Ch-4 PULSE MODULATION & DEMODULATION :→

Chapter Coverage

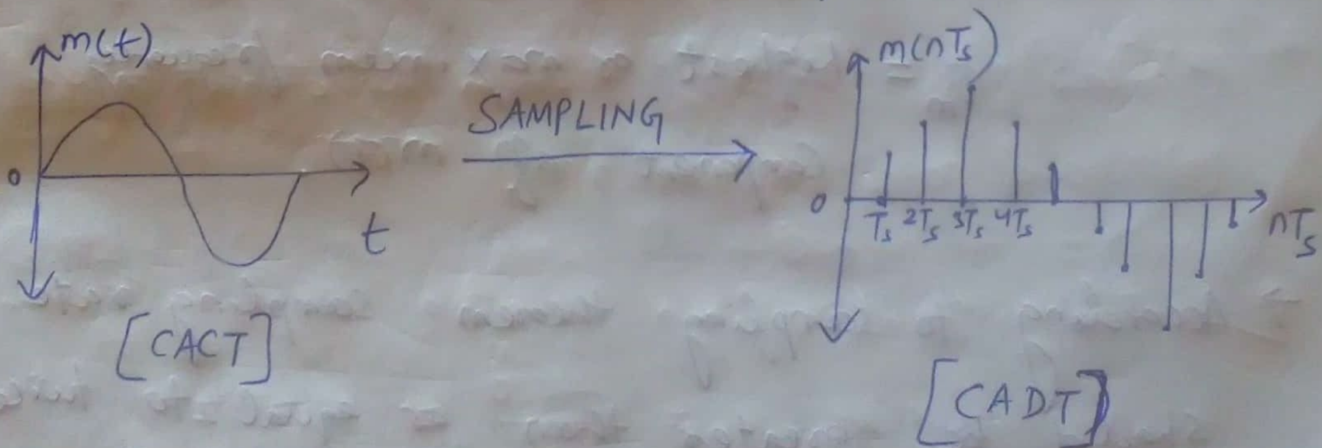
Sampling theorem (Low pass & [redacted]), natural sampling, flat top sampling, [redacted] [redacted], PAM, channel bandwidth for PAM, TDM, PWM, PPM.

SAMPLING: It is the process which converts a continuous amplitude continuous time (CACT) signal to continuous amplitude discrete time (CADT) signal.

→ CACT is analog signal → $m(t)$

CADT is discrete time signal → $m(nT_s)$

*



→ But SAMPLING process is done in accordance with a theorem called SAMPLING Theorem.

→ Sampling thm can be studied in two categories

(i) sampling theorem for Low pass signal

(ii) " " " Bandpass signal

(i) Sampling theorem for Lowpass signal :

"If the highest frequency contained in an analog signal $m(t)$ is f_m and the signal is sampled at a rate $f_s \geq 2f_m$, then $m(t)$ can be exactly recovered from its sample values."

→ Here f_s = sampling rate = sampling frequency

$\frac{1}{f_s} = T_s$ = sampling time interval

f_m = highest or maximum frequency component of $m(t)$.

→ According to sampling theorem sampling rate should be greater than or equal to twice the maximum frequency.

i.e $f_s \geq 2f_m$

$$\Rightarrow \frac{1}{T_s} \geq 2f_m$$

$$\Rightarrow \boxed{T_s \leq \frac{1}{2f_m}}$$

* Sampling are of 3 types

① oversampling if $f_s > 2f_m$

② critical sampling if $f_s = 2f_m$

③ Under sampling if $f_s < 2f_m$

* The sampling rate $f_s = 2f_{max} = 2f_m$ is called Nyquist rate or Nyquist frequency.

* Nyquist Interval $= T_s = \frac{1}{2f_{max}} = \frac{1}{2f_m}$

* Undersampling ($f_s < 2f_{max}$) is totally avoided in Communication.

~~(ii) Sampling theorem for bandpass signals:~~

~~"A bandpass signal with highest frequency f_H and bandwidth B, can be recovered from its samples through bandpass filtering by sampling it with frequency $f_s = \frac{2f_H}{K}$, where K is the largest integer not exceeding $\frac{f_H}{B}$."~~

~~→ All frequencies higher than f_s but below $2f_H$ may or may not be useful for band pass filtering sampling depending on overlap of shifted spectra.~~

(Q) A signal $m(t) = 2\cos 6000\pi t + 4\cos 8000\pi t + 6\cos 10000\pi t$ is to be truthfully represented by its samples. What is the minimum sampling rate from

(a) Low pass sampling theorem consideration and

~~(b) band pass consideration?~~

Sol^n^: The signal given has two frequency components.

$$\text{highest } f_H = \frac{10000}{2} = 5000\,Hz$$

$$\text{lowest } f_L = \frac{6000}{2} = 3000\,Hz$$

(a) Minimum sampling frequency from low pass consideration $= 2f_H = 2 \times 5000 = 10000$ sps

(b) $BW = f_H - f_L = 5000 - 3000 = 2000$ Hz

$k = \text{floor}\left(\frac{f_H}{B}\right) = \text{floor}\left(\frac{5000}{2000}\right) = \text{floor}(2.5)$
$= 2$

→ Then required sampling frequency

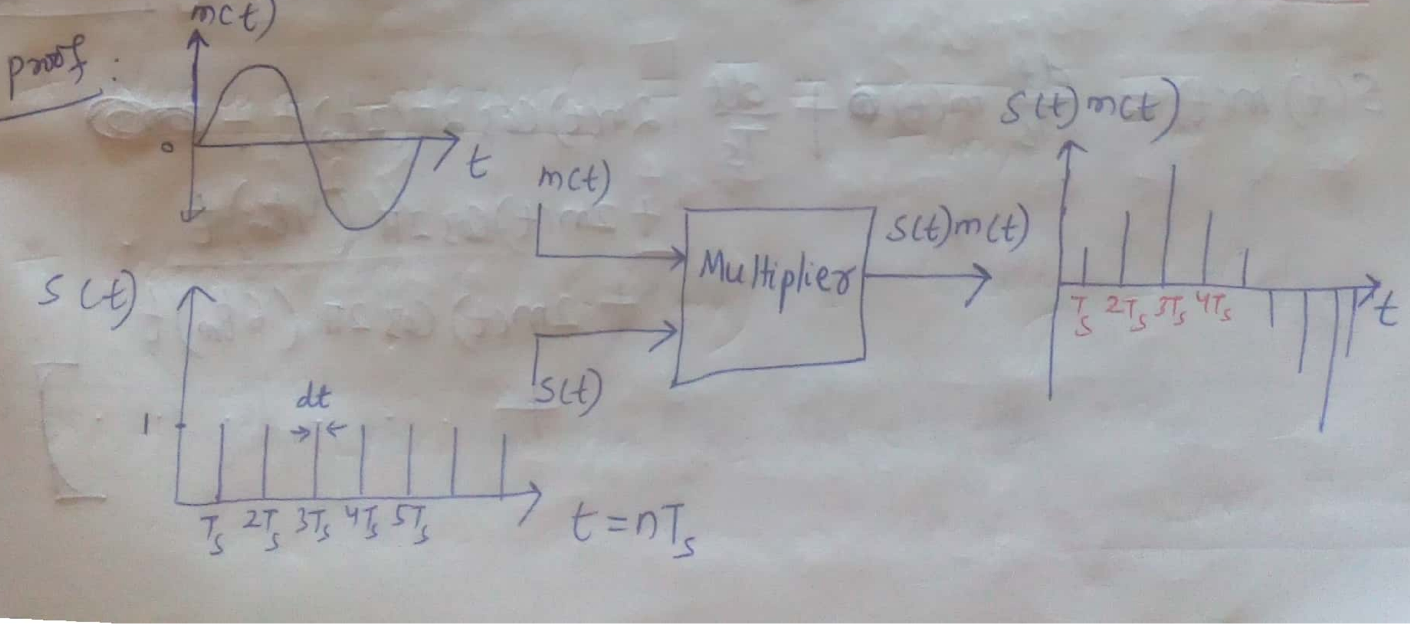
$= \frac{2f_H}{k} = \frac{2 \times 5000}{2} = 5000$ sps

sps = samples per second

* proof of Low pass sampling Theorem :

→ Lowpass sampling thm suggests to set sampling frequency $f_s \geq 2f_{max}$. Then why not $f_s < 2f_{max}$?

proof :



Here $m(t)$ = baseband message signal (CACT)

$s(t)$ = periodic pulse train of period T_s and width dt.

Here T_s = sampling time interval

$s(t)m(t)$ = Sampled signal (CADT)

In mathematics

$$s(t) = \frac{dt}{T_s} + \frac{2dt}{T_s}\left(\cos 2\pi \frac{t}{T_s} + \cos 2\times 2\pi \frac{t}{T_s} + \cdots\right)$$

sampled signal

$$s(t)m(t) = \frac{dt}{T_s} m(t) + \frac{dt}{T_s}\left(2m(t)\cos 2\pi \frac{t}{T_s} + 2m(t)\cos 2\times 2\pi\frac{t}{T_s} + \cdots\right)$$

$$= dt f_s m(t) + dt f_s \left[2m(t)\cos 2\pi f_s t + 2m(t)\cos 2\pi (2f_s)t + \cdots\right]$$

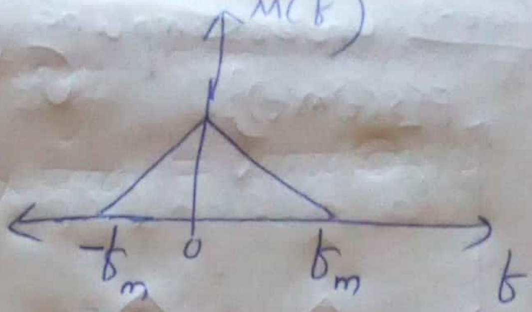
The important is to set the value of T_s

Case-1: Critical sampling $f_s = 2f_m \Rightarrow \frac{1}{T_s} = 2f_m \Rightarrow \boxed{T_s = \frac{1}{2f_m}}$

$$s(t)m(t) = \frac{dt}{T_s} m(t) + \frac{dt}{T_s}\left[2m(t)\cos 2\pi(2f_m)t + 2m(t)\cos 2\pi(4f_m)t + 2m(t)\cos 2\pi(6f_m)t + \cdots\right]$$

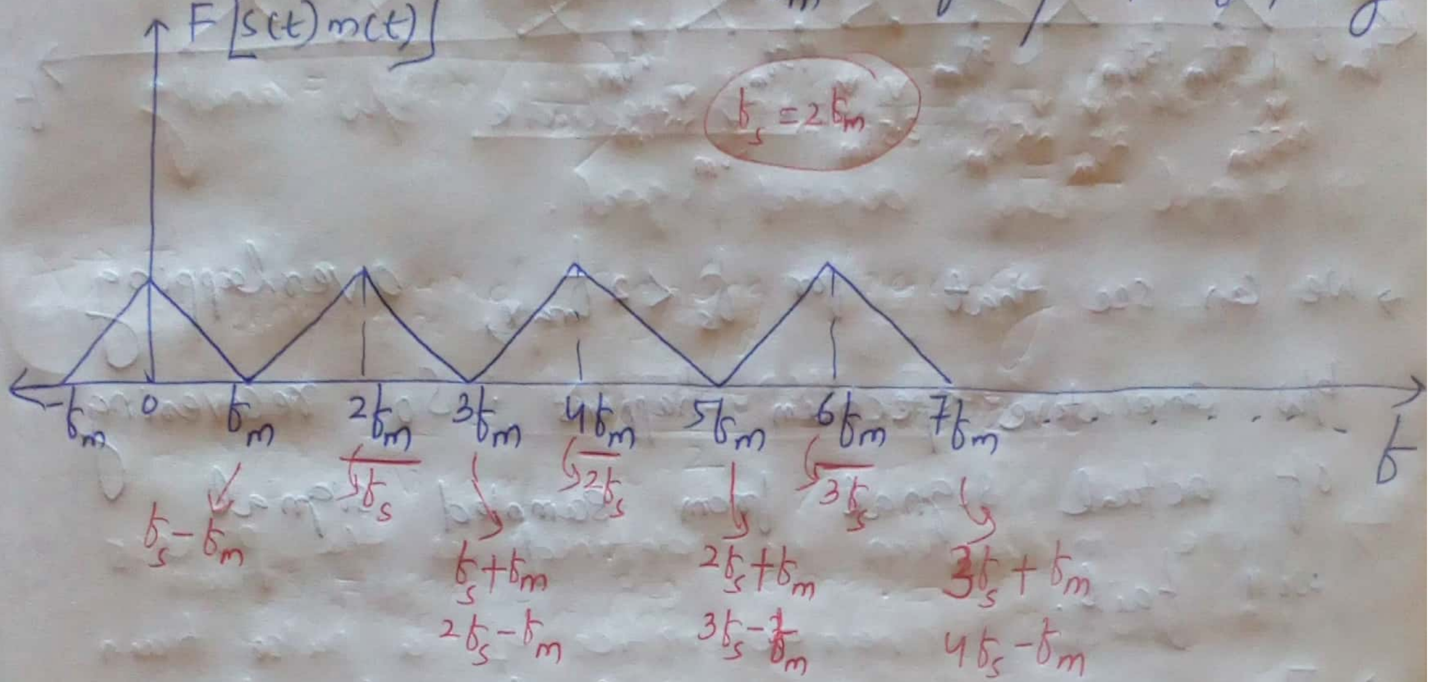
Spectral diagram :

Let $m(t)$ is such that



f_m = highest / max^m frequency

$f_s = 2f_m$



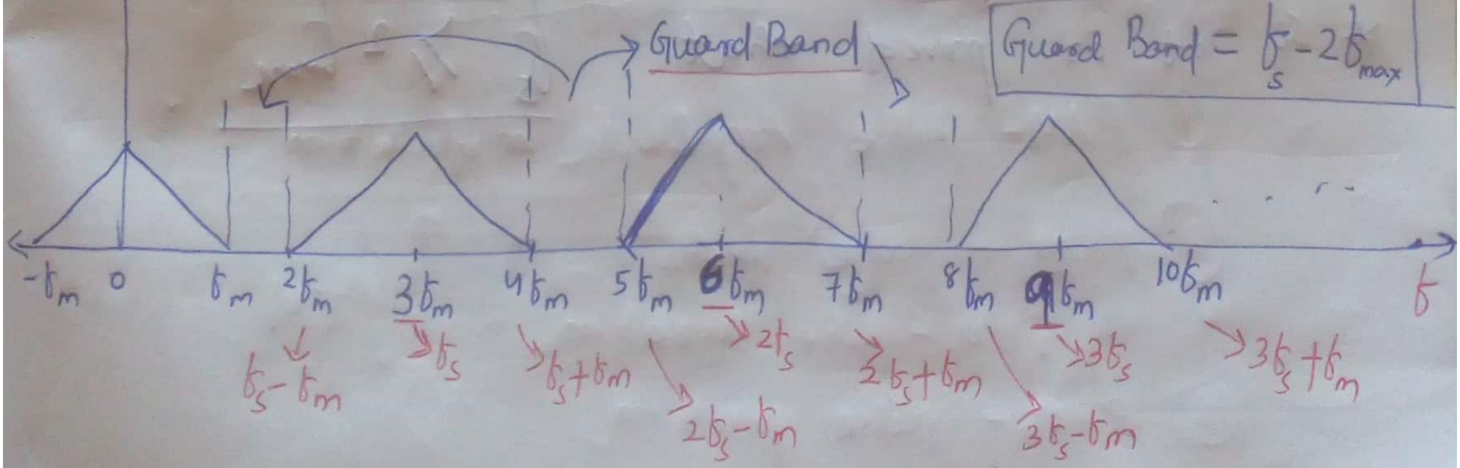
* From Case-1 Spectral diagram we can draw Spectral diagram of Case-2 ($f_s > 2f_m$) and Case-3 ($f_s < 2f_m$).

