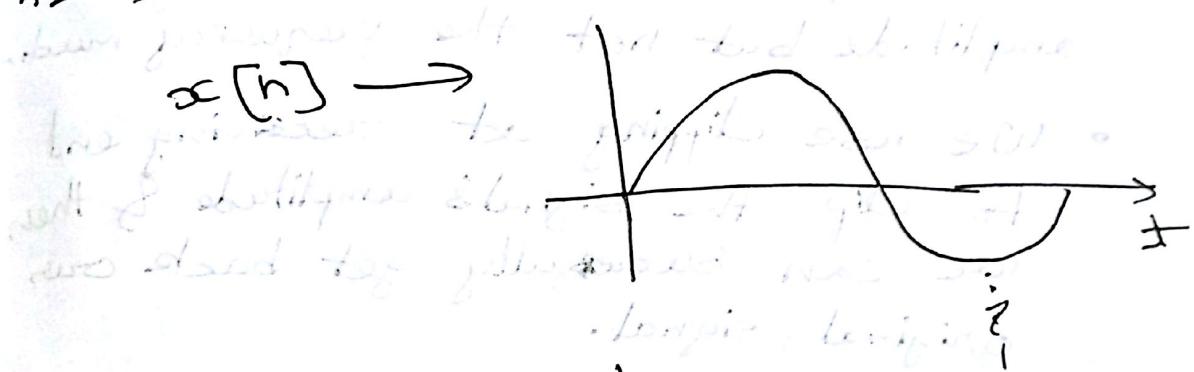
$$\Rightarrow x_R[n] \cdot \frac{\sin 2\omega_n}{2} - x_i[n] \left( \frac{1 - \cos 2\omega_n}{2} \right)$$

⇒ On low pass we  
get  $\frac{x_i[n]}{2}$

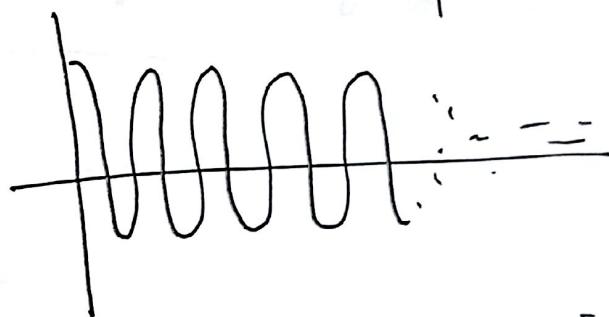
\* Note: Different bands ( $\omega_c$ ) are allocated & when we send them, we also do a band pass filtering instead of low pass so that we get our signal at our band ( $\omega_c$ ).  $\xrightarrow{\text{using - filter}}$

\* Note: In AM signals, noise can modify the signal being sent so we can end up having varied amplitudes. (Noise). So we switch to frequency modulated signals.  $\xrightarrow{\text{switching to F.M.}}$

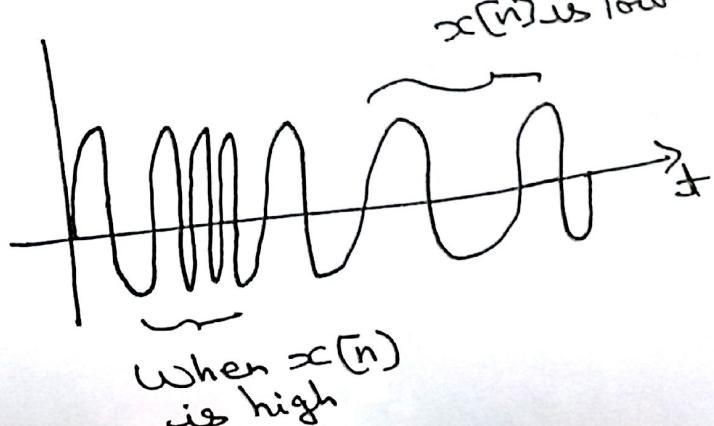
## \* II) Frequency Modulation:



Carrier  $\rightarrow$



$s[n] \rightarrow$



$$\cdot x[n] = A \sin(\omega_x n)$$

$$c[n] = A \sin(\omega_c n)$$

$$\Rightarrow s[n] = A \sin(2\pi(f_c + A \sin \omega_x n)n)$$

- We send a signal hidden in the frequency.

— x —

\* Note: To get back the original signal, there are ways to figure out frequency at a certain given point?

— x —

### Noise :

- Generally this noise effects only the amplitude but not the frequency much.
- We use clipping at receiving end to clip the signal's amplitude & then we can successfully get back our original signal.

— x —

# Dither

- Noise can be helpful at times to help preserve information while quantizing etc.

⇒ Audio we add noise twice (Like <sup>sum of</sup> 2 dice, so a lot more cases produce the average →  $\bar{x}$ )

(Ex: For images instead of colour, we can use intensity to show information (using dithers))

## \* Stochastic Signal Processing

- Spectral Power Density  $\Rightarrow ?$  (SPD)

$$\hookrightarrow \sum_{k=1}^M \frac{1}{(2N+1)} \sum_{n=-N}^N |x[n]|^2 \rightarrow \overline{|x[n]|^2}$$

Practically we consider some large  $N$ .  $\rightarrow M$  experiments, so we average our results.

⇒ Energy doesn't tell us much but power (SPD) does.

- Filtering in time domain also affects the SPD

## \* Principle Component Axis (PCA)

Note: Variance =  $\frac{\sum (x_i - \bar{x})^2}{n-1}$

\* Covariance ( $H, M$ ) =  $\frac{\sum (H_i - \bar{H})(M_i - \bar{M})}{N-1}$

- Similarity b/w two datasets, if one is increasing & other is decreasing covariance  $\Rightarrow$  -ve.
- If both are increasing or decreasing, its covariance  $\Rightarrow$  +ve.

- We can use PCA for camera angles etc
- \* to capture maximum information from a 3D point cloud.  $\rightarrow$  the PCA helps us

\* Principle component axis

find the maximum variance axis.

Points when projected to that axis have maximum variance

## \* Eigen Vectors / Values: (ex, in slides)

- $\Sigma e = \lambda e \rightarrow ①$ ,  $\lambda$  = Eigen values  
 $e$  = Eigen vector.

$e_n^{-1} = e_n^T \quad \} \text{The eigen vector matrix} \rightarrow e_n = [e_1 | e_2 | \dots | e_n]$

$$\Rightarrow \Sigma(e) - \lambda e = 0$$

$$\Rightarrow (\Sigma - I\lambda)e = 0$$

- This is only possible when  $|\Sigma - I\lambda| = 0$
- From this we can get the two eigen values

$\Rightarrow$  On substituting each eigen value in (1),  
 \* we can get the corresponding eigen  
 \* vectors.  $\rightarrow$  Eigen vectors should be  
 unit vectors

\* Now if we use  $\Sigma =$  Covariance matrix

$$\text{Cov matrix} = \begin{bmatrix} \text{cov}(1,1) & \text{cov}(1,2) & \dots & \text{cov}(1,N) \\ \text{cov}(2,1) & \text{cov}(2,2) & \dots & \vdots \\ \vdots & \vdots & \ddots & \vdots \\ \vdots & \vdots & \vdots & \text{cov}(N,N) \end{bmatrix}$$

If we consider N datasets  
 ↳ Like coordinate axes  $x_1, x_2, x_3$

- Now if we calculate the eigen vectors, we get the maximum variance axis.

\*  $\rightarrow$  Use the top eigen vector corresponding to the max eigen value

Other eigen vectors are orthogonal  
 ↳ This is the principle component axis.

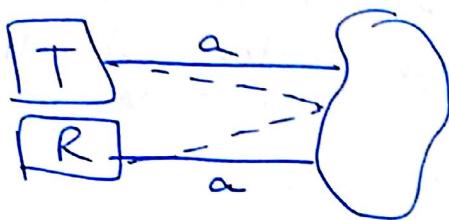
\* Note: Multiplying points with an eigen vector, we get the points on the new axis. (Assuming the eigen vector to be the new axis)

# RADAR

(Radio Detection & Ranging)

- In radio waves

$$t = \frac{2a}{c}$$



, Sampling  $\rightarrow$  Best resolution in time is 44.1 KHz

$\Rightarrow \frac{1}{44100} \text{ s}$  minimum

$$\Rightarrow \frac{1}{44100} = \frac{2a}{3 \times 10^8}$$

$$\therefore a \approx 3.9 \text{ KM}$$

- So by the time we can take another sample, the object must have ~~travelled~~ about 4 KM from the Transmitter & Receiver.
- So we cannot detect objects nearer than 4 KM.

$\Rightarrow$  Doppler's Effect:

$$F = \left( \frac{c}{c + v_s} \right) \times F_0 \quad \left. \begin{array}{l} \text{Source moving} \\ \text{towards receiver} \end{array} \right\}$$

$$F = \left( \frac{c + v_r}{c + v_s} \right) \times F_0 \quad \left. \begin{array}{l} \text{Both velocities} \\ \text{in the same axis.} \\ \text{-----} \end{array} \right\}$$

→ ①

$$\Rightarrow F = \frac{\left(1 + \frac{v_2}{c}\right)}{\left(1 + \frac{v_3}{c}\right)} \times F_0 = \left(1 + \frac{v_2}{c}\right) \left(1 - \frac{v_3}{c}\right) \times F_0$$

$$\therefore \frac{1}{1-v_3} \approx 1+v_2 \quad \left\{ \text{From Cn.P} \right.$$

$$\therefore F = \left(1 - \frac{v_3}{c} + \frac{v_2}{c} - \frac{v_2 v_3}{c^2}\right) \times F_0$$

$$\Rightarrow F - F_0 \approx \frac{(v_2 - v_3)}{c} \times F_0$$

$$\therefore \Delta f = \frac{\Delta v}{c} \times f_0 \rightarrow (2)$$

\* Now when we send out a signal at some  $f$ , say  $f = 2.4 \text{ GHz}$

$\Rightarrow$  On coming back, we observe

$$f' = f \times \left(\frac{c-v}{c+v}\right), v = \text{velocity}$$

$$\approx f \times \left(1 - \frac{v}{c}\right)^2 \quad \begin{matrix} \text{of moving} \\ \text{object} \end{matrix}$$

$$f' \approx f \times \left(1 - \frac{2v}{c}\right)$$

$\therefore$  We have  $\Delta f$  &  $f_0$ , so we can get  $\Delta v$ .

$$\therefore \Delta v = v_2 - v_3 \quad \text{since}$$

$$\Delta v = v_2 \quad \because v_3 \text{ is } 0.$$

$\Rightarrow$  The problem here is its hard to ~~get~~ differentiate  $f'$  from  $f$  when we plot the DFT.

$\Rightarrow$  Suppose our waves are cosines,

$$\bullet \text{Source} = \cos \omega_0 t = \cos 2\pi f_0 t$$

$$\text{Recd} = \cos \omega t = \cos 2\pi f t$$

\* On multiplying the two signals,

$$\cos \omega_0 t \cdot \cos \omega t = \cos(\omega + \omega_0)t + \cos(\omega - \omega_0)t$$

$\Rightarrow$  So on plotting the FFT, the peak at  $\omega + \omega_0$  is huge so we can ignore it & consider

$$\Delta F = \frac{\omega_0 - \omega}{2\pi} \rightarrow \text{A peak is observed}$$

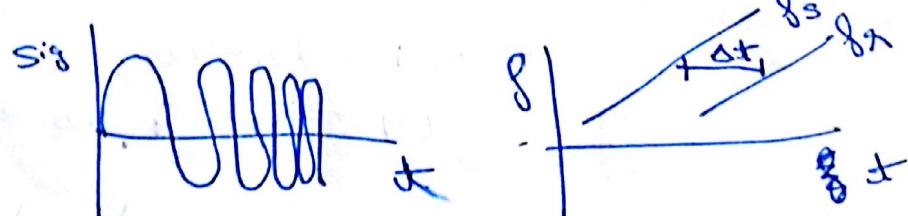
in FFT.

which is easy to find.

From here we can get  $\Delta V$ .

\* Now we face another problem, we can get the velocities of objects, but how do we get distances of objects in the digital domain?

$\rightarrow$  We send a 'Chirp' signal,



• Using the FFTs of the sent & received signals we get  $\Delta t$ .