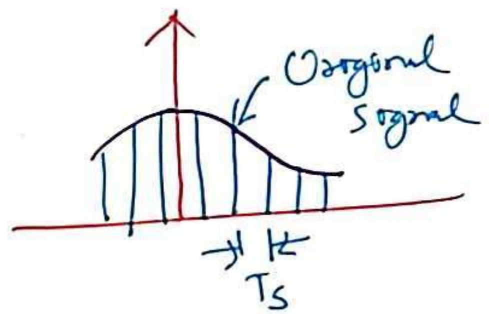
Sampling Theory

- It is a process to convert continous time signal into discreate signal.
- Sufficient no of samples must be taken, so that the orignal signal is reconstructed properly.
- No of samples to be taken depends on maximum signal freq. present in the signal.

- It is a process to convert continuous time signal to discrete signal.
- Sufficient no of samples must be taken, so that the original signal is reconstructed properly.
- No of samples to be taken depends on maximum signal freq. present in the signal.
- Different types of sampling.
  - Ideal samples.
  - Natural samples.
  - Flat-top samples.

- Different types of sampling.
  - Ideal samples.
  - Natural samples.
  - Flat-top samples.



- Statement of sampling theorem

i] A band limited signal of finite energy, which has no freq. component higher than f_m (Hz), is completely described by its sample values at uniform intervals less than or equal to $1/2f_m$

$$\boxed{T_s \leq 1/2f_m}$$

completely described by its sample values at uniform intervals less than or equal to $1/2f_m$

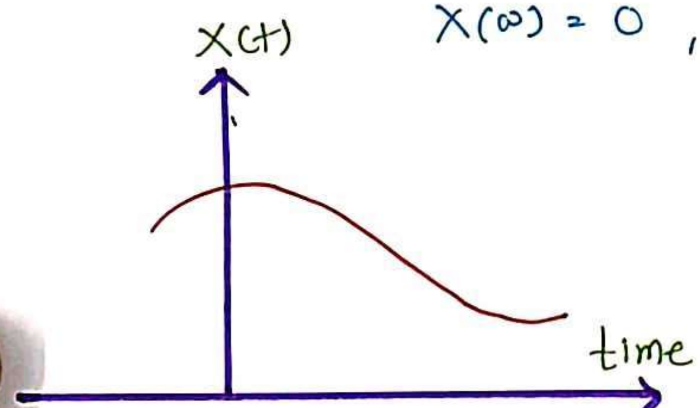
$$\boxed{T_s \leq 1/2f_m}$$

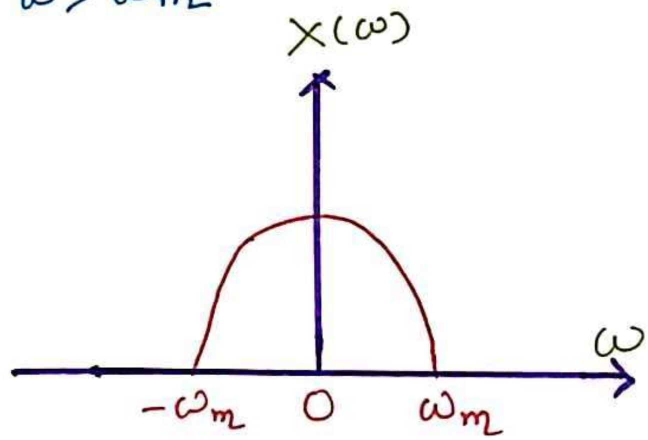
ii] A band limited signal of finite energy, which has no freq. components higher than f_m (Hz), may be completely recovered from the knowledge of its samples taken at the rate of $2f_m$ samples per second.

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

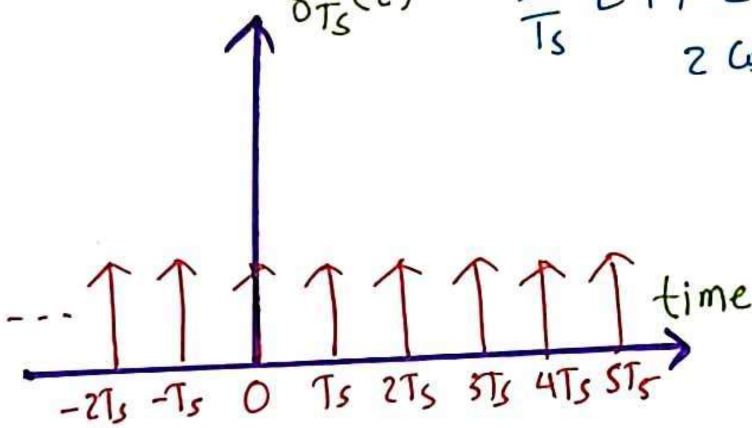
→ If signal is band limited to f_m

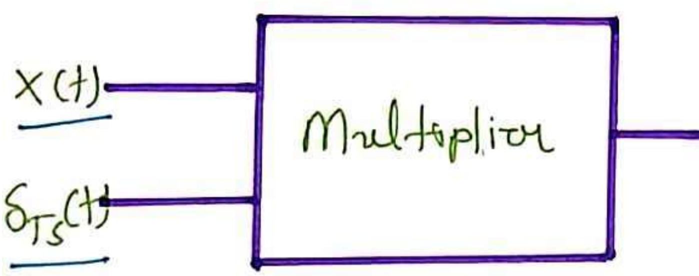
$X(\omega) = 0 \, , \quad \omega > \omega_m$





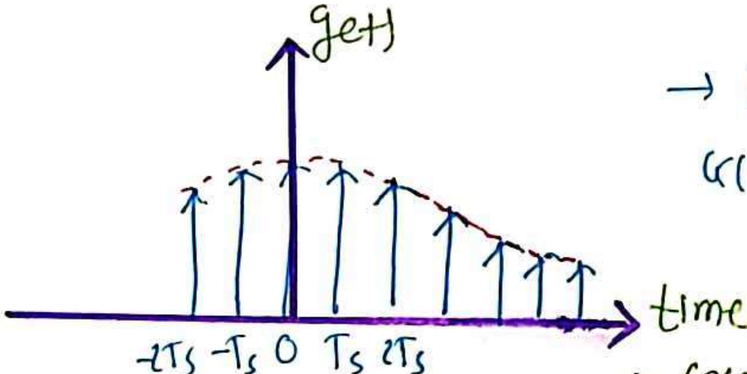
$$\delta_{T_s}(t) = \frac{1}{T_s}\left[1 + 2\cos\omega_s t + 2\cos 2\omega_s t + 2\cos 3\omega_s t + \dots\right]$$





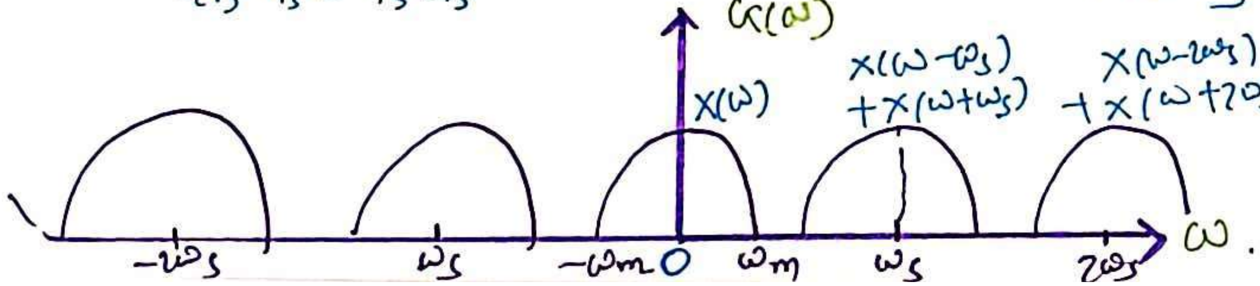
$$g(t) = x(t)\,\delta_{T_s}(t)$$

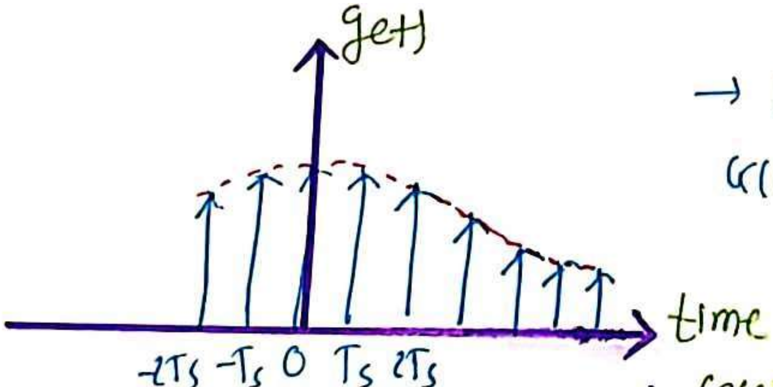
$$= x(t)\left[\frac{1}{T_s}\left(1 + 2\cos\omega_s t + 2\cos 2\omega_s t + 2\cos 3\omega_s t + \dots\right]\right)$$



→ In freq domain

$$G(\omega) = \frac{1}{T_s}\Big[\underline{X(\omega)} + \underline{X(\omega-\omega_s)} + \underline{X(\omega+\omega_s)} + X(\underline{\omega-2\omega_s}) + \underline{X(\omega+2\omega_s)} + \cdots\Big].$$

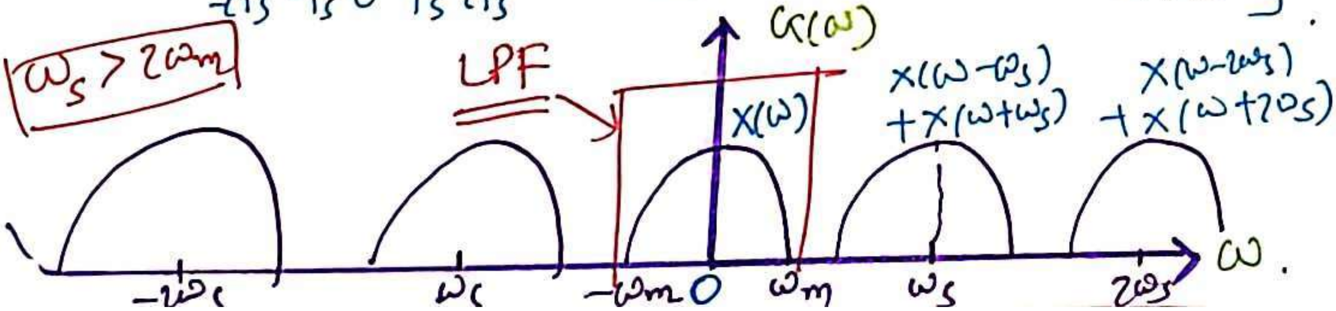


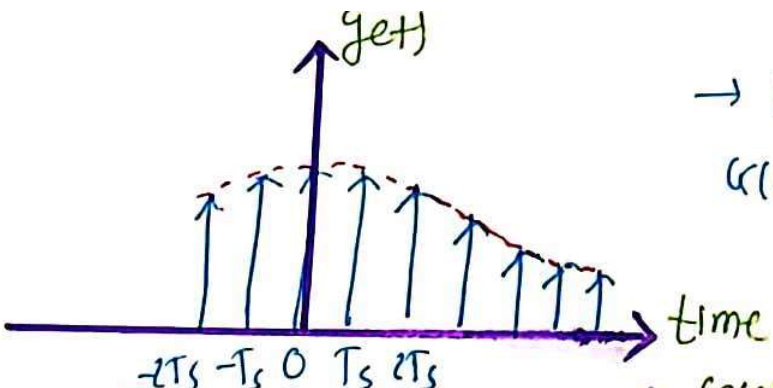


→ In freq domain

$$G(\omega) = \frac{1}{T_s}\Big[X(\omega) + X(\omega-\omega_s) + X(\omega+\omega_s) + X(\omega-2\omega_s) + X(\omega+2\omega_s) + \dots\Big].$$

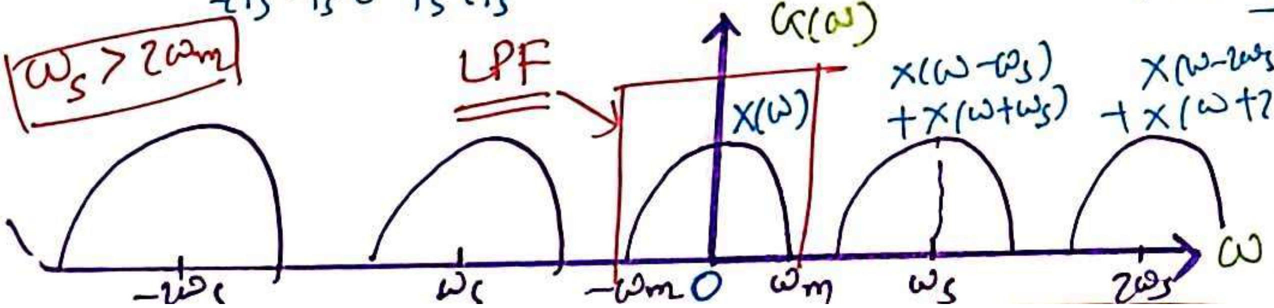
$\boxed{\omega_s > 2\omega_m}$





→ In freq domain

$$G(\omega) = \frac{1}{T_s}\left[X(\omega) + X(\omega-\omega_s) + X(\omega+\omega_s) + X(\omega-2\omega_s) + X(\omega+2\omega_s) + \dots\right].$$



→ As long as $f_s > 2f_m$, $G(\omega)$ will repeat Periodically without overlapping.

→ Spectrum $G(\omega)$ extends upto ∞ freq. but our purpose is to extract original Spectrum $X(\omega)$ out of the Spectrum $G(\omega)$.

→ At receiver we place LPF of freq ω_m. So we can extract original Information.

→ $f_s > 2f_m$. To avoid successive cycles not to overlape

→ Spectrum $x(\omega)$ extends upto ∞ freq. but our Purpose is to extract original Spectrum $X(\omega)$ out of the Spectrum $x(\omega)$.

→ At receiver we place LPF of freq ω_m. So we can extract original Information.

→ $f_s > 2f_m$, To avoid successive cycles not to overlape

$f_s = 2f_m$, Successive cycles just touch eachother.

$f_s < 2f_m$, Successive cycles overlape eachother

→ Hence, for reconstruction without distortion

... back original information.

→ $f_s > 2f_m$, To avoid successive cycles not to overlape

$f_s = 2f_m$, successive cycles just touch eachother.

$f_s < 2f_m$, Successive cycles overlape eachother

→ Hence, for reconstruction without distortion

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

→ $f_s = 2f_m$, thus f_s is reffered as Nyquist rate.

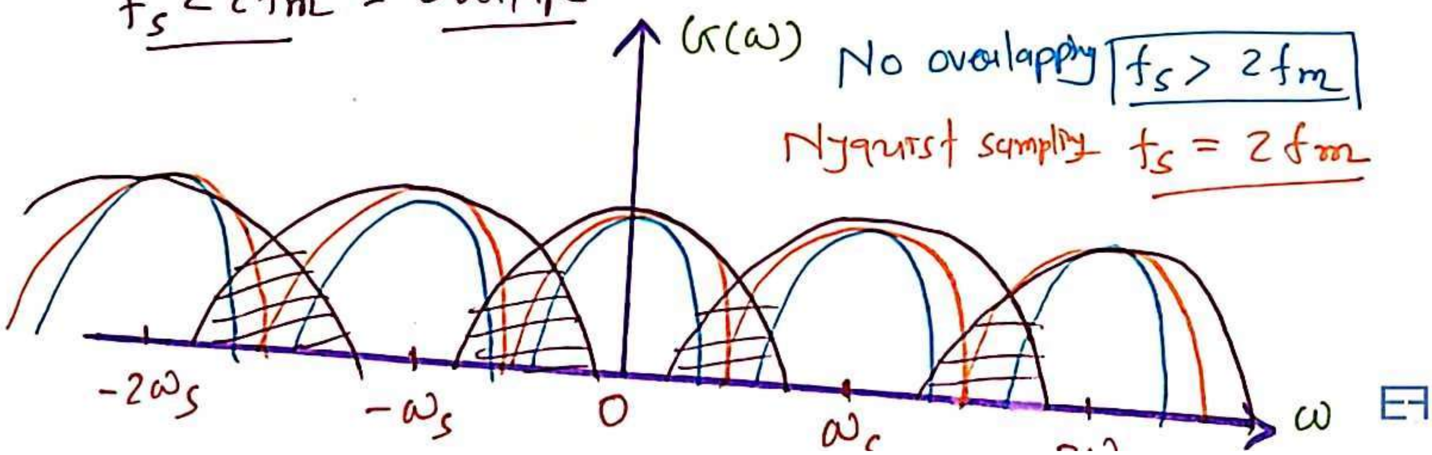
$T_s = 1/2f_m$, T_s is Nyquist Interval.

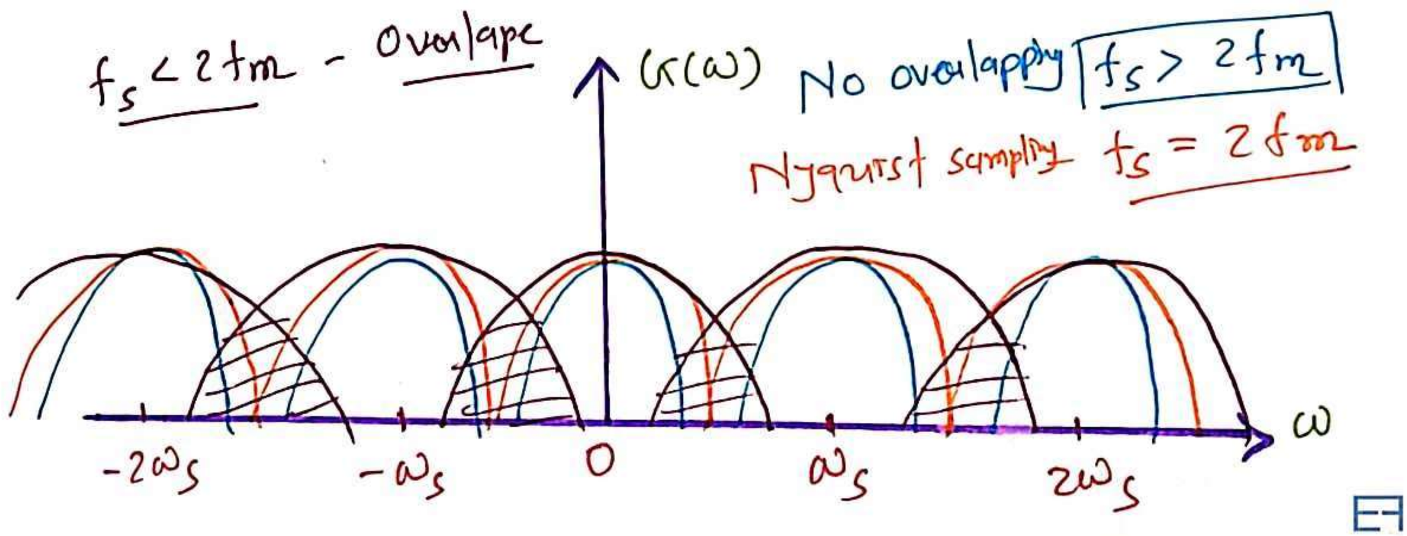
$\boxed{f_s \geq 2f_m}$

→ $f_s = 2f_m$, Here f_s is reffered as Nyquist rate

$T_s = 1/2f_m$, T_s is Nyquist Interval.

$f_s < 2f_m$ - Overlape




$f_s < 2f_m$ - Overlape
$G(\omega)$
No overlapping $\boxed{f_s > 2f_m}$
Nyquist sampling $f_s = 2f_m$
$-2\omega_s$
$-\omega_s$
0
ω_s
$2\omega_s$
ω

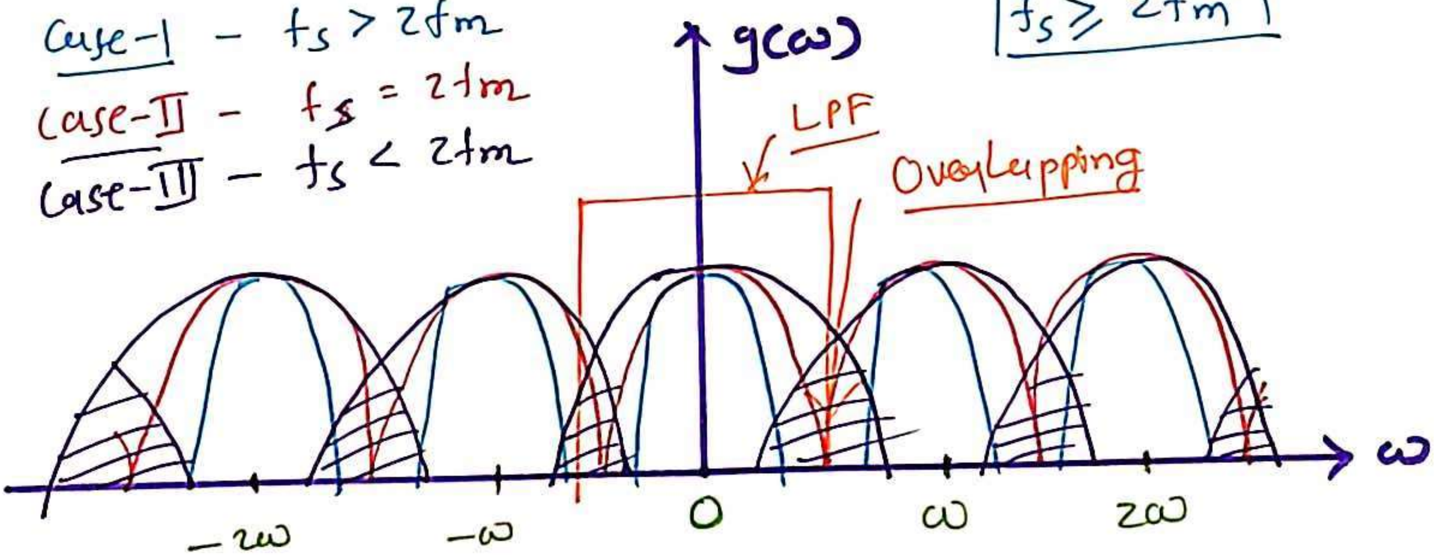
✲ Effect of under sampling - Aliasing

Case-I - $f_s > 2f_m$

Case-II - $f_s = 2f_m$

Case-III - $f_s < 2f_m$

$\boxed{f_s \geq 2f_m}$



- If $f_s < 2f_m$, then successive samples (gets) of $X(\omega)$ will overlap each other.
- Due to Aliasing effect, It is not possible to recover original signal $x(t)$ by LPF.
- Here due to overlape of one region to other region, signal $x(t)$ is distorted.
- So before we go for sampling, we pass original signal through LPF. This is even reffered as pre-alias filter, other name is band limit filter.

EF

Time due to overlape of one region to other region, Signal x(t) is distorted.

- So before we go for sampling, we pass original signal through LPF. This is even reffered as pre - alias filter, other name is band limit filter.

- In Short, to avoid aliasing.

  1) Pre alias Filter can be used

  2) $f_s \geq 2f_m$

Examples on Sampling & Nyquist Rate

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

Calculate the Nyquist rate for this signal.

$f_1 = \dfrac{\omega_1}{2\pi} = 25\ Hz$

$f_2 = \dfrac{\omega_2}{2\pi} = 150\ Hz$

$f_3 = \dfrac{\omega_3}{2\pi} = 50\ Hz$

→ Max freq. $f_m = 150\ Hz$.

→ $f_s = 2f_m$

$= 2 \times 150$

$= 300\ Hz$

... Interval for it

2) Find the Nyquist rate & Nyquist Interval for the signal $x(t) = \frac{1}{2\pi} \cos(4000\pi t) \cos(1000\pi t)$

$$= \frac{1}{4\pi} [2 \cos(4000\pi t) \cos(1000\pi t)]$$

$$= \frac{1}{4\pi} [\underline{\cos(3000\pi t)} + \underline{\cos(5000\pi t)}]$$

$$\rightarrow \left.\begin{aligned} f_1 &= \frac{\omega_1}{2\pi} = 1500\ Hz \\ f_2 &= \frac{\omega_2}{2\pi} = 2500\ Hz \end{aligned}\right\} \rightarrow \text{Max freq. } f_m = 2500\ Hz.$$

$$\rightarrow f_s = 2f_m = 2(2500) = 5000\ Hz$$

$$\rightarrow T_s = 1/f_s = 1/5000 = 0.2\ msec$$

$\rightarrow f_s = 2 f_m$

$= 2(2500)$

$= 5000\ Hz$

$\rightarrow T_s = 1/f_s$

$= 1/5000$

$= 0.2\ msec$

3) Determine the Nyquist rate for a continuous-time signal $x(t) = 6\cos 50\pi t + 20 \sin 300\pi t - 10 \cos 100\pi t$

$\rightarrow f_1 = \frac{\omega_1}{2\pi} = 25\ Hz$

$f_2 = \frac{\omega_2}{2\pi} = 150\ Hz.$

$f_3 = \frac{\omega_3}{2\pi} = 50\ Hz$

$\rightarrow$ Max freq. $f_m = 150\ Hz$

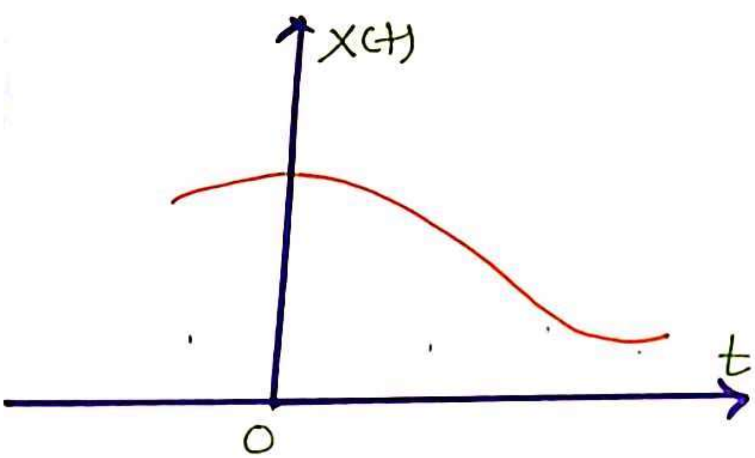
$\rightarrow f_s = 2 f_m$

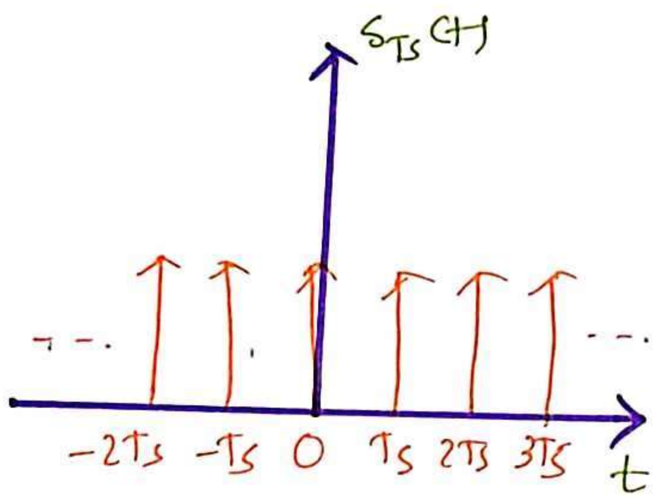
$= 2 \times 150$

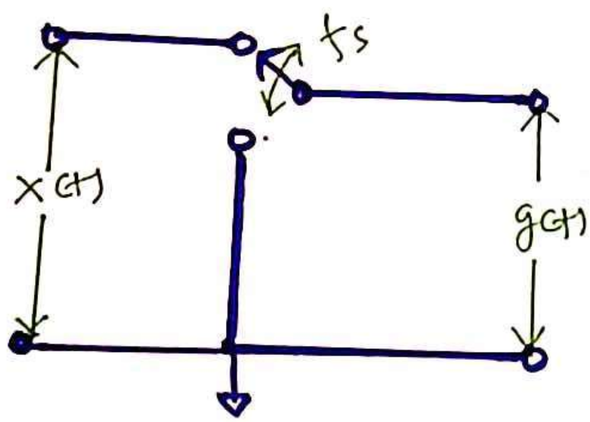
$= 300\ Hz$

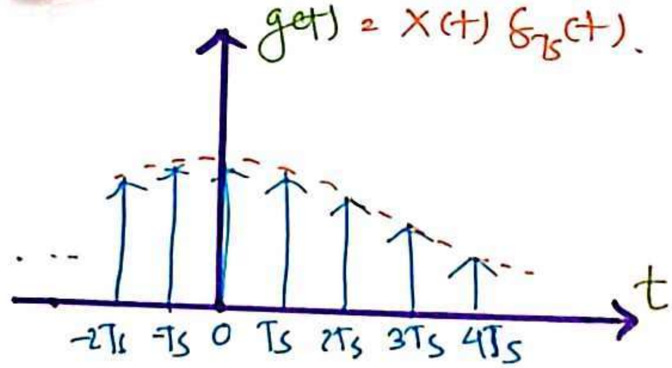
...antaneous Sampling or Impulse Sampling or Ideal Sampling

→ It uses principle of multiplication









- To generate ideal samples train, we use switching sampler.
- If we assume, closing time $t \to 0$, then it has to be consider ideal impulse train.

· Impulse train

To generate ideal samples train, we use switching sampler.

- If we assume, closing time $t \to 0$, then it has to be consider ideal impulse ~~train~~.
- Impulse train

$$\delta_{T_s}(t) = \sum_{n=-\infty}^{\infty} \delta(t-nT_s)$$

→ Practically not Possible

→ High Noise Interference.

- Output

$$g(t) = x(t)\,\delta_{T_s}(t)$$

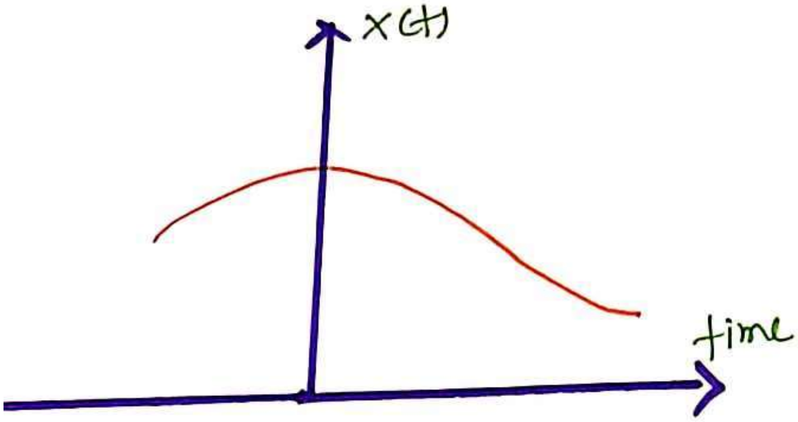
$$= x(t) \sum_{n=-\infty}^{\infty} \delta(t-nT_s)$$

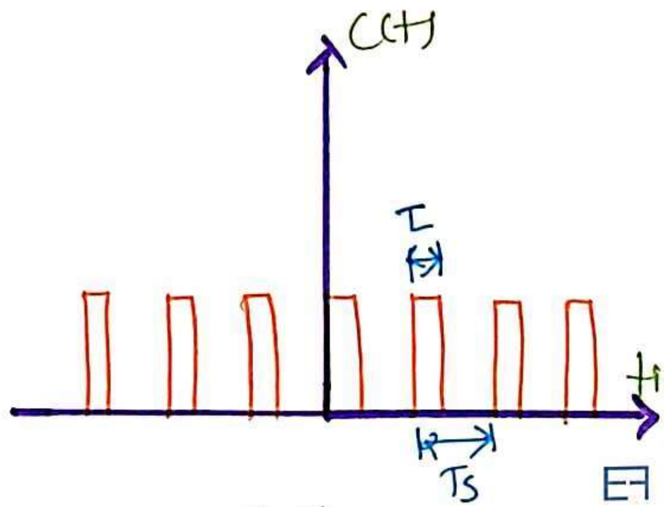
- In Freq Domain, $G(\omega) = f_s \sum_{n=-\infty}^{\infty} X(f-nf_s)$

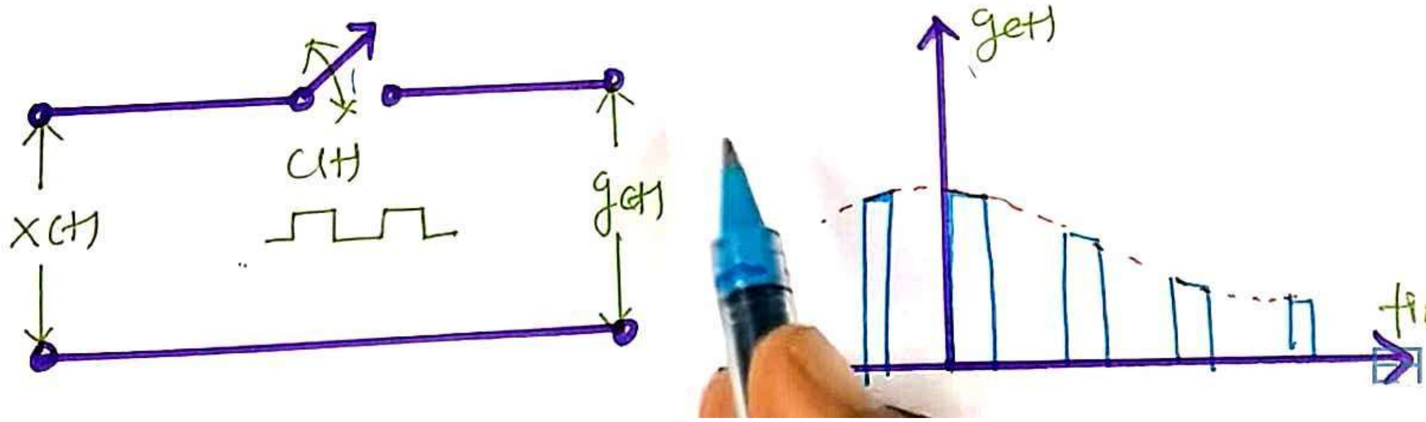
EF

Natural Sampling

- It uses chopping principle







x(t)
c(t)
g(t)
g(t)
t

- $g(t) = x(t)$, $c(t) = A$

$g(t) = 0$, $c(t) = 0$

- So Mathematically

$g(t) = x(t)\, c(t)$

$$= \frac{\tau A}{T_s} \sum_{n=-\infty}^{\infty} x(t)\ \text{sinc}\ (f_n \tau)\ e^{j2\pi f_s t}$$

- freq. domain

$$G(\omega) = \frac{\tau A}{T_s} \sum_{n=-\infty}^{\infty} \sin\ (n f_s \tau)\ X\ (f - n f_s)$$

EF

$g(t) = 0 \quad , \quad c(t) = 0$

- So Mathematically

$$g(t) = x(t)\, c(t)$$

$$= \frac{\tau A}{T_s} \sum_{n=-\infty}^{\infty} x(t)\, \text{sinc}\,(f_n \tau)\, e^{j2\pi f_s t}$$

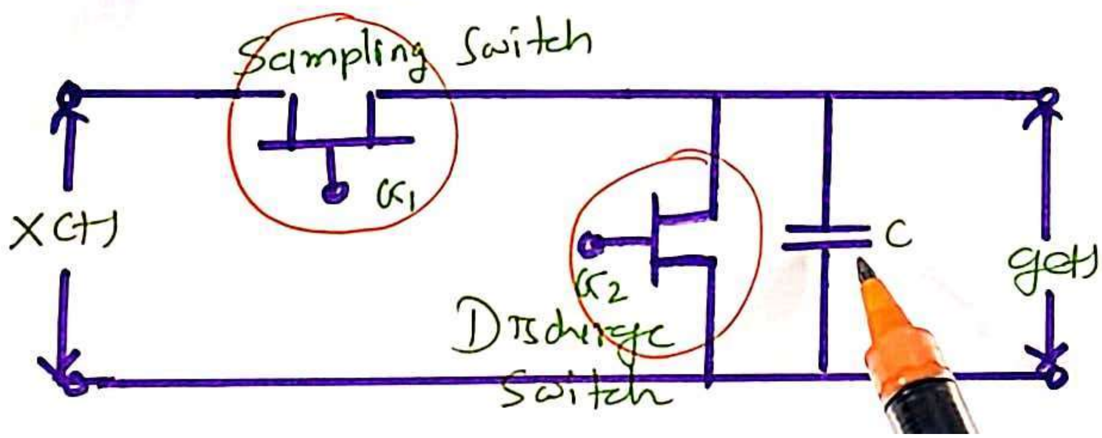
- freq. domain

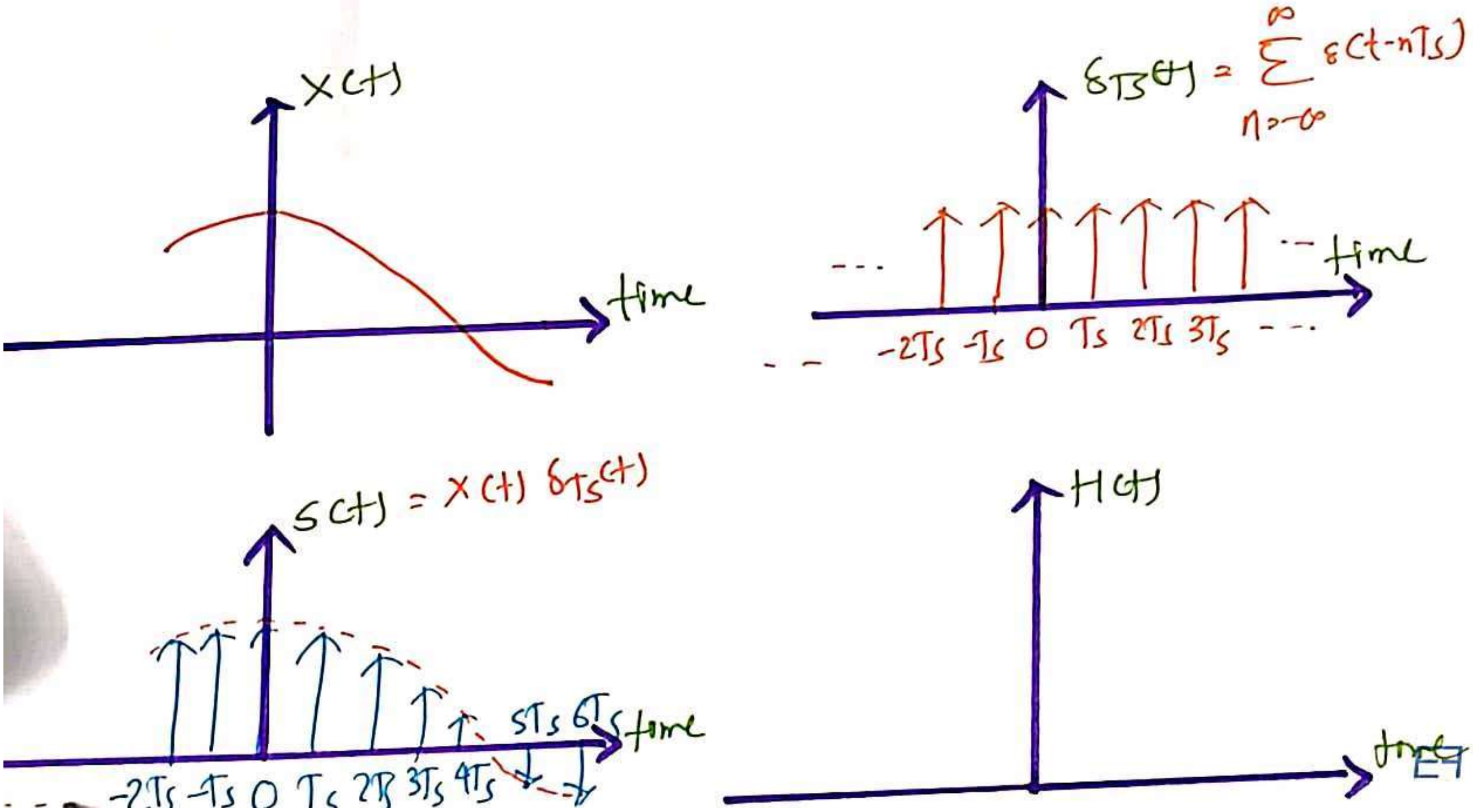
$$G(\omega) = \frac{\tau A}{T_s} \sum_{n=-\infty}^{\infty} \sin\,(n f_s \tau)\, X\,(f - n f_s)$$

- This method is used practically.
- Noise Interference is Less.

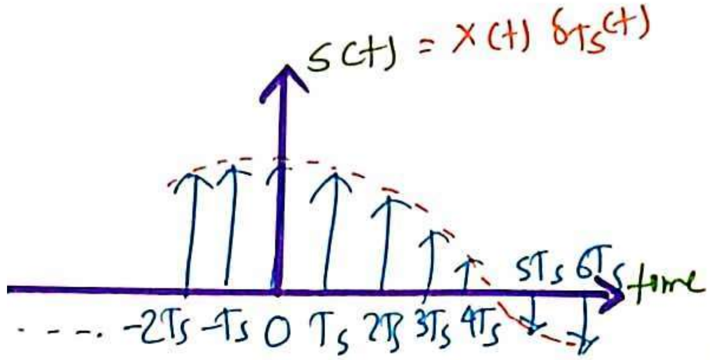
Flat Top Sampling [PAM]

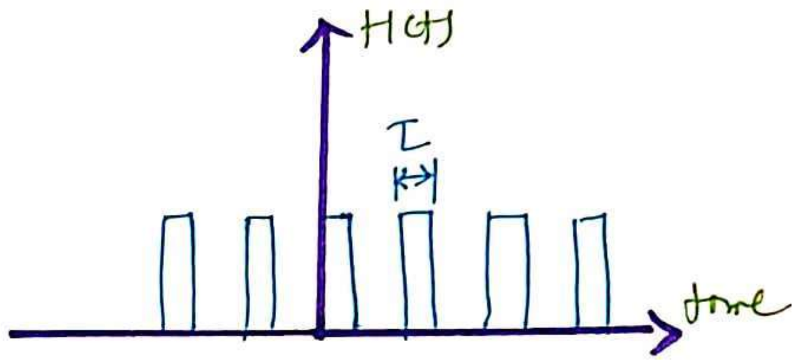
- It uses Sample and hold circuit
- It is practically possible like natural sampling but Flat top sampling is easier compared to natural sampling.
- It has very high noise Interference.

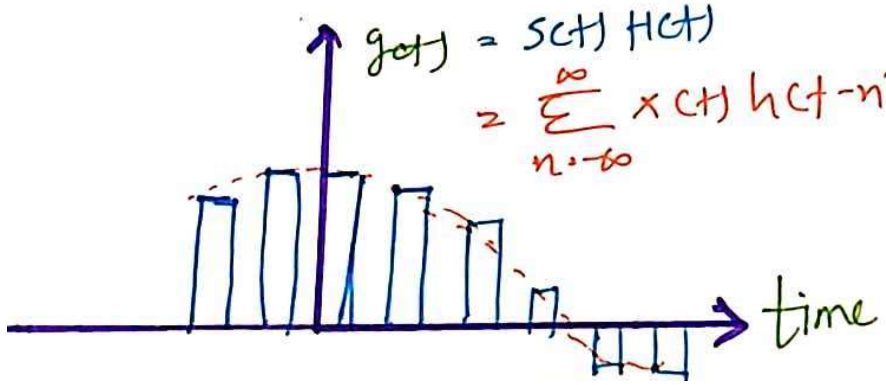




x(t)
time
$\delta_{T_s}(t) = \sum_{n=-\infty}^{\infty} \delta(t-nT_s)$
time
$-2T_s$ $-T_s$ 0 T_s $2T_s$ $3T_s$
$s(t) = x(t)\,\delta_{T_s}(t)$
$-2T_s$ $-T_s$ 0 T_s $2T_s$ $3T_s$ $4T_s$ $5T_s$ $6T_s$
time
H(t)
time





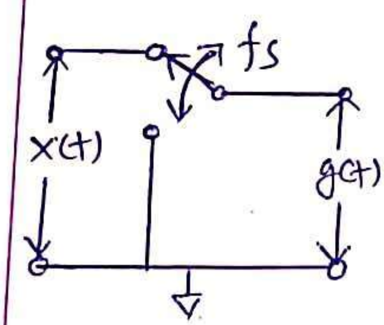
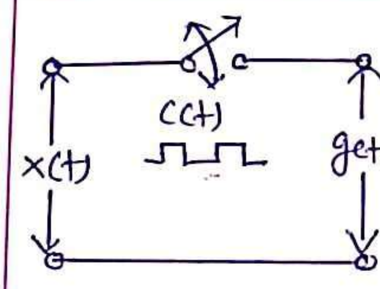
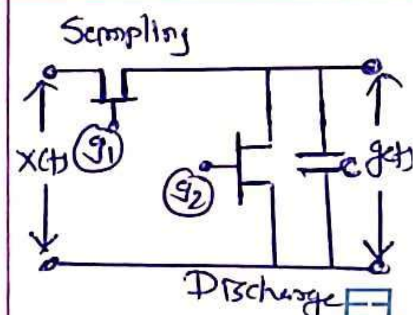


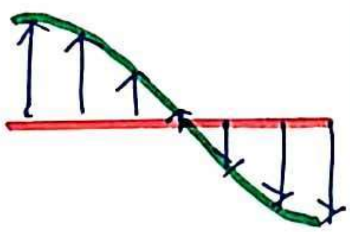
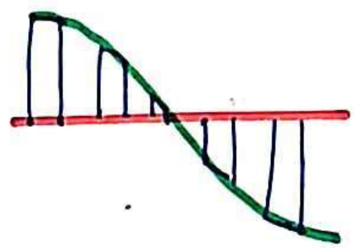
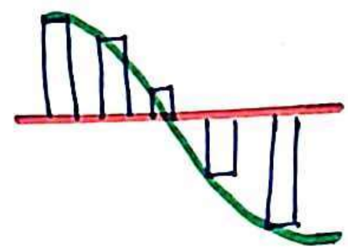
$$g(t) = s(t)\,H(t) = \sum_{n=-\infty}^{\infty} x(t)\,h(t - nT_s)$$

→ freq. domain

$$G(\omega) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s)\,H(f)$$

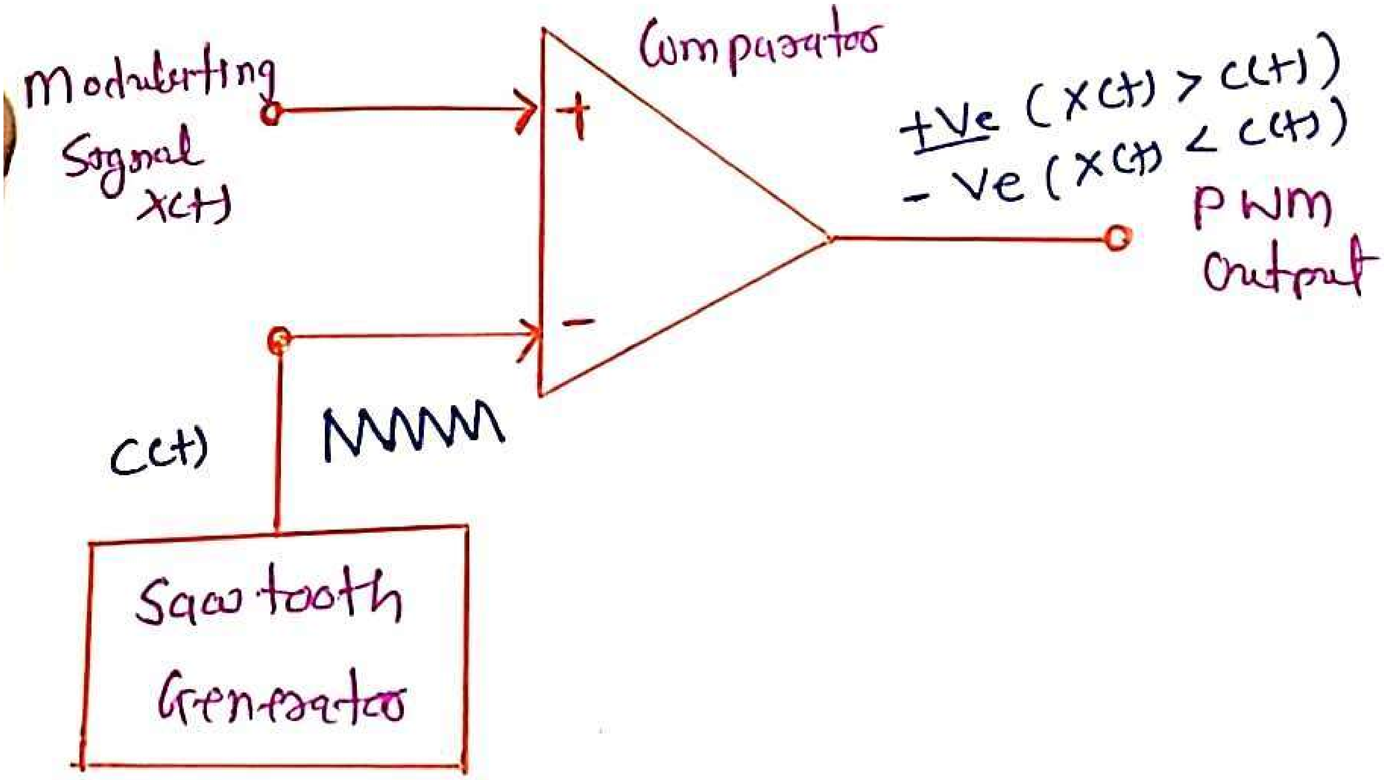
Performance Comparison of Sampling techiques.

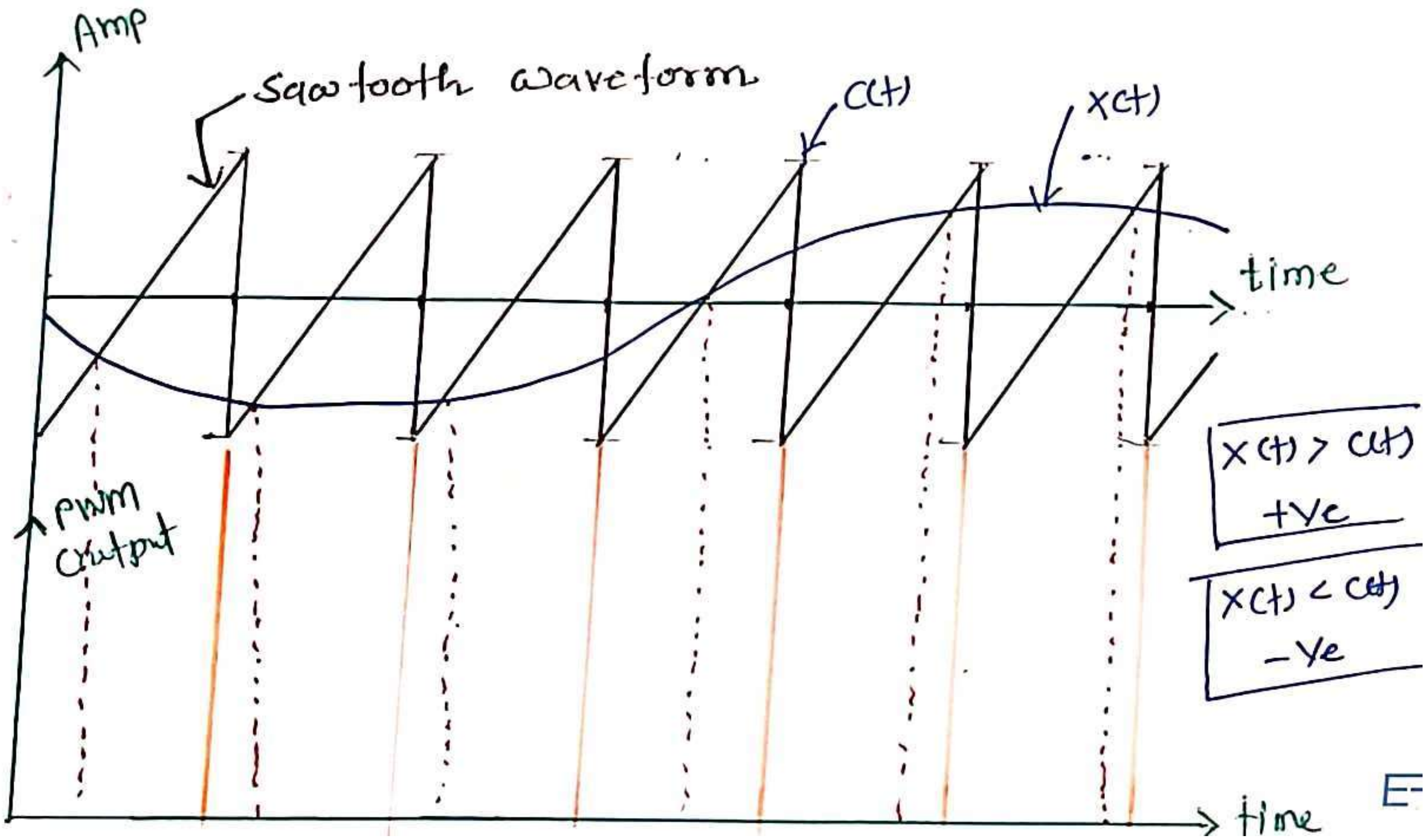
Performance Parameter	Ideal Sampling	Natural Sampling	Flat top Sampling
Sampling Principle	Multiplication	Chopping	Sample & Hold Circuit
Generation Circuit			

Waveforms			
Feasibility	Practically not possible	used Practically	used Practically.
Noise Interference	Very high	Less	high

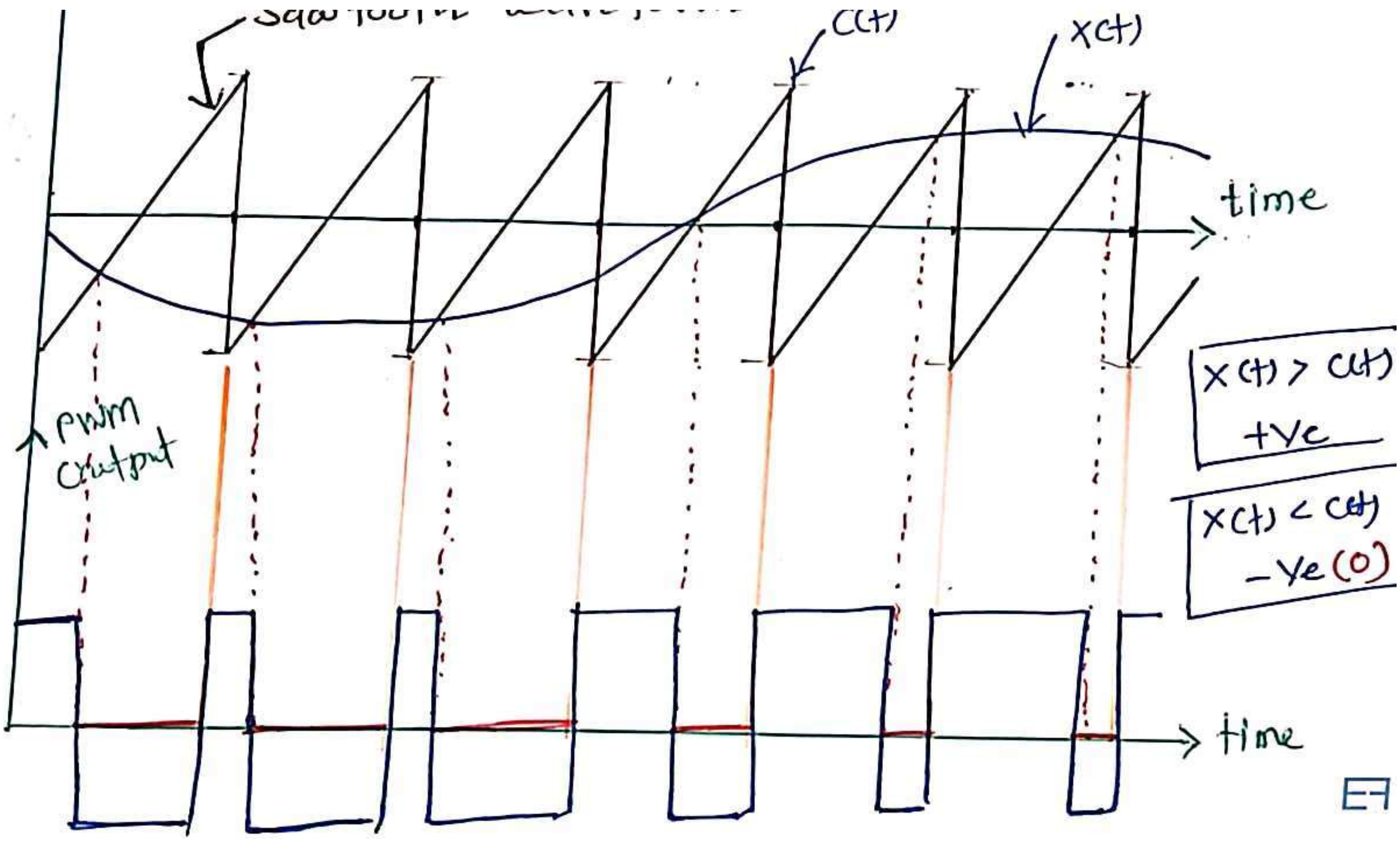
Feasibility	Practically not possible	used practically	used practically.
Noise Interference.	Very high	Less	high
Time domain Representation	$g(t) = \sum_{n=-\infty}^{\infty} x(t)\,\delta(t-nT_s)$	$g(t) = \frac{\tau A}{T_s} \sum_{n=-\infty}^{\infty} x(t)\,\mathrm{Sinc}(n f_s \tau)\, e^{j2\pi n f_s t}$	$g(t) = \sum_{n=-\infty}^{\infty} x(t)\, h(t-nT_s)$
Freq. domain Representation	$G(f) = f_s \sum_{n=-\infty}^{\infty} X(f-nf_s)$	$G(f) = \frac{\tau A}{T_s} \sum_{n=-\infty}^{\infty} \mathrm{Sinc}(n f_s \tau)\, X(f-nf_s)$	$G(f) = f_s \sum_{n=-\infty}^{\infty} X(f-nf_s)\, H(f)$

PWM (Pulse Width Modulation)



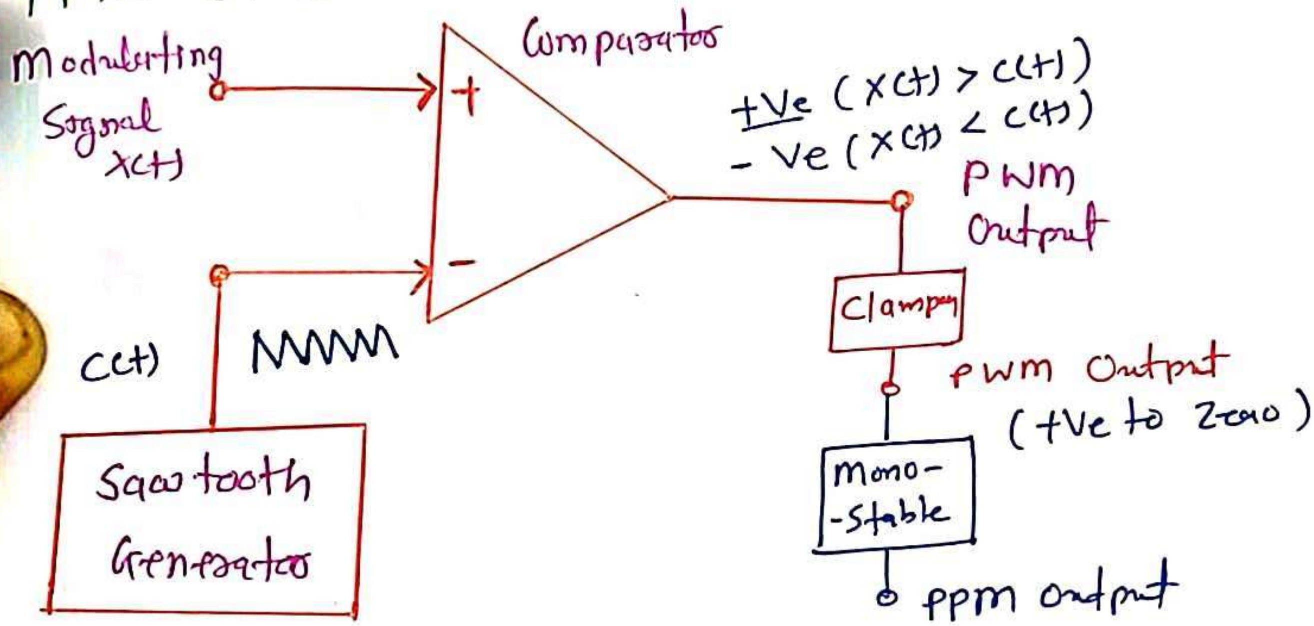


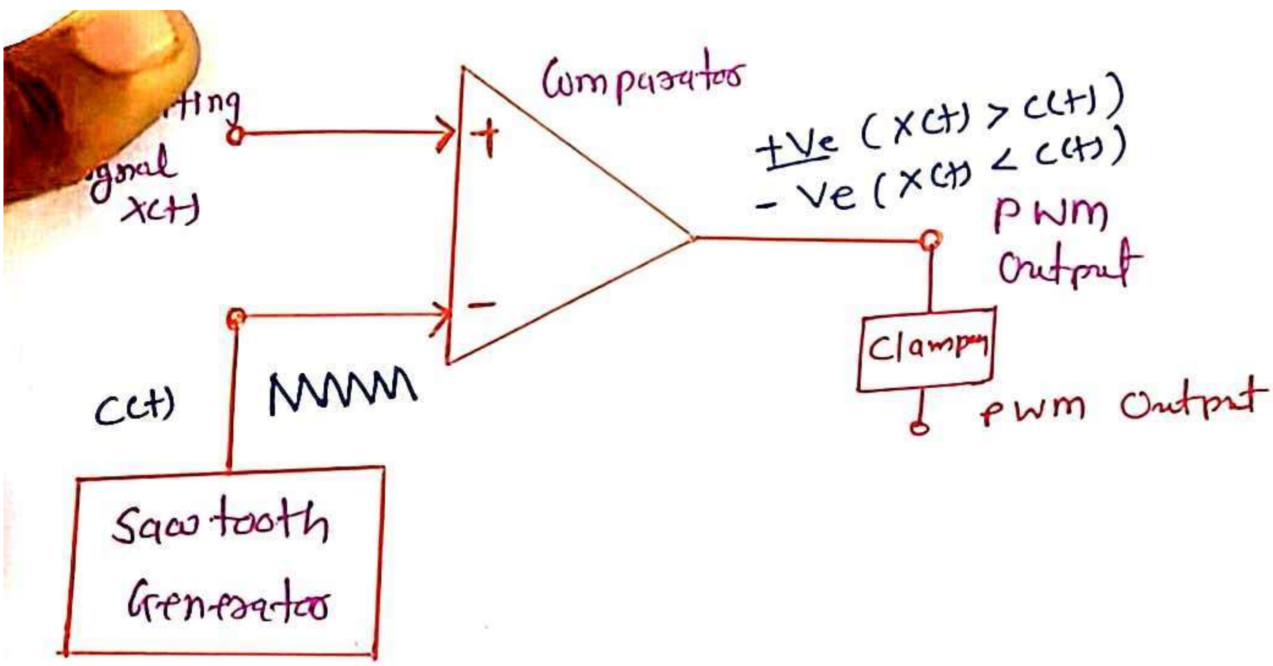
Amp
Sawtooth waveform
C(t)
X(t)
time
PWM
Output
X(t) > C(t)
+Ve
X(t) < C(t)
-Ve
time



C(t)
X(t)
time
PWM Output
X(t) > C(t)
+Ve
X(t) < C(t)
-Ve (0)
time

PWM (Pulse Width Modulation)
PPM (Pulse Position Modulation)



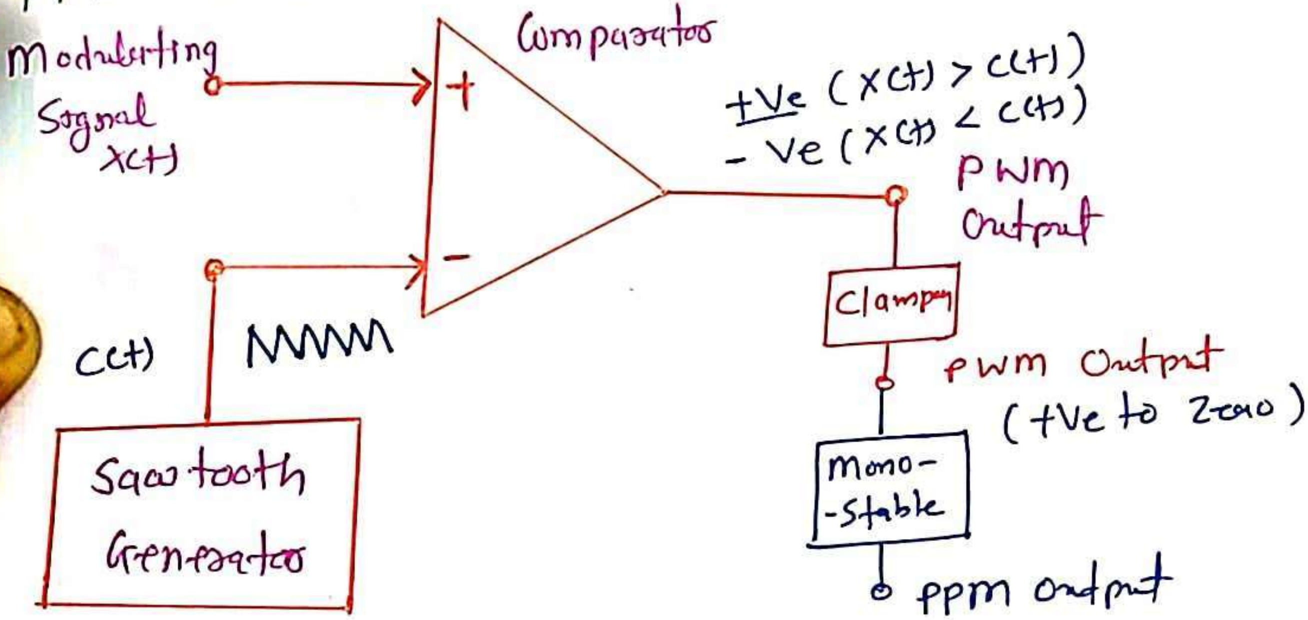


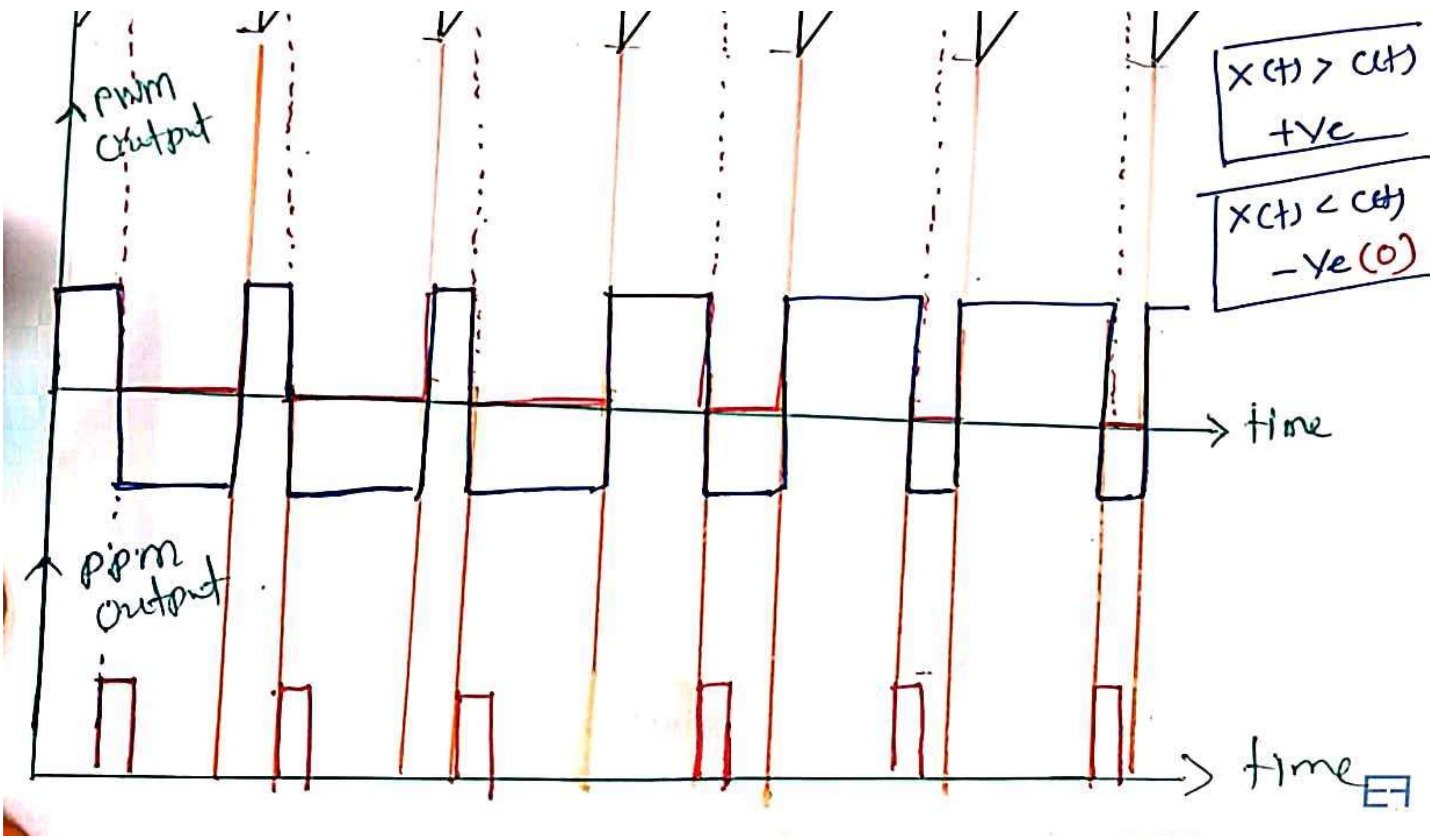
Amp ↑

Sawtooth waveform, c(t), x(t)

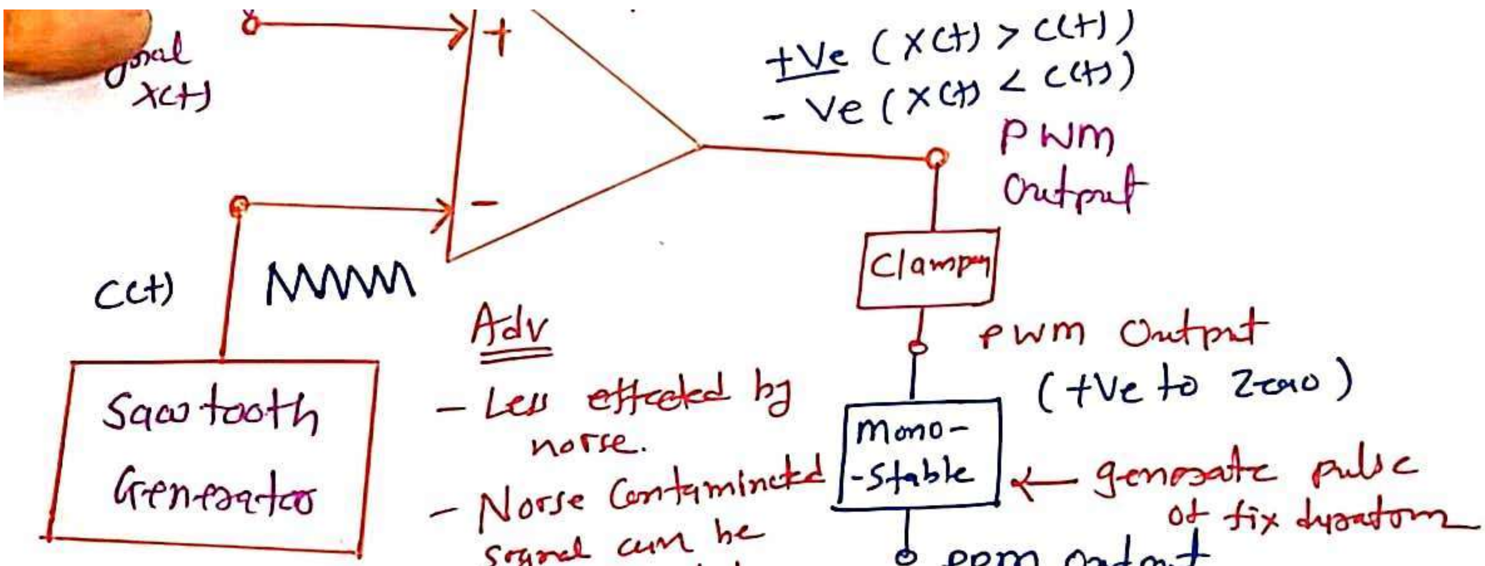
E

PWM (Pulse Width Modulation)
PPM (Pulse Position Modulation)



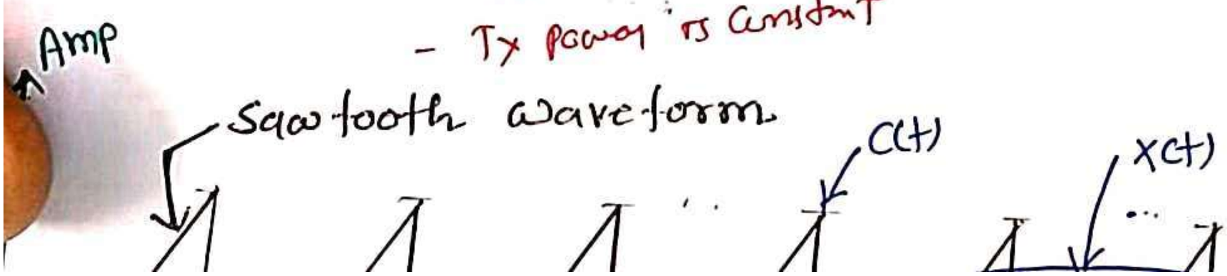


PWM output
PPM output
time
time
X(t) > C(t)
+Ve
X(t) < C(t)
-Ve (0)

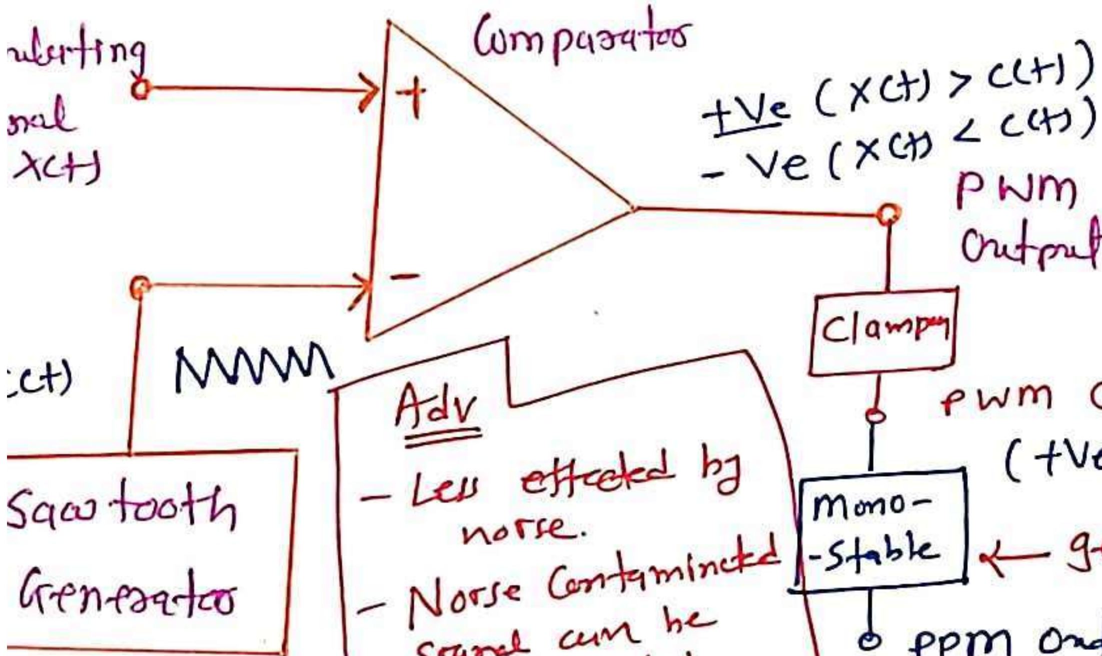


Adv

- Less effected by noise.
- Noise Contaminated signal can be reconstructed.
- Tx power is constant



18) (Pulse Width Modulation)
m (Pulse Position Modulation)



P

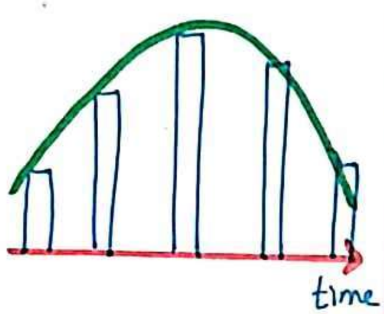
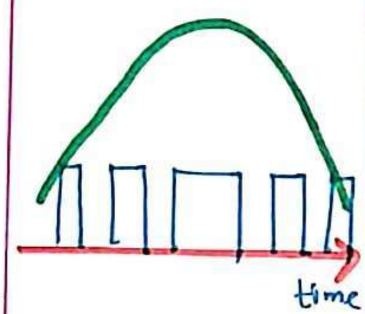
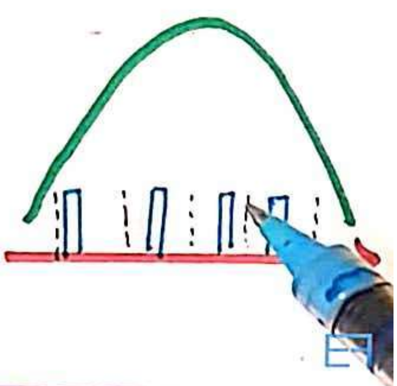
Adv

- Less effected by noise.
- Noise Contaminated signal can be reconstructed.
- Tx power is constant

Disadv

- Synch. is req.d at demodulation.
- High BW.

Performance Comparison of Pulse analog Modulation

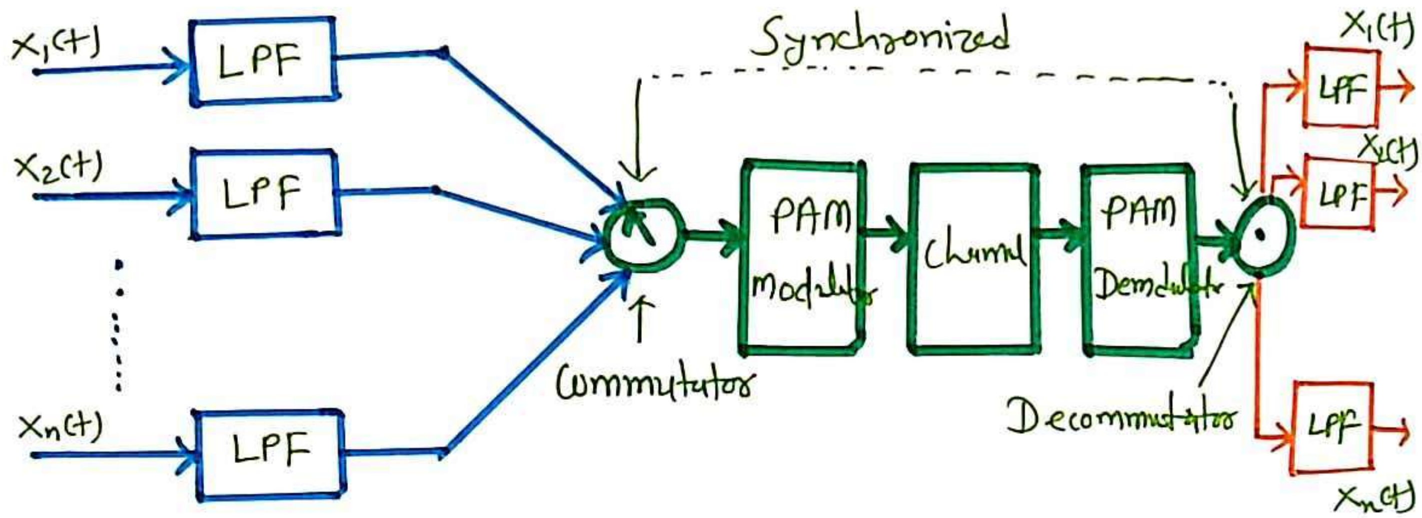
Performance Parameter	PAM	PWM	PPM
Waveform			

	time	time	time
Working Principle	Amplitude of Pulse proportional to amp. of modulating signal.	- Width of Pulse is proportional to amplitude of modulating signal	- Relative Position of Pulse is proportional to amp. of modulating signal
Bandwidth	- BW is depending on width of Pulse	- BW depends on rise time of Pulse.	- BW depends on rise time of pulse.
Transmitted Power	- Varies w.r.t time	- Varies w.r.t time	- Constant.

Bandwidth	- BW is depending on width of pulse	- BW depends on rise time of Pulse.	- BW depends on rise time of pulse.
Transmitted Power	- Varies w.r.t time	- Varies w.r.t time	- Constant.
Noise Interference	- Max	- Min	- Less
-	- It is similar to AM	- It is similar to FM	- It is similar to PM.

Time Division Multiplexing (TDM)

- The sampled PAM waveform is off for most of the time.
- During the OFF period, the channel can be used to transmit samples of other waveforms.
- The concept of Interleaving samples from several signals into a single waveforms is called TDM.

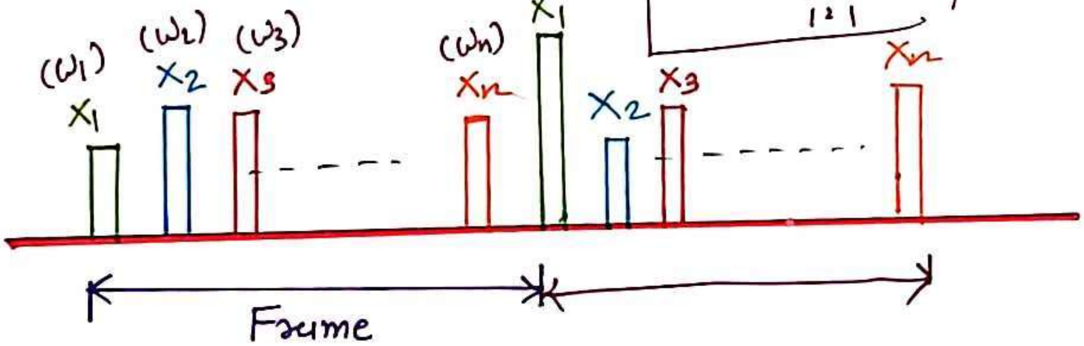


$x_1(t)$
LPF
$x_2(t)$
LPF
$x_n(t)$
LPF
Synchronized
Commutator
PAM Modulator
Channel
PAM Demodulator
Decommutator
$x_1(t)$
LPF
$x_2(t)$
LPF
LPF
$x_n(t)$

- The minimum bandwidth (TDM-PAM)

$$W = \omega_1 + \omega_2 + \omega_3 + \dots + \omega_n$$

$$\boxed{W = \sum_{i=1}^{N} \omega_i}$$



A signal $x_1(t)$ is Band-limited to 4.2 kHz and three other signals $x_2(t)$, $x_3(t)$ and $x_4(t)$ are band limited to 1.4 kHz each. Assume that these signals are transmitted by TDM.

(a) Set-up a scheme TDM for realizing this multiplexing requirement with each signal sampled at its Nyquist rate.

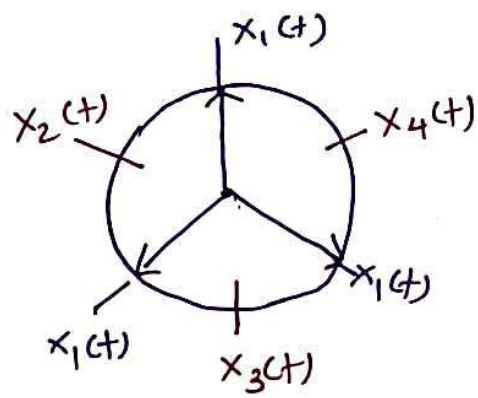
(b) Identify the required speed of the Commutator in samples/sec.

(c) Identify the minimum Bandwidth requirement of the channel.

- $x_1(t) = 4.2$ KHz
  $x_2(t) = 1.4$ KHz
  $x_3(t) = 1.4$ KHz
  $x_4(t) = 1.4$ KHz

(a) Nyquist rate

$x_1(t)$	$B_1 = 4.2$ KHz	$f_{s_1} = 2B_1 = 8.4$ KHz
$x_2(t)$	$B_2 = 1.4$ KHz	$f_{s_2} = 2B_2 = 2.8$ KHz
$x_3(t)$	$B_3 = 1.4$ KHz	$f_{s_3} = 2B_3 = 2.8$ KHz
$x_4(t)$	$B_4 = 1.4$ KHz	$f_{s_4} = 2B_4 = 2.8$ KHz



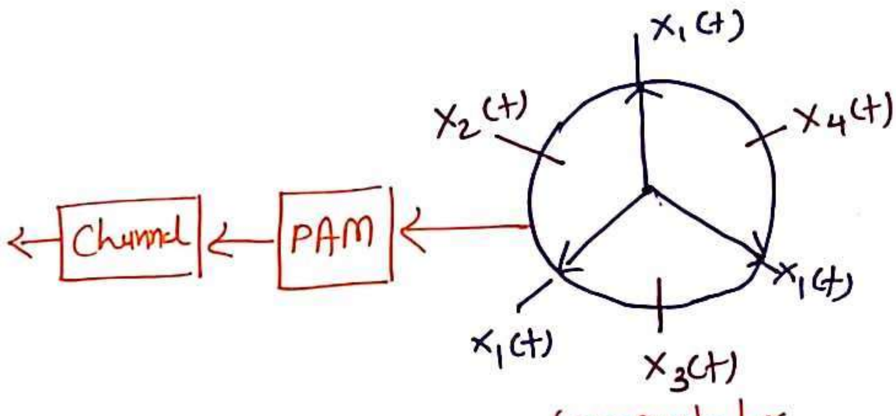
- If Commutator is rotating by 2800 rot/sec

E

- $x_1(t) = 4.2$ KHz
  $x_2(t) = 1.4$ KHz
  $x_3(t) = 1.4$ KHz
  $x_4(t) = 1.4$ KHz

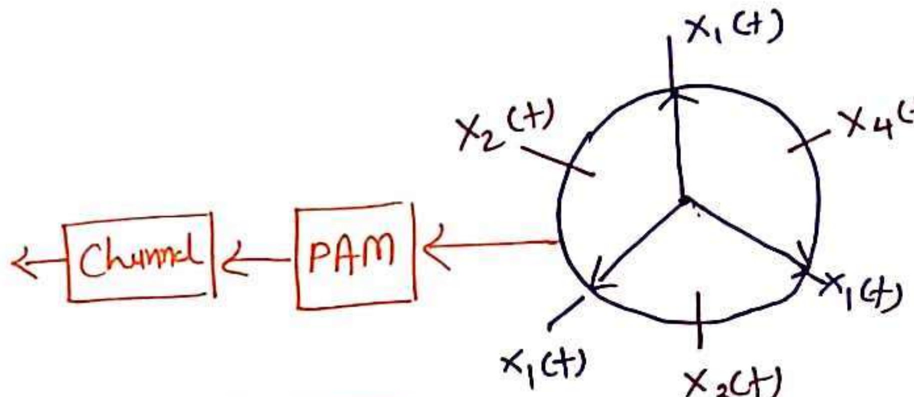
(a) Nyquist rate

$x_1(t)$	$B_1 = 4.2$ KHz	$f_{s_1} = 2B_1 = 8.4$ KHz
$x_2(t)$	$B_2 = 1.4$ KHz	$f_{s_2} = 2B_2 = 2.8$ KHz
$x_3(t)$	$B_3 = 1.4$ KHz	$f_{s_3} = 2B_3 = 2.8$ KHz
$x_4(t)$	$B_4 = 1.4$ KHz	$f_{s_4} = 2B_4 = 2.8$ KHz



Commutator.

- If Commutator is rotating by 2800 rot/sec



- If Commuta
is rotating
by 2800 ro

Commutator.

Minimum BW

$B = B_1 + B_2 + B_3 + B_4$

$= 4.2 + 1.4 + 1.4 + 1.4$

$= 8.4$ KHz

Signals	Samples/sec
$x_1(t)$	8400 Samples/sec
$x_2(t)$	2800
$x_3(t)$	2800
$x_4(t)$	2800
	16800 samples/sec

(2) Assume that the Commutator O/p is Quantized with $L = 1024$ and the result is binary coded. what is the O/p bit rate ? [take ref. of example 1]

$-\ L = 1024 = 2^m$

$\Rightarrow m = \log_2 L$

$= \log_2 1024$

$= 10$ bits/sample.

$\rightarrow$ Output bitrate

$R_b = m \times f_s$

$= \left(\frac{\text{bits}}{\text{sample}}\right)\left(\frac{\text{samples}}{\text{sec}}\right)$

$= 10 \times 16.8$ KHz

$= \underline{168 \text{ Kbpsec}}$

Two analog signals $x_1(t)$ & $x_2(t)$ are to be transmitted over a common channel by means of TDM.
The highest freq. of $x_1(t)$ is 3 KHz and that of $x_2(t)$ is 4.5 KHz. What is the minimum permissible sampling rate ?

- $x_1(t)$ — $B_1 = 3\,KHz$
- $x_2(t)$ — $B_2 = 4.5\,KHz$

The highest freq. of $x_1(t)$ is 3 KHz and that of $x_2(t)$ is 4.5 KHz. What is the minimum permissible Sampling rate ?

- $x_1(t)$ — $B_1 = 3$ KHz
- $x_2(t)$ — $B_2 = 4.5$ KHz
- Composite signal of $x_1(t)$ and $x_2(t)$ is $x_1(t) + x_2(t)$.
- Bandwidth of Composite Signal is highest of $x_1(t)$ and $x_2(t)$.
- highest of $x_1(t)$ & $x_2(t)$ is 4.5 KHz.
- min Sampling rate $f_s = 2W$

$$= 2 \times 4.5$$

$$= 9 \text{ K samples/sec.}$$

The T1 carrier system used in digital telephony multiplexers 24 voice channels based on 8 bit PCM. Each voice signal is put through a LPF with the cut off frequency of 3.4 KHz. The LPF O/P is sampled at 8KHz. Then a single bit is added at the end of the frame for the purpose of Synchronization. Calculate

a) Bit duration

b) Transmission rate

c) Nyquist Bandwidth

b) Transmission rate

c) Nyquist Bandwidth

- T_1 Carrier System
- It uses TDM-PCM

- Voice Signal
- 300 Hz - 3.4 kHz
- $f_s = 2B = 2 \times 3.4$ kHz $= 6.4$ kHz
- we prefer over sampling

$$\boxed{f_s = 8 \text{ kHz}}$$

1 ~ → LPF → quantizer →

2 ~

3 ~

⋮

24 ~

Voice signal.

$f_c = 3.4$ kHz

$m = 8$ bits

$L = 2^m = 2^8 = 256$ levels

- It uses TDM-PCM

- 300 Hz - 3.4 kHz

$$1 \sim \rightarrow \boxed{LPF} \rightarrow \boxed{quant} \rightarrow$$

2 ~

3 ~

⋮

24 ~

Voice signal.

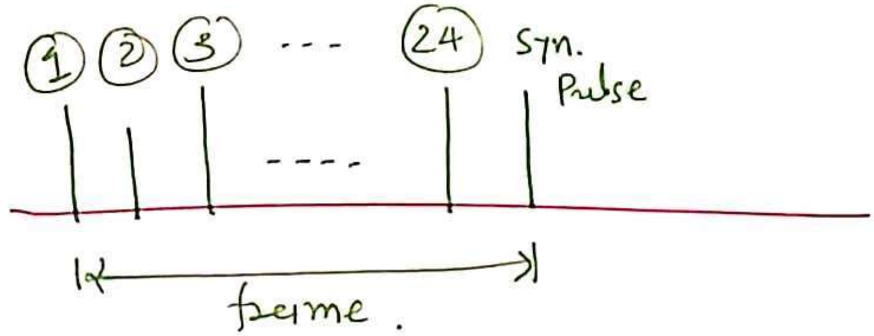
$f_c = 3.4\ kHz$

$m = 8$ bits

$L = 2^m = 2^8 = 256$ levels

- $f_s = 2B = 2 \times 3.4\ kHz = 6.4\ kHz$
- we prefer over sampling

$$\boxed{f_s = 8\ kHz}$$



- bit rate $= R_b = n f_s = 8 \times 8000 = 64$ Kbpsec
- Frame period $T_f = \frac{1}{f_s} = \frac{1}{8000} = 125$ µsec
- bits in one frame. $= 8(24) + 1 = 193$ bits.

(a) - bit duration $= \frac{T_f}{\text{No of bits}} = \frac{125\ \mu\text{sec}}{193} = 0.648\ \mu$sec

(b) - Transmission rate $R_b = 1/T_b = 1/0.648\ \mu\text{sec} = 1.544$ mbps

- bit rate $= R_b = n f_s = 8 \times 8000 = 64$ Kbpsec
- Frame period $T_f = \frac{1}{f_s} = \frac{1}{8000} = 125\ \mu sec$
- bits in one frame. $= 8(24) + 1 = 193$ bits.

(a) bit duration $= \frac{T_f}{\text{No of bits}} = \frac{125\ \mu sec}{193} = 0.648\ \mu sec$

(b) - Transmission rate $R_b = 1/T_b = 1/0.648\ \mu sec = 1.544$ mbps

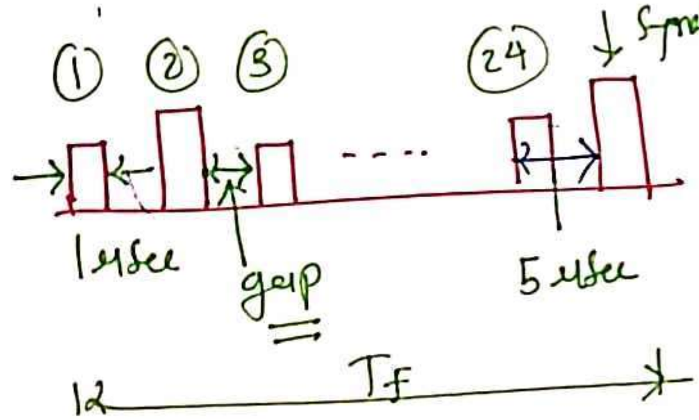
(c) - $R_b = 2B \Rightarrow B = R_b/2 = 1.544/2 = 0.772$ MHz or 772 kHz

24 Voice Signals are uniformly sampled and then multiplexed with TDM. The sampling operation uses flat Top samples with 1 μsec duration. For Synchronization an extra pulse of 1 μsec duration is added. The highest freq. of each voice signal is 3.4 KHz.

a) Calculate the spacing between succesive pulses of multiplexed signal by assuming a sampling rate of 8 KHz

b) Repeat for Nyquist rate of sampling.

- $T_f = \frac{1}{f_s} = \frac{1}{8000} = 125\ \mu sec$



- for 1 pulse time (without spacing)

$$= \frac{T_F}{\text{No of Pulses}}$$

$$= \frac{125\ \mu sec}{25}$$

$$= 5\ \mu sec$$

- Spacing bet.^{n} Pulses $= 5\ \mu sec - 1\ \mu sec$

$$= 4\ \mu sec.$$

- Spacing bet.^{n} Pulses = 5 μsec − 1 μsec
  = 4 μsec.

- $T_f = \frac{1}{f_s} = \frac{1}{6800} = 147.06\ \mu sec$

- For 1 pulse time without spacing

$$= \frac{T_F}{\text{No of Pulses}} = \frac{147.06}{25} = 5.88\ \mu sec$$

- Spacing = 5.88 − 1 = 4.88 μsec.

- Lowering sampling rate $\uparrow^{se}$ in Spacing bet.^{n} Successive Pulses.

Eight message signals are sampled & multiplexed with TDM. The TDM output is passed through a LPF before transmission. Six of the I/P signals have a BW of 4 KHz and the other two are band limited to 12 KHz.

(a) Identify the minimum sampling rate of composite signal. also find the overall sampling rate.

(b) Design an asynchronous TDM for this application. Assume commutator speed to be 8000 rotations/sec

(c) compare the transmission bandwidth requirement for for part (a) and (b)

- $\underbrace{x_1(t) \quad x_2(t)}_{12\ KHz} \quad \underbrace{x_3(t) \quad x_4(t) \quad \cdots \quad x_8(t)}_{4\ KHz.}$

- for composite signal consider highest bandwidth of signal.

$$f_s = 2W$$
$$= 2(12\ KHz)$$
$$= 24\ KHz$$

- overall sampling rate for 8 channels

$$= 8 \times f_s$$
$$= 8 \times 24 = 192000\ \text{samples/sec}$$

$f_s = 2W$

$= 2(12\ KHz)$

$= 24\ KHz$

- overall sampling rate for 8 channels

$= 8 \times f_s$

$= 8 \times 24 = 192000$ samples/sec

$\rightarrow f_s = 2B \Rightarrow B = \frac{f_s}{2}$

$\Rightarrow B = \frac{192000}{2}$

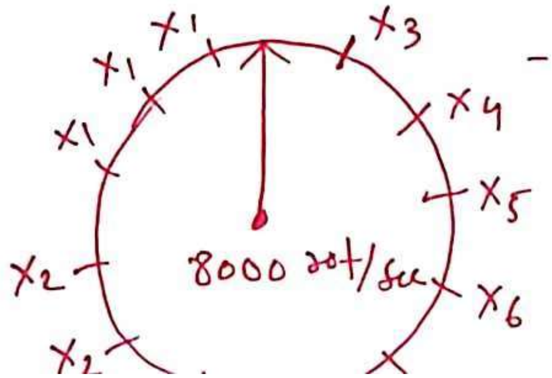
$\boxed{B = 96\ KHz}$

Commutator speed 8000 rot/sec.

$\underbrace{X_1 \quad X_2}_{12\ KHz} \qquad \underbrace{X_3 \quad X_4 \quad \cdots \quad X_8}_{4\ KHz}$

$\downarrow \qquad\qquad \downarrow$

$f_s = 2B \qquad f_s = 2B$



- Sampling rate

$= 12 \times 8000$

$= 96000$ samples/sec

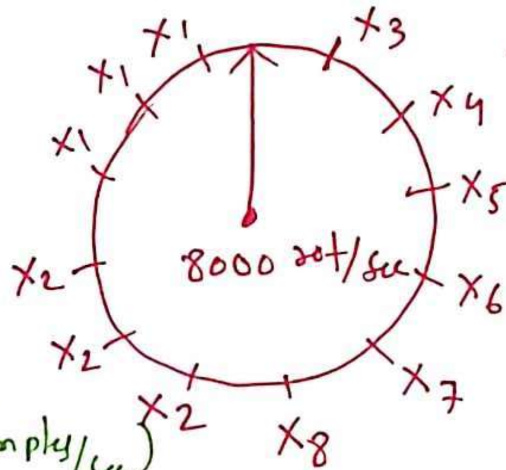
$= 8 \times 24 = 1$[illegible] /sec $\qquad \boxed{B = 96\ KHz}$

) Commutator speed 8000 rot/sec.

$$\underbrace{x_1 \quad x_2}_{12\ KHz} \qquad \underbrace{x_3 \quad x_4 \quad \cdots \quad x_8}_{4\ KHz}$$

$f_s = 2B$
$= 24\ KHz$ (samples/sec)

$f_s = 2B$
$= 8000$ (samples/sec)



- Sampling rate
$= 12 \times 8000$
$= 96000$ samples/sec

$\rightarrow B = \frac{f_s}{2}$
$= \frac{96000}{2} = 48\ KHz$