* Case-2 : oversampling ($f_s > 2f_m$ or $T_s < \frac{1}{2f_m}$)

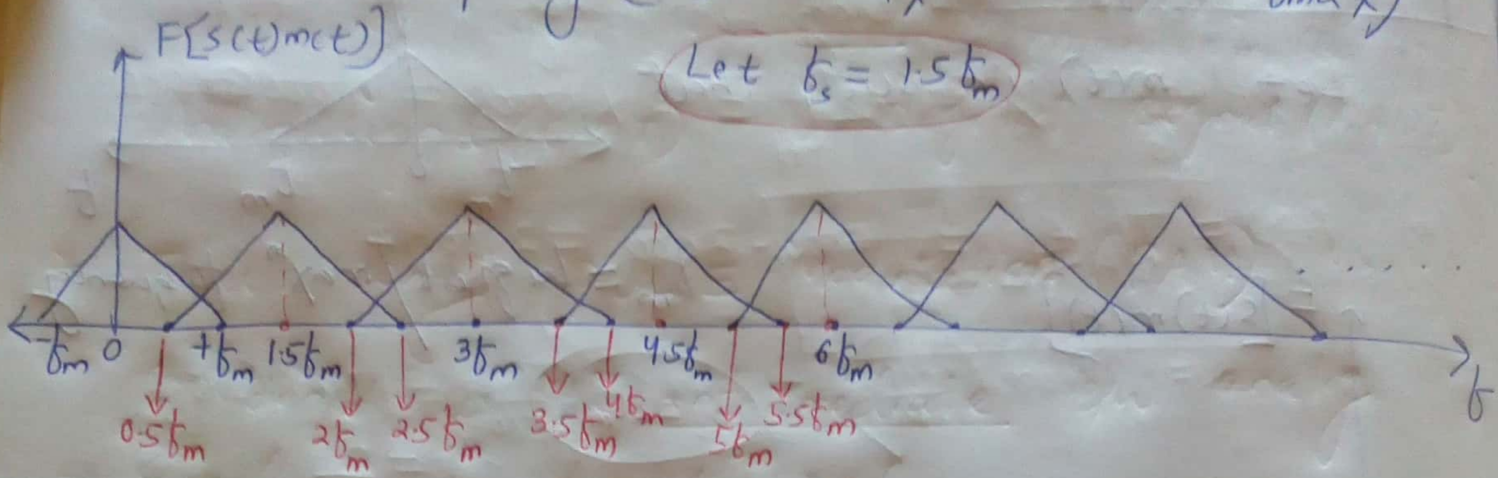
$F[s(t)m(t)]$

Let $f_s = 3f_m$

* $\boxed{\text{Guard Band} = f_s - 2f_{max}}$



Case-3: Undersampling ($f_s < 2f_{max}$ or $T_s > \frac{1}{2f_{max}}$)



→ We can see that for $f_s < 2f_{max}$, overlapping b/w consecutive spectrum happens. So recovering of actual signal from sampled signal will fail.

→ ~~The effect~~ The signal loss due to overlapping of consecutive spectrum in case of $f_s < 2f_{max}$ is known as ALISING.

* To remove alising, we use Anti ALising Filter before Sampling.

* So it is proved that $f_s \geq 2f_{max}$.

* Sampling technique is a pulse Modulation Technique but with a rule $f_s \geq 2f_{max}$.

In communication we will study 3-types of Conventional modulation techniques.

1. Analog Modulation → AM; FM, PM (Here baseband = analog, carrier = analog sinusoidal)

ACT (4th sem)

2. Pulse Modulation → PAM; PWM; PPM (Here baseband = analog, carrier = square pulse signal)

DCT (5th sem)

3. Digital modulation (Here baseband = digital, carrier = sinusoidal)

* Sampling process is a PAM-type of pulse Modulation.

* Sampling is of 3-types / PAM is of 3-types

1. Ideal / Instantaneous Sampling / PAM
2. Natural Sampling / PAM
3. Flat-Top Sampling / PAM

* Sampling process is used to convert Analog Signal (CACT) to Digital Signal (DADT)

* That's why PULSE Modulation (PAM) is considered as the intermediate b/w analog signal and digital signal.

PULSE MODULATION :

→ pulse modulation is defined as the process in which characteristics of a pulse train are changed according to baseband signal.

→ Here carrier = ~~[illegible]~~ pulse train
message = analog signal

→ Like analog modulation pulse modulation is of 3 - types.

1. PAM - pulse Amplitude modulation
2. PWM - pulse width "
   or PDM - pulse Duration "
3. PPM - pulse position "

$\left.\begin{matrix} \text{PWM} \\ \text{PDM} \\ \text{PPM} \end{matrix}\right\}$ PTM

* PWM or PDM & PPM togetherly called as pulse Time Modulation (PTM).

PAM

Def^n: In PAM, Amplitude of pulse train will be changed according to baseband signal.

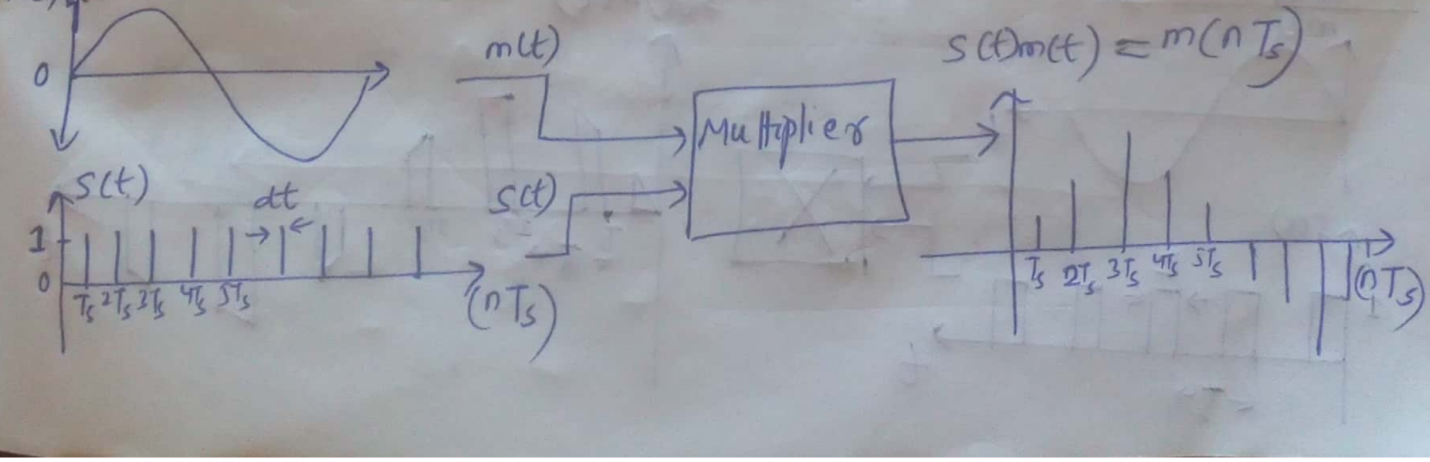
→ PAM signals are known as Sampled signal if $f_s \geq 2 f_{max}$.

→ 3-types of PAM signal.

1. Ideal / Instantaneous PAM
2. Natural PAM
3. Flat Top PAM

(i) Ideal PAM / Ideal Sampling:

Generation of Ideal PAM:



→ It is called instantaneous PAM or sampling because width of pulse train is infinitesimally small i.e dt.

Here sampling interval $= T_s$

→ We can write $s(t)\, m(t) = m(nT_s)$

→ $s(t)$ can be expressed as

$$s(t) = \frac{dt}{T_s} + \frac{2dt}{T_s}\left(\cos 2\pi f_s t + \cos 2\times 2\pi f_s t + \cdots\right)$$

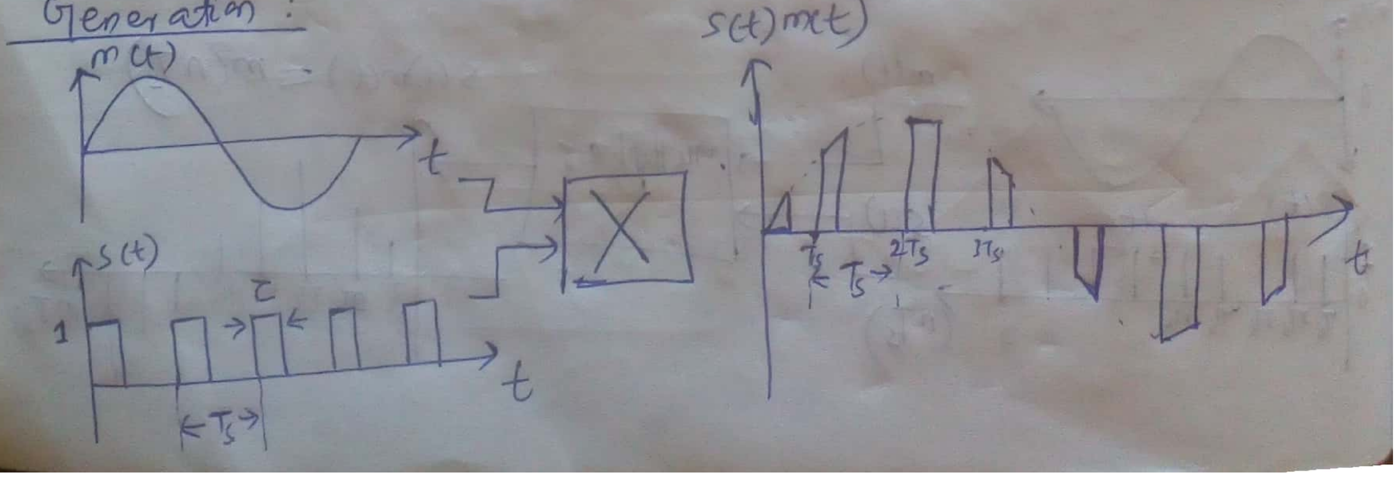
Limitations of Ideal PAM :

(i) Difficult to construct such ideal pulse train of very short time dt.

(ii) In this sampling, each sampled signal contains infinitesimal energy so when this signal will be communicated it will be lost by external noise.

(ii) Natural PAM / SAMPLING :

Generation :



Here

$$s(t) = \frac{\tau}{T_s} + \frac{2\tau}{T_s}\left(C_1 \cos 2\pi f_s t + C_2 \cos 2\cdot 2\pi f_s t + \cdots\right)$$

where

$$C_n = \frac{\sin\left(\frac{n\pi\tau}{T_s}\right)}{(n\pi\tau/T_s)}$$

→ Natural Sampled signal

$s(t)m(t)$

$$= \frac{\tau}{T_s} m(t) + \frac{2\tau}{T_s}\left[m(t) C_1 \cos 2\pi f_s t + m(t) C_2 \cos 4\pi f_s t + \cdots\right]$$

→ To demodulate message signal $m(t)$ from $s(t)m(t)$ we have to use a Lowpass Filter.

then LPF will result

$$s_0(t) = \frac{\tau}{T_s} m(t)$$

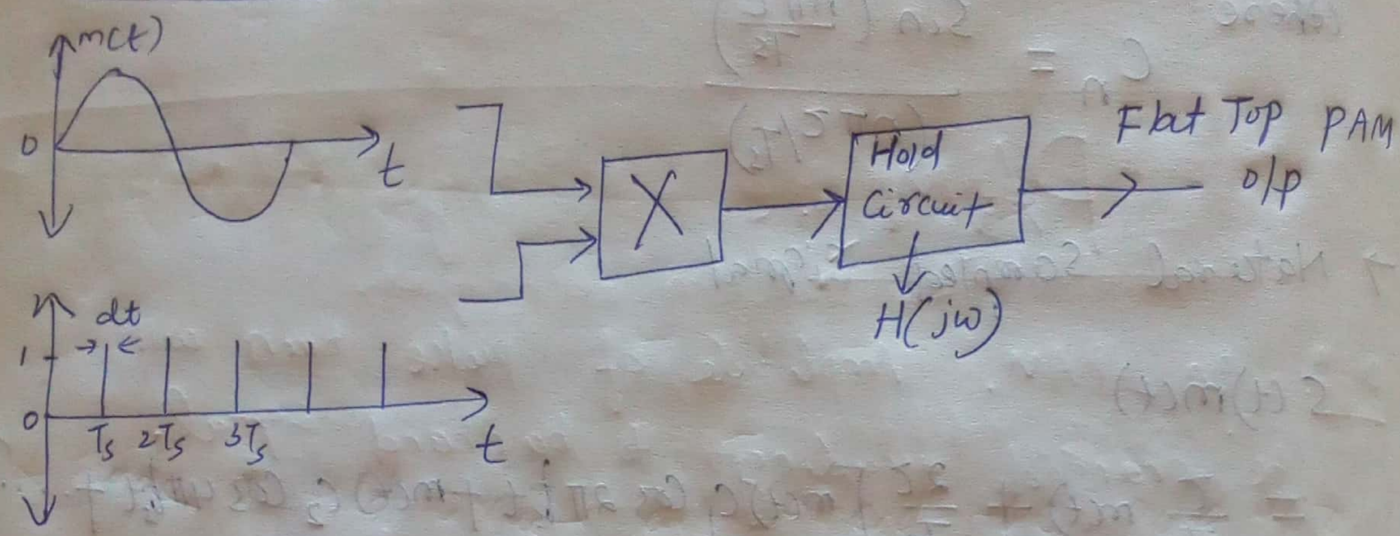
Limitation of Natural PAM :

→ The sampled signal of Natural PAM has top part not constant rather it follows the waveform of the signal which is not frequently employed.

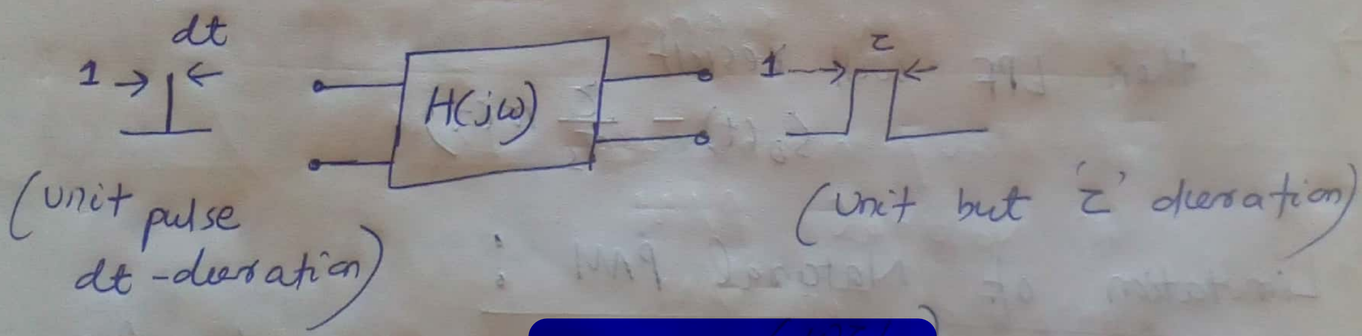
→ Due to nonuniform top electronic circuits face difficulty to process it.

(iii) Flat Top PAM / Flat-Top Sampling :-

Generation :

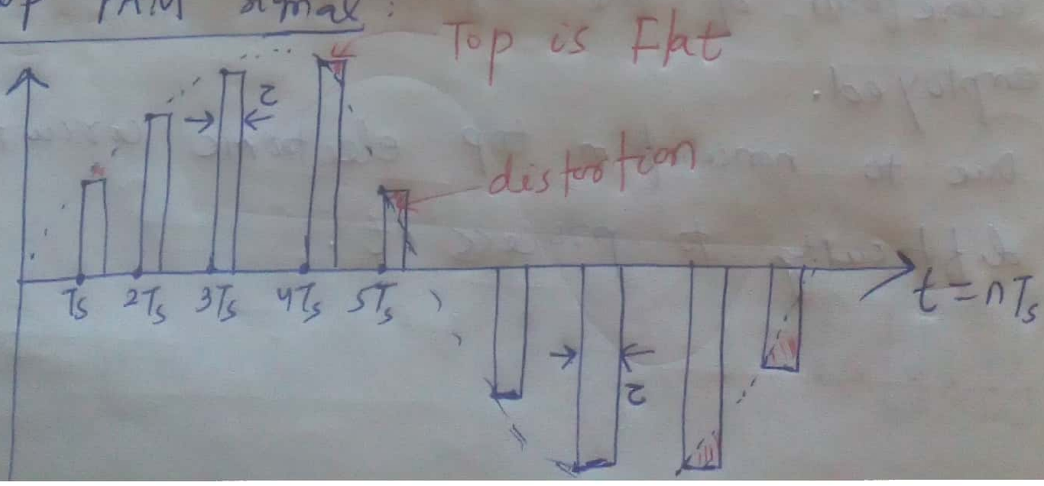


* $H(j\omega)$ = Transfer function of Hold circuit. It helps to extend the time duration of pulse.



[redacted]

→ Flat Top PAM signal :



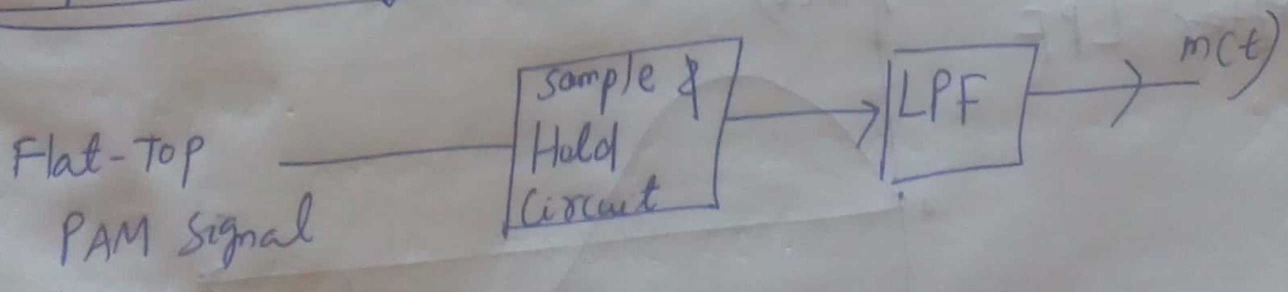
→ The gap b/w original signal and flat-top signal is appered as a distortion. It is known as Aperture Effect Distortion.

→ More the width of the top (τ) more will be distortion. If in order to reduce distortion 'τ' will be reduced then signal strength will also be decreased which is a challenge to mantain desired SNR.

→ So not ~~[illegible]~~ compormising 'τ' value and to reduced distortion, we will use an Equalizer at the receiver side.

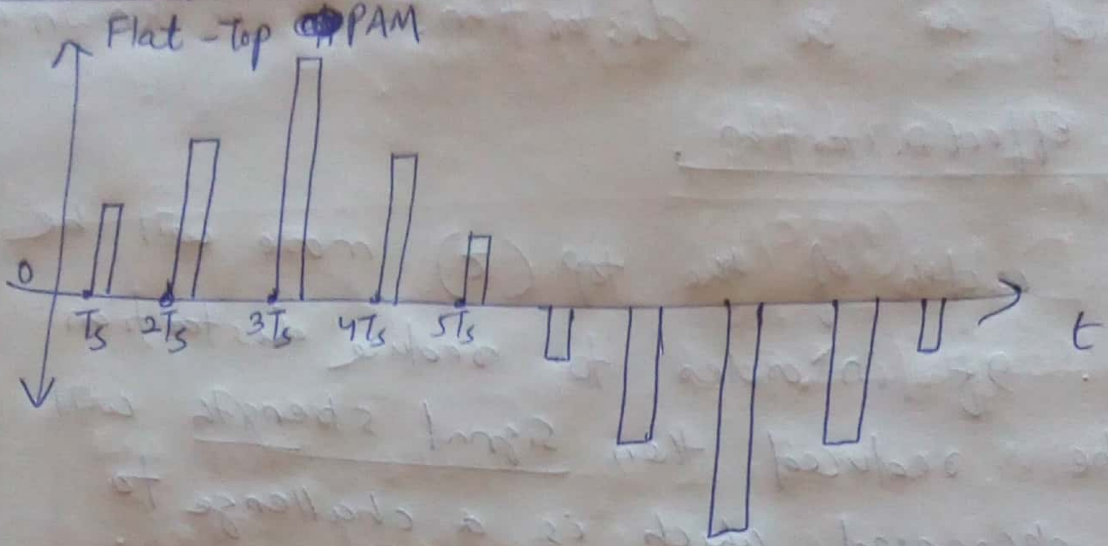
* To recover m(t) from Flat-Top Sampled signal, only Lowpass Filter is not sufficient

* Signal Recovery through Holding:

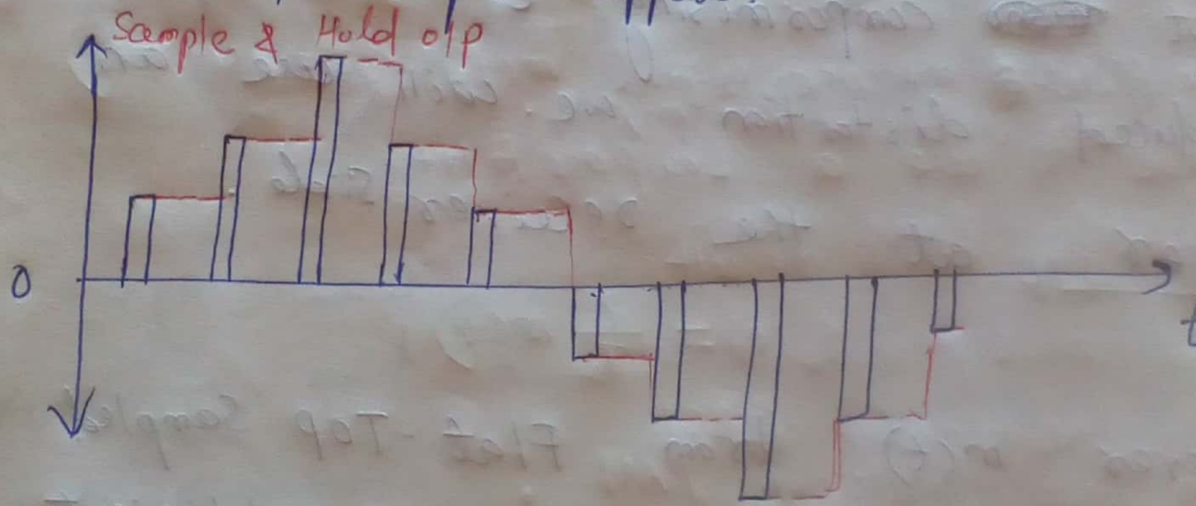


Scanned by CamScanner

operation of Recovery.

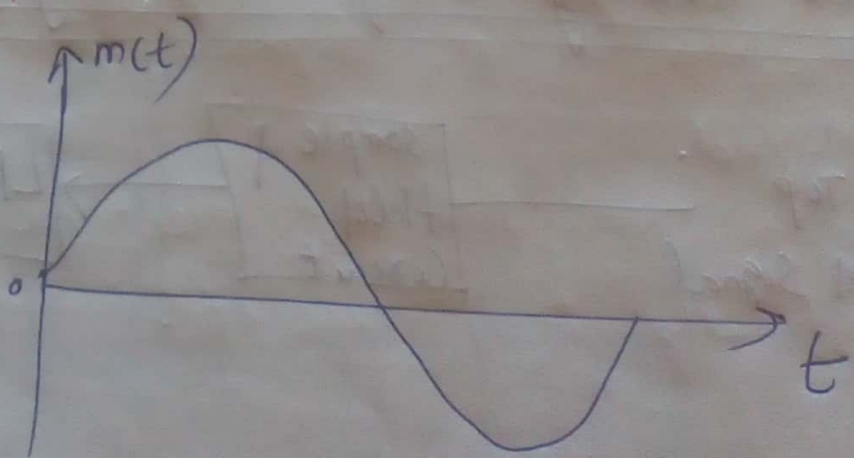


* sample & Hold circuit will extend the pulse upto the next sample appers.

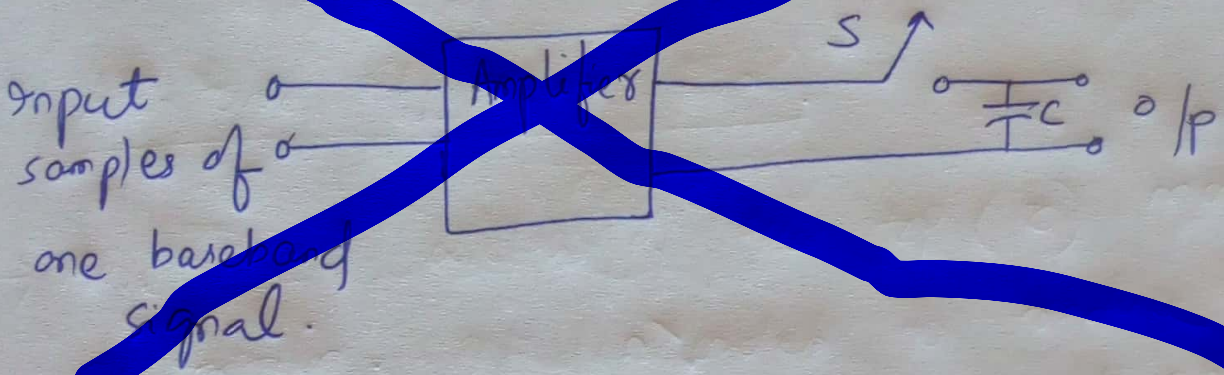


* The Red lined is the output of Sample & Hold Ckt.

* After LPF



~~* Hold ckt uses a Capacitor as below~~



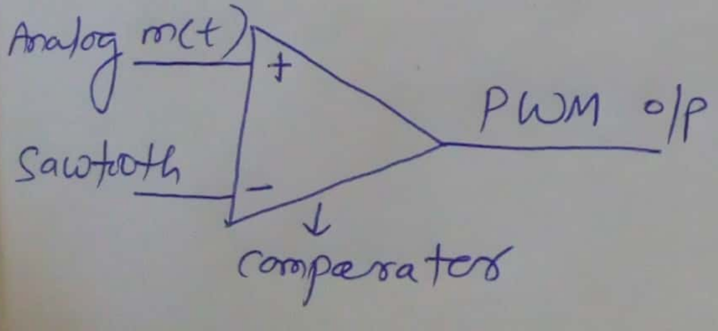
PWM & PPM :

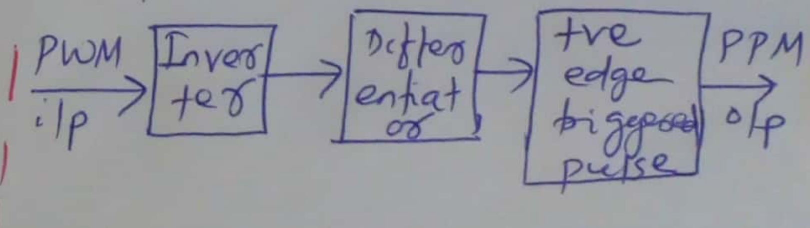
PWM → In pulse width modulation, width of pulse will be changed according to message signal.

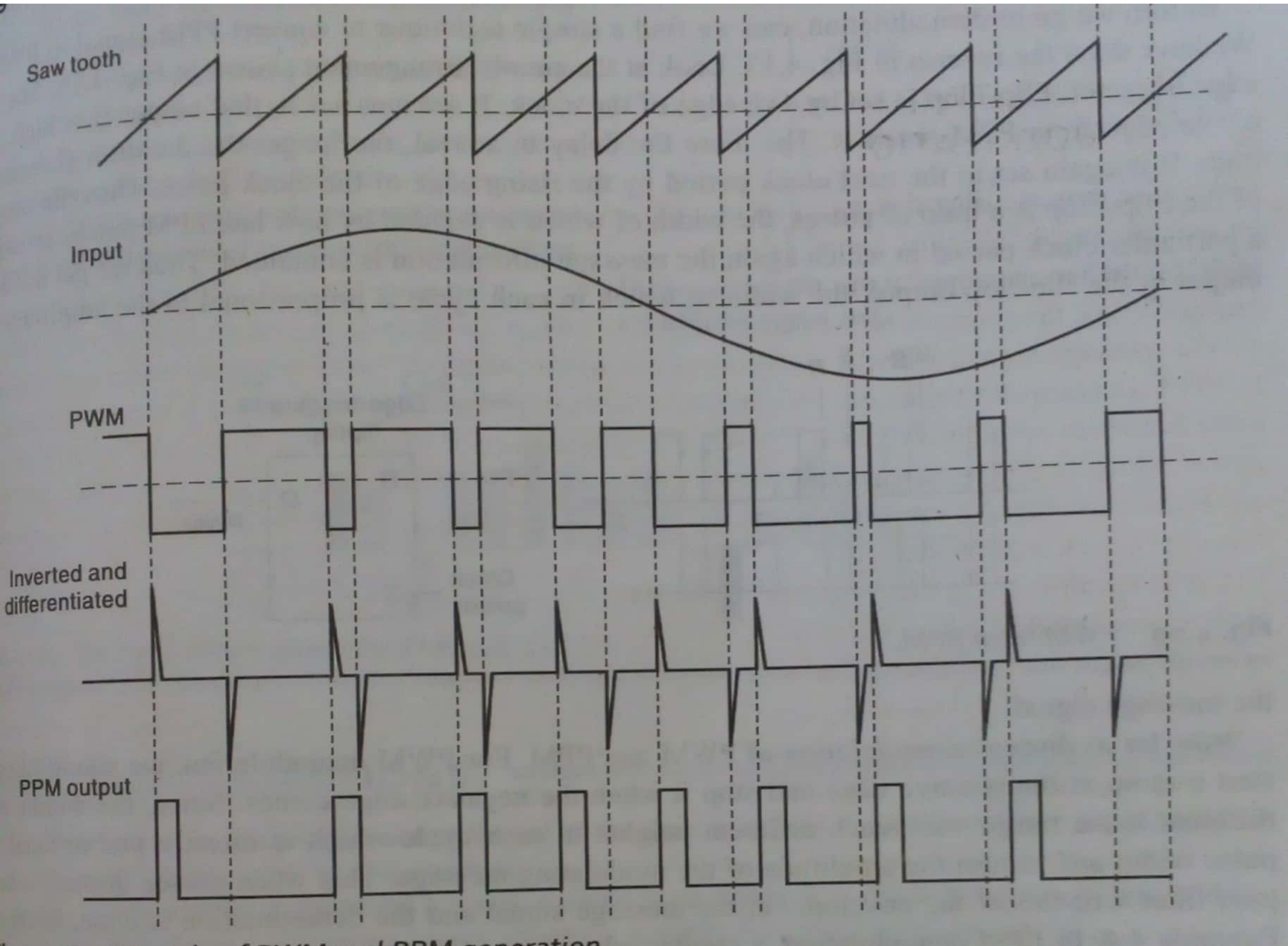
PPM → In pulse position modulation, the time position of occurance of pulse will be changed according to the message signal.

Generation of PWM & PPM :

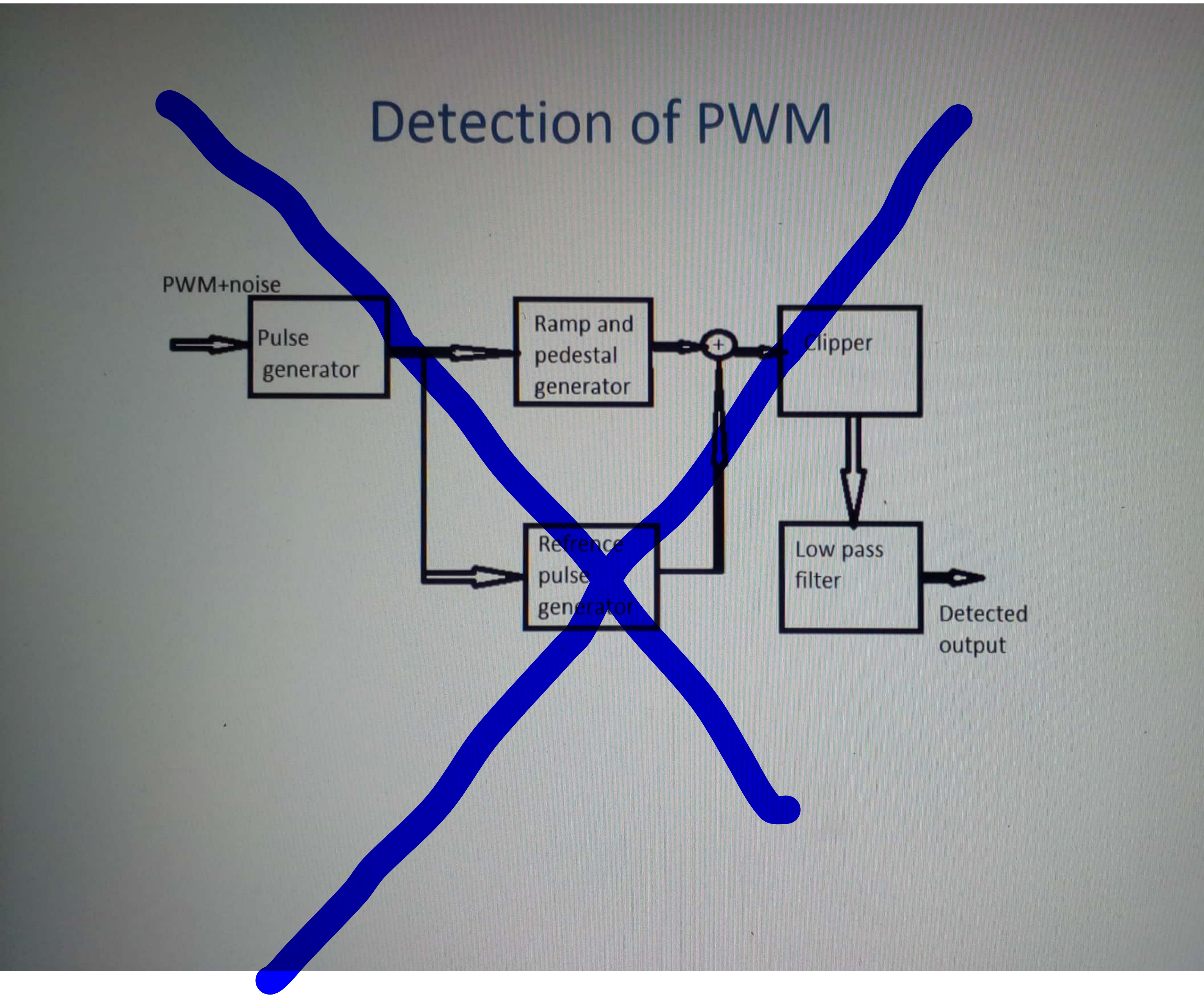
Block Dgm :

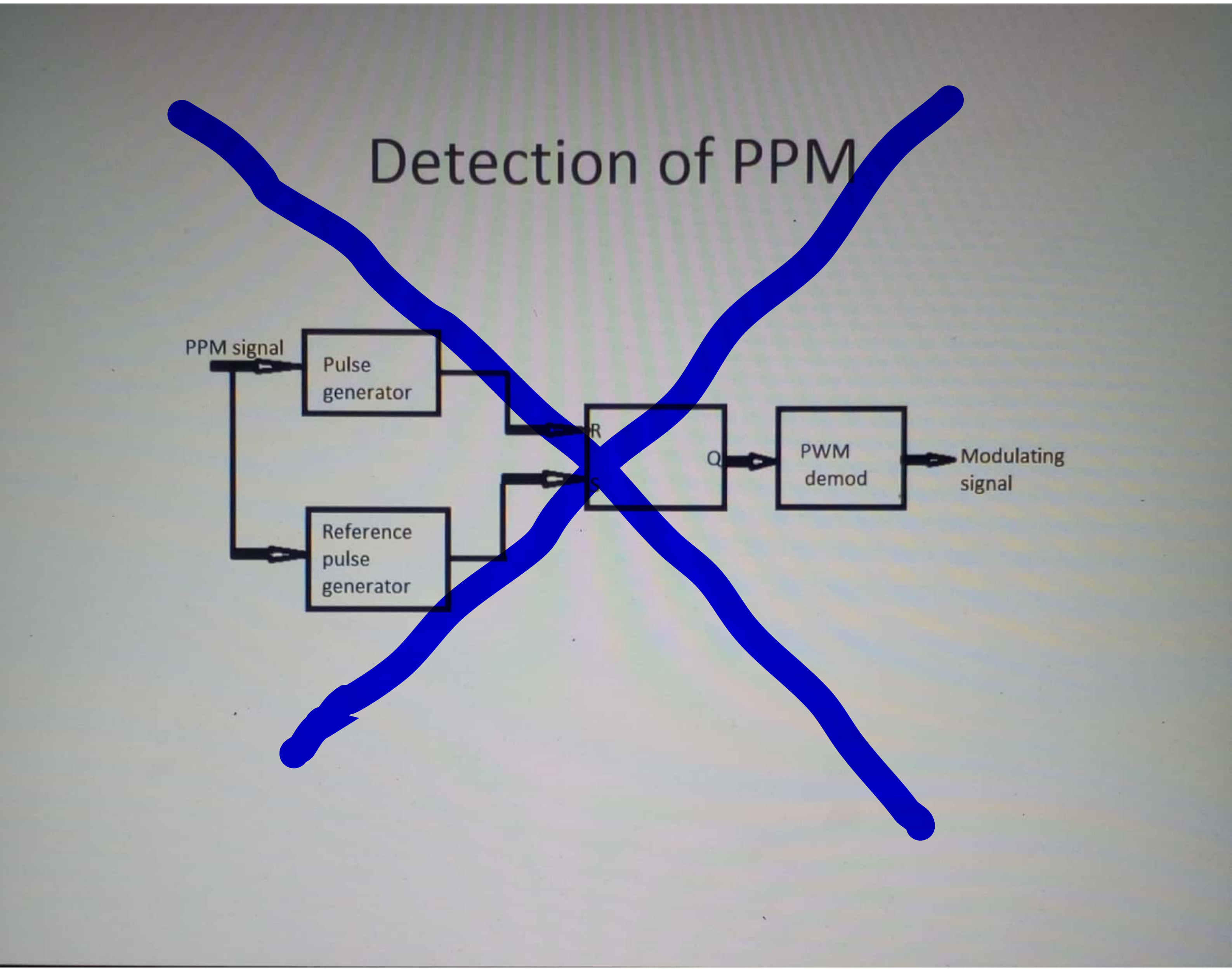






ig. 4.18 *Principle of PWM and PPM generation.*


Detection of PWM
PWM+noise
Pulse generator
Ramp and pedestal generator
+
Clipper
Refrence pulse generator
Low pass filter
Detected output


Detection of PPM
PPM signal
Pulse generator
Reference pulse generator
R
S
Q
PWM demod
Modulating signal

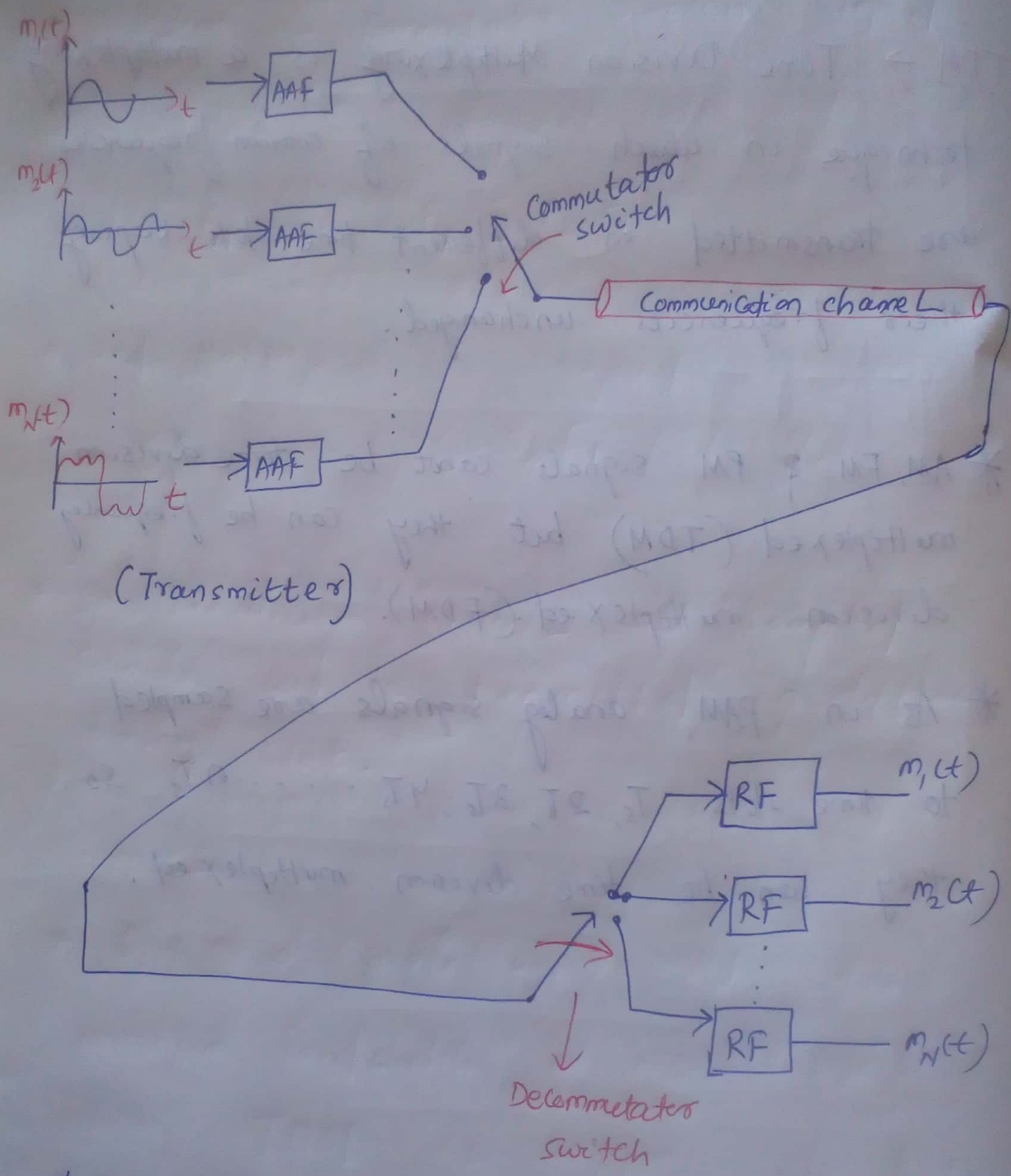
PAM - TDM :

TDM → Time Division Multiplexing is a multiplexing technique in which signals of common frequencies are transmitted in different time slots keeping their frequencies unchanged.

* AM, FM & PM signals canot be time division multiplexed (TDM) but they can be frequency division multiplexed (FDM).

* As in PAM, analog signals are sampled to time slots T_s, $2T_s$, $3T_s$, $4T_s$ ..... nT_s, so they can be time division multiplexed.

Block Diagram of PAM-TDM :



* Here AAF - Anti Aliasing Effect

RF - Reconstruction Filter

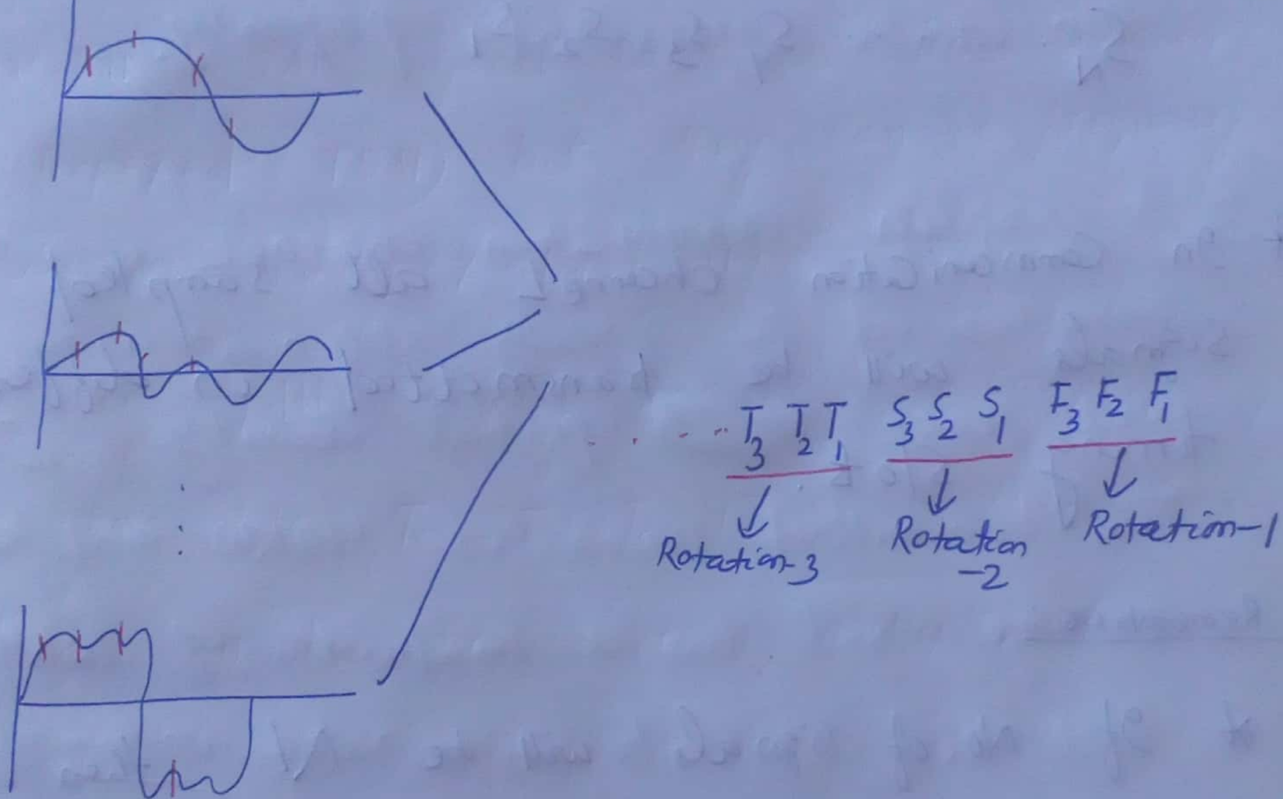
Scanned by CamScanner

→ AAF will bandlimit the input signals.

→ Commutator switch will rotate to cut samples from each signal.

<u>Explanation</u>

Let signals are



* The red spots are samples of each signal.

F_1, F_2, F_3 → First samples of Signal 1,2,3 (Let)

S_1, S_2, S_3 → Second samples of ,,

T_1, T_2, T_3 → Third ,, ,, ,,

→ If 'N' number of signals, then in First rotation of Commutator switch sampled signals will be

$$F_N \ldots\ldots F_4\ F_3\ F_2\ F_1.$$

In 2nd rotation

$$S_N \ldots\ldots S_4\ S_3\ S_2\ S_1$$

* In Communication channel all sampled signals will be transmitted in different timing slots.

~~Remember:~~

~~* If No. of signals will be N, then minimum bandwidth of all signals after TDM is~~ $\cancel{Nf_m}$ ~~if~~ $\cancel{f_m}$ ~~is the highest frequency of each signal after AAF.~~