ACS

UNIT-IV

Analog Pulse Modulation Schemes

- Pulse Amplitude Modulation
- Natural sampling
- Flat-top sampling
- Pulse Amplitude modulation (PAM) and demodulation
- Pulse Time modulation
- Pulse Duration and Pulse Position Modulation & demodulation schemes
- PPM spectral analysis
- illustrative Problems.

Radio Receiver Measurements

- sensitivity
- Selectivity

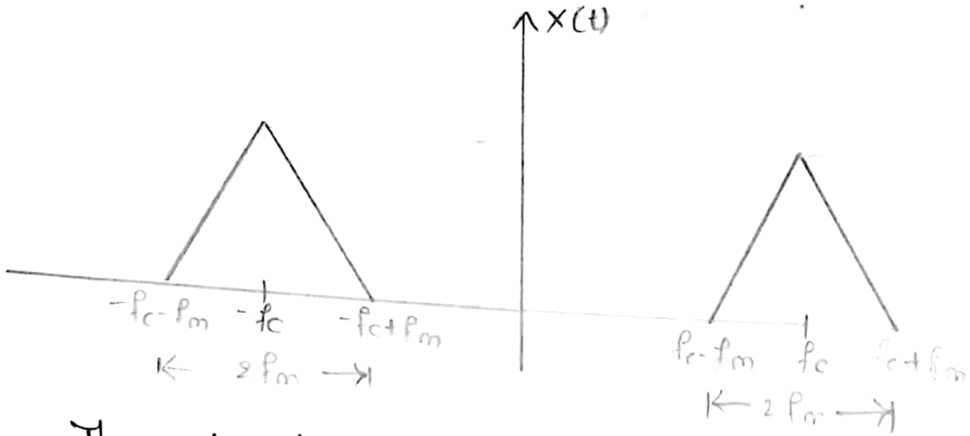
&

- Fidelity

UNIT- IV

Q: State and prove sampling theorem for band pass signals.

Ans:- A band pass signal x(t) whose maximum bandwidth is 2W can be completely represented and recovered from its samples if sampling frequency is equal to twice the bandwidth.



The band pass signal represented in terms of its inphase and quadrature components. The bandwidth of the band pass signal is $2f_m$ and it is centered at 'f_c'. The frequencies present in the bandpass signal are from f_c-f_m to f_c+f_m & $f_c > f_m$.

$x_I(t)$ — inphase component of x(t)

$x_Q(t)$ — Quadrature component of x(t)

$$\therefore \quad x(t) = x_I(t)\cos 2\pi f_c t - x_Q(t)\sin 2\pi f_c t \rightarrow (1)$$

The inphase and quadrature components are obtained by multiplying x(t) by $\cos 2\pi f_c t$ & $\sin 2\pi f_c t$ & suppressing the sum frequencies by means of LPF. These components contain only low frequency components.

After some mathematical manipulation on equation (1), we obtain the reconstruction formulae as

$$x(t) = \sum_{n=-\infty}^{\infty} x\left(\frac{n}{4f_m}\right) \operatorname{sinc}\left(2f_m t - \frac{n}{2}\right) \cos\left(2\pi f_c\left(t - \frac{n}{2f_m}\right)\right)$$

If 4W samples/sec are taken, then the band pass signal B.W 2W can be completely recovered from its samples.

$\therefore$ Minimum sampling rate = Twice of B.W
$= 2(2W) = 4W$

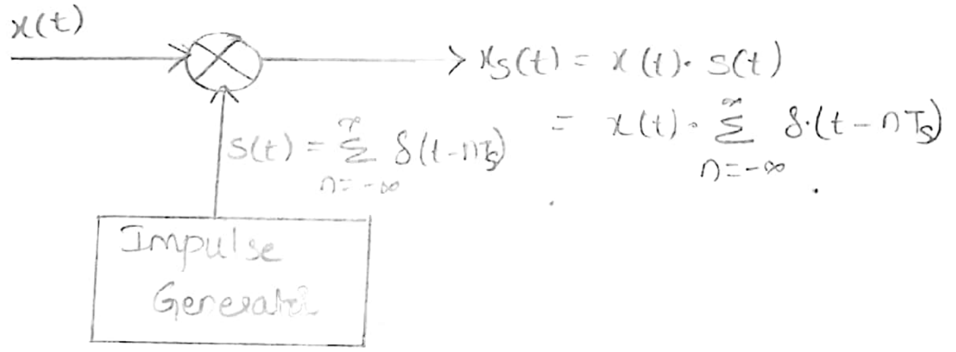
$$x\left(\frac{n}{4f_m}\right) = x(nT_s) \quad \text{where} \quad T_s = \frac{1}{4f_m}.$$

