Channel capacity :- the channel capacity

Represents the maximum amount of Information that can be transmitted by channel per second.

To acheived this rate of transmission, the Information has to be processed properly or coded in Most efficient Manner.

$$\boxed{C = B \log_2 \left(1 + \frac{S}{N}\right)}$$

B → Bandwidth of channel in HZ

S → S/g Power

N → Noise Power

~~This formula is Actually given by Shannon in his Journal paper after~~

Now if BW of channel is Infinite. will capacity of channel become Infinite???

Let us assume the Noise present in Channel is AWGN (Additive White Gaussian Noise) for AWGN $N = \eta B$

$$C = B \log_2 \left(1 + \frac{S}{\eta B}\right)$$

$$= \lim_{B \to \infty} \frac{B \log_2 \left(1 + \frac{S}{\eta B}\right)}{S/\eta B} \times \frac{S}{\eta \cancel{B}}$$

$$= \frac{S}{\eta} \lim_{B \to \infty} \frac{\log_2 (1 + S/\eta B)}{S/\eta B}$$

$$= \frac{S}{\eta} \log_2 e \qquad \left[\because \lim_{x \to 0} \frac{1}{x} \log_2 (1+x) = \log_2 e\right.$$

$$\boxed{\lim_{B \to \infty} C = 1.44 \frac{S}{\eta}}$$

We conclude that even if BW is infinite capacity will finite.

Sampling theorem

IN Digital comm. Analog signal can't be transmitted it should be first converted into Digital where the first step is sampling.

→ A continuous time signal is first converted to discrete ~~d~~ time signal by sampling process

The sufficient No. of samples of signal must be taken So that original signal is Represented in its samples Completely. Also it should be possible to Recover or reconstruct the original signal completely from its samples.

Statement of Sampling theorem (M. Important)

i) A band-limited s/g of finite energy, which has no freq-component higher than f_m HZ, is completely described by its sample value at uniform interval less than or equal to $\frac{1}{2f_m}$ second apart

or

ii) A Band limited signal of finite energy, which has no freq. components higher than f_m HZ, may be completely Recovered from the knowledge of its samples taken at the rate of $2f_m$ samples per second.

from above two we conclude

$$\boxed{f_s \geq 2f_m}$$

$f_s \rightarrow$ sampling freq $f_m \rightarrow$ max freq present in signal

If $\boxed{f_s = 2f_m}$

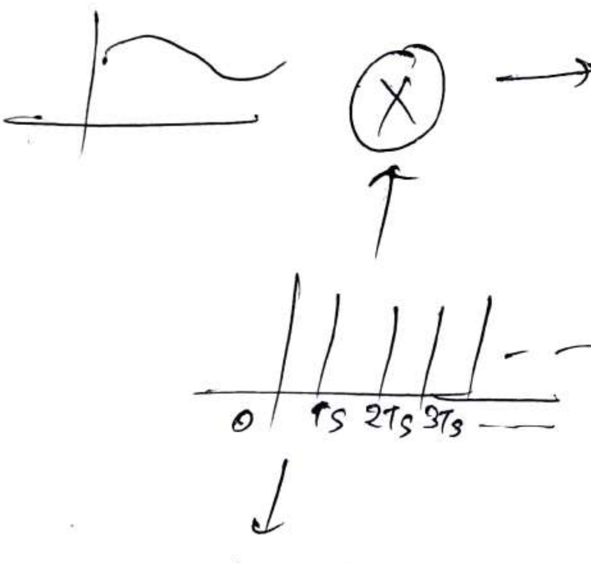
then it is called Nyquist sampling rate

Similarly sampling interval will be

$$\boxed{T_s = \frac{1}{2f_m} \text{ second}}$$

<u>proof</u> why $f_s \geq 2f_m$

Let $m(t)$



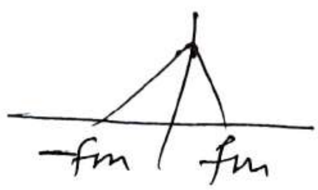
$m(t)\left[\frac{1}{T_s} + \frac{2}{T_s}(\cos\omega_s t + \cos 2\omega_s t + \cdots)\right]$

Train of Impulse

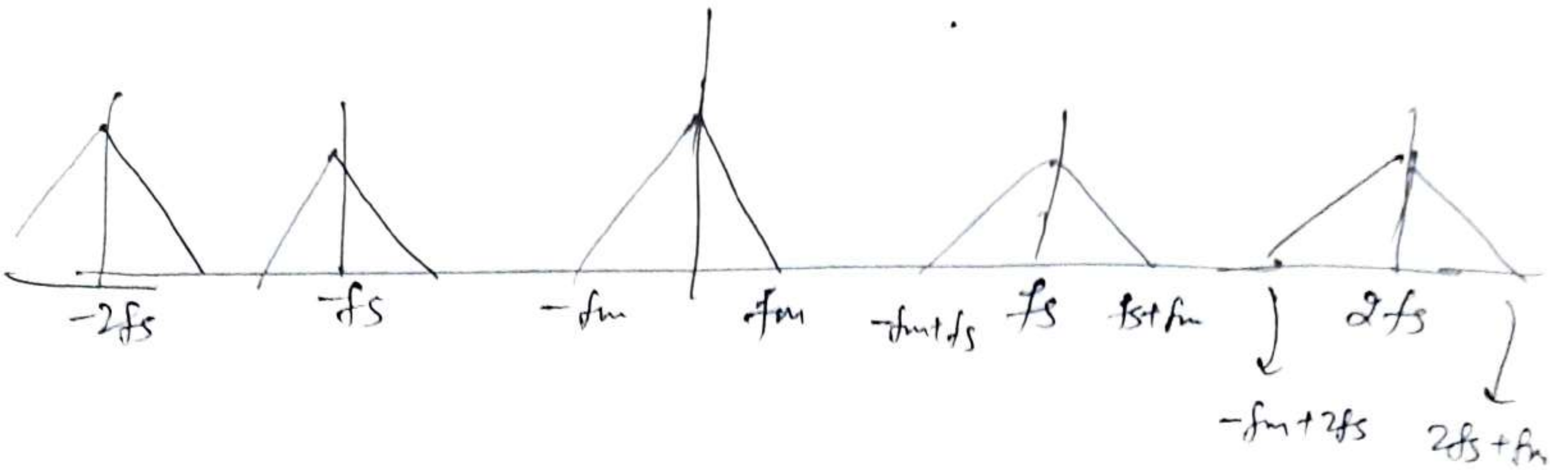
$$\frac{1}{T_s} + \frac{2}{T_s}(\cos\omega_s t + \cos 2\omega_s t + \cdots)$$

$$\Rightarrow \frac{m(t)}{T_s} + \frac{2}{T_s}\left[m(t)\cdot\cos\omega_s t + m(t)\cos 2\omega_s t + \cdots\right] \quad (1)$$

Let $m(t) \longleftrightarrow m(f)$



So fourier transform of eqn (1) will be



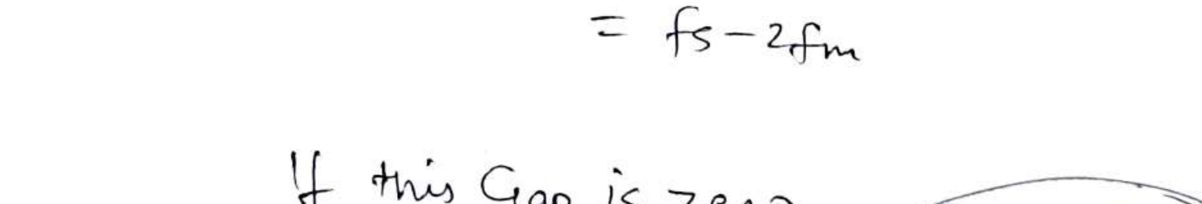
the Gap B/w two sample ~~two~~ is

$= \cancel{fs + fm / fm - 2fs}$

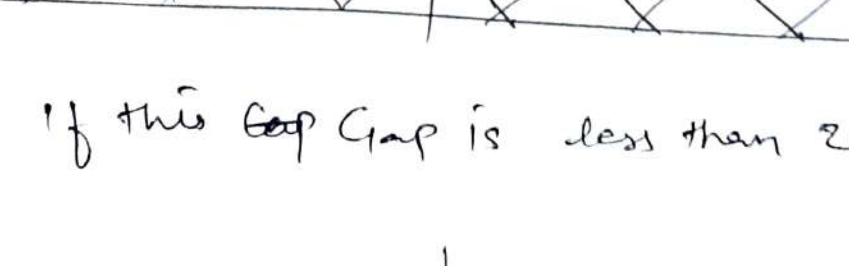
$= 2fs - fm - fs - fm$

$= fs - 2fm$

If this Gap is zero $\Rightarrow$ $fs = 2fm$ Nyquist Interval



If this ~~Gap~~ Gap is less than zero $\Rightarrow fs \leq 2fm$



It Result in ~~Antil.~~ ~~Anti~~.aliasing which Can't be ~~Rostore~~ ~~Resto~~ Restore using LPF filter.

Hence we conclude that for proper Reconstruction $\boxed{f_s \geq 2f_m}$

Type of Sampling

1. ~~N~~ Ideal Sampling
2. Natural
3. flat-top

(1) Ideal Sampling

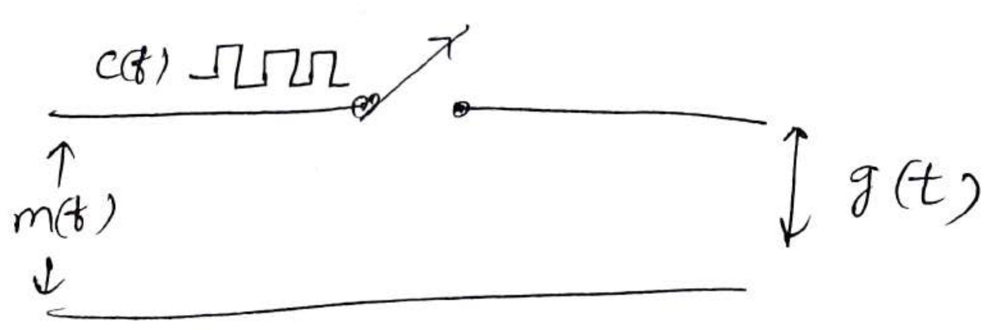
→ The Proof of Sampling theorem is used Ideal or Impulse Sampling In which we simply get the Sample of msg signal by multiplying message with impulse train fixed at a particular Interval.

But practically Impulse can't be generated.

→ It uses multiplication principle.

(2) Natural Sampling

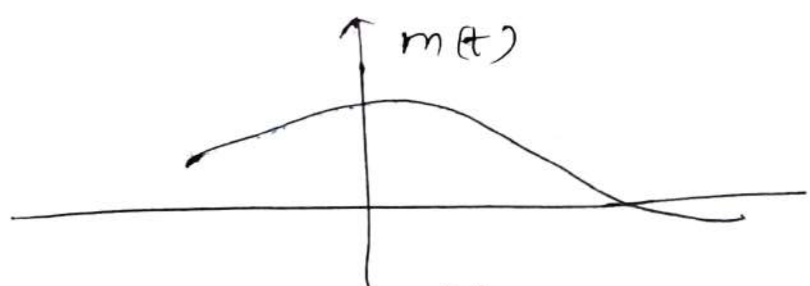
→ It Uses chopping principle

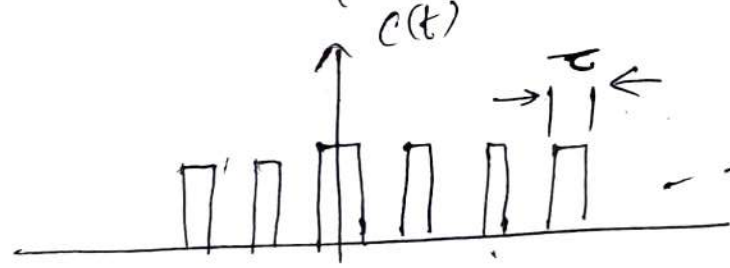


In Ideal sampling, the sampling Results in samples whose width τ approaches zero. Due to this power content in instantaneously sampled pulse is negligible. Thus this method is ~~se~~ not suitable for transmission purpose

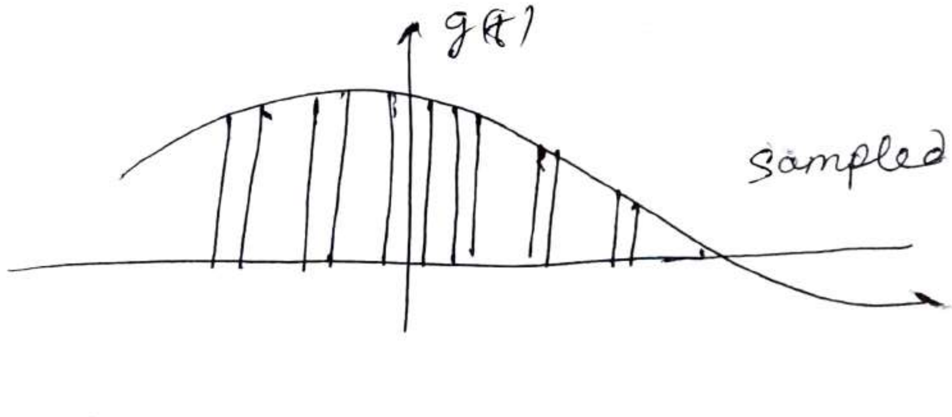
Natural ~~sa~~ sampling is practical Method.

In Natural Sampling pulse has finite width equal to 'τ'. sampled at rate of f_s Hertz





sampling f^{x^n} waveform

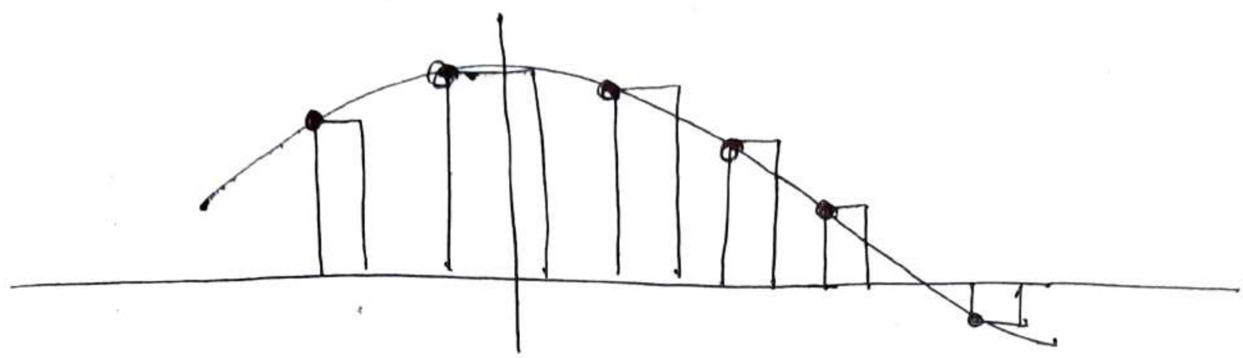


sampled signal

③ <u>Flat top</u>

→ It is also called practical sampling
→ It is based on sample & hold concept.

In flat top sampling the top of samples remain constt. and equal to instanteneous value of baseband signal m(t) at start of sampling.



Sampled signal

Difference Chart

	Ideal	Natural	Flat top
Principle	It uses multiplication	It used chopping principle	It uses sample & hold circuit
Noise Interference	Noise Interference is max.	Noise interference is min.	Interference is max
Feasibility	It is not practically possible	This method is used practically	This method is also used practically
Ckt