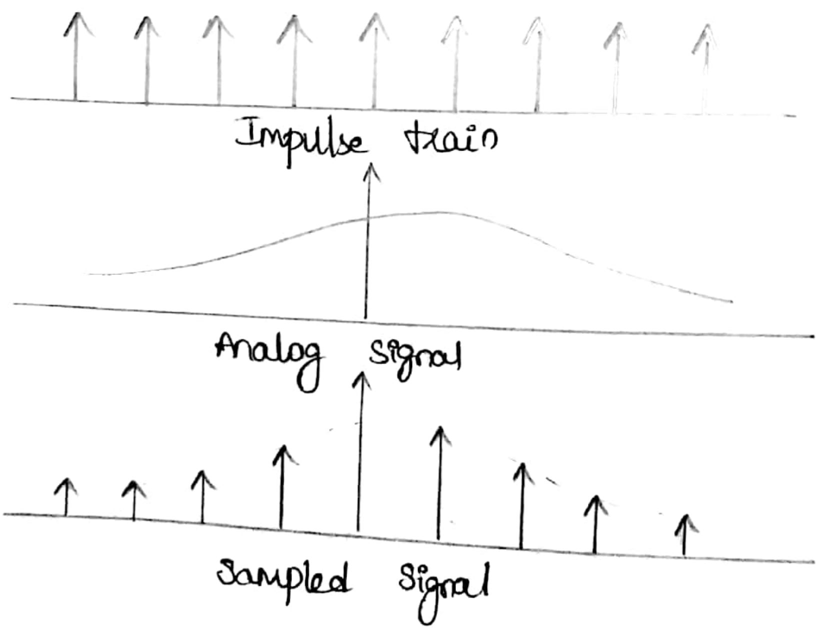
Q: State and prove the ideal sampling and reconstruction of low pass signals with neat sketches.

A:- Sampling theorem statement:

A continuous-time signal can be completely represented in its samples and recovered back if the sampling frequency is twice of the highest frequency content of the signal. i.e $f_s \geq 2f_m$.

Proof:-





$x(t) \rightarrow$ input signal to the sampler

$x_\delta(t) \rightarrow$ sampled signal of $x(t)$

$$x_\delta(t) = \sum_{n=-\infty}^{\infty} x(t) \cdot \delta(t - nT_s) \rightarrow (1)$$

Applying fourier transform to the above equation

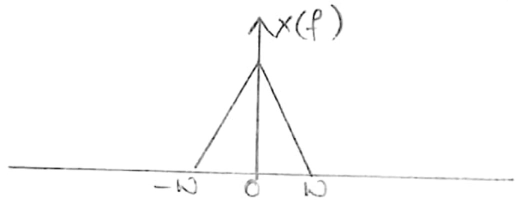
$$x_\delta(t) \xleftrightarrow{FT} X_\delta(f)$$

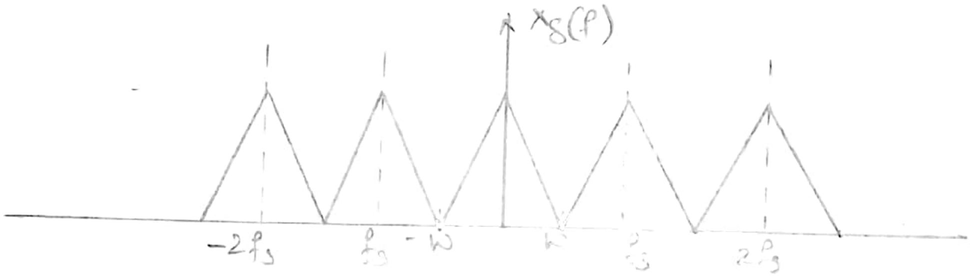
$$x(t)\delta(t-nT_s) \xleftrightarrow{FT} X(f) * f_s\, \delta(f - nf_s)$$

$$x(t)\, \delta(t - nT_s) \longleftrightarrow f_s \cdot X(f - nf_s)$$

$$X_\delta(f) = f_s \cdot \sum_{n=-\infty}^{\infty} X(f - nf_s)$$

$$= \cdots\cdots + f_s \cdot X(f+2f_s) + f_s X(f+f_s) + f_s X(f) + f_s X(f-2f_s) + f_s X(f - f_s) + \cdots\cdots$$





Reconstruction of $x(t)$ from its samples:

$$x(t) = \text{IFT}[X(f)]$$

$$= \int_{-\infty}^{\infty} X(f) \cdot e^{j2\pi ft}\, df$$

$$= \int_{-\infty}^{\infty} \left[\frac{1}{f_s} \sum_{n=-\infty}^{\infty} x(nT_s)\, e^{-j2\pi f nT_s}\right] \cdot e^{j2\pi ft}\, df$$

$$= \sum_{n=-\infty}^{\infty} x(nT_s) \frac{1}{f_s} \int_{-W}^{W} e^{j2\pi f(t - nT_s)}\, df$$

$$= \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \frac{1}{f_s} \left[\frac{e^{j2\pi f(t-nT_s)}}{j2\pi(t-nT_s)} \right]_{-w}^{w}$$

$$= \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \frac{1}{f_s} \left[\frac{e^{j2\pi w(t-nT_s)} - e^{-j2\pi w(t-nT_s)}}{j2\pi(t-nT_s)} \right]$$

$$= \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \frac{1}{f_s} \left[\frac{\sin 2\pi w(t-nT_s)}{\pi(t-nT_s)} \right]$$

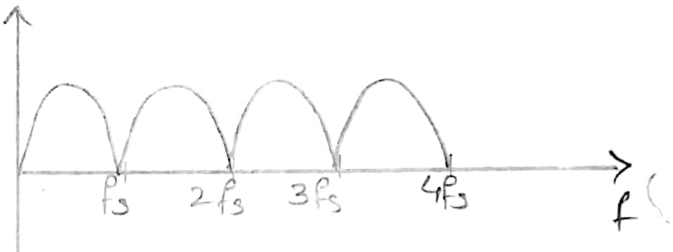
$$= \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \frac{\sin 2\pi w(t-nT_s)}{\pi(f_s t - f_s nT_s)}$$

$$x(t) = \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \frac{\sin\pi(2wt-n)}{\pi(2wt-n)} \quad ; \quad \left\{ \begin{array}{l} f_s = 2w \\ T_s = \frac{1}{2w} \end{array} \right.$$

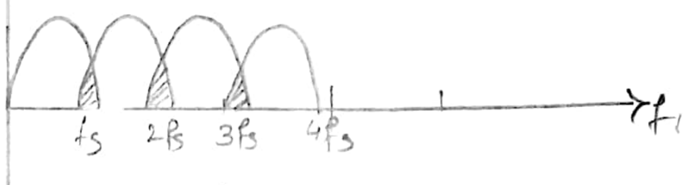
Q: What is aliasing Effect? How it can be eliminated? Explain with neat diagram.

Aliasing Effect:

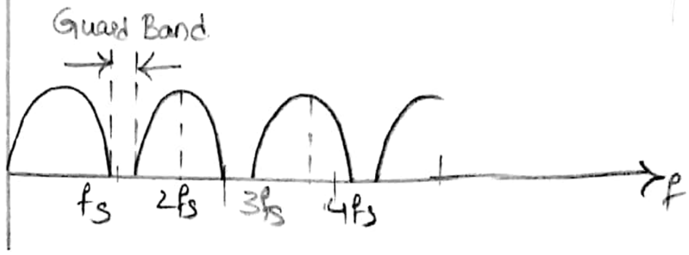
The spectrum of sampled signal overlap with neighboring ones. When a band limited signal is sampled at rate lower than nyquist rate, $f_s < 2f_m$ (or) sampling interval is higher than nyquist interval (T_s) as shown in figure. The aliasing effect is not that serious in under sampling process, only some aliasing is produced in this process.



$f_s = 2f_m$



$f_s < 2f_m$



$f_s > 2f_m$

Aliasing is the phenomenon in which the higher frequency components are combined with lower frequency components in the spectrum of its sampled version.

In order to eliminate the effect of aliasing, we may use prior to sampling a low pass (anti-aliasing) filter and then filtered signal is sampled at a rate slightly higher than the nyquist rate. ($f_s \geq 2f_m$)

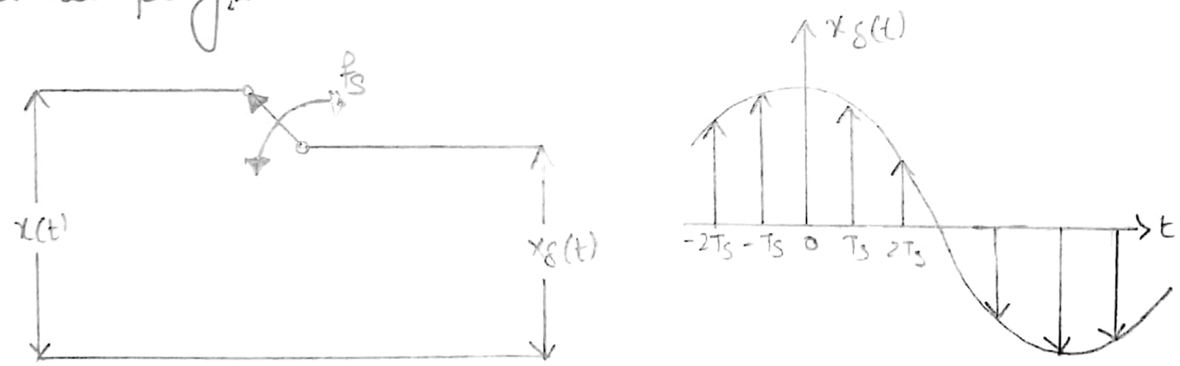
Hence, the sampling rate should atleast be equal to twice the maximum frequency component present in the signal $m(t)$. This means that, atleast two sample per cycle are needed for a complete recovery of the signal from $s(t)$. Thus, the minimum sampling rate is given by

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

Q: Discuss about different types of sampling. (8)
Differentiate between ideal sampling and Practical sampling.

A:- There are 3 types of sampling : (1) Ideal sampling
(2) Natural sampling
(3) Flat-top sampling

Ideal sampling:-



It uses switching sampler. If closing time 't' of the switch approaches zero the output $x_\delta(t)$ gives only instantaneous value.

The width of the pulse approaches zero, the instantaneous sampling gives train of impulses $x_\delta(t)$. The area of each impulse in the sampled version is equal to instantaneous value of input signal $x(t)$.

The train of impulses can be represented mathematically as

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

This is called sampling function.

The sampled signal $x_s(t)$ is given by multiplication of $x(t)$ & $\delta_s(t)$

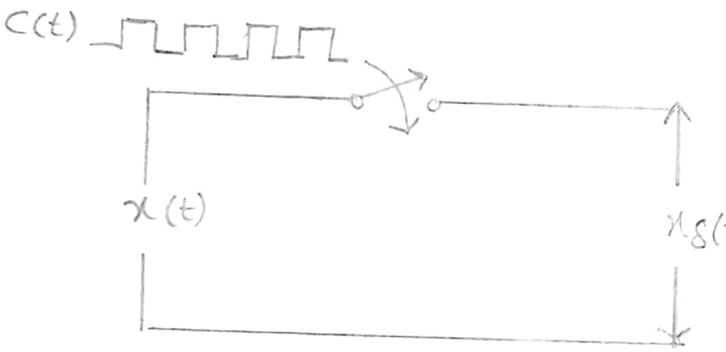
$$\therefore x_s(t) = x(t) \cdot \delta_s(t)$$

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

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

$$X_s(F) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s)$$

Natural Sampling:-



In natural sampling the pulses has finite width τ. It is also called as chopper sampling because the waveform of sampled signal appears to be chopped from the original signal waveform.

Let us consider an analog continuous time signal $x(t)$ to be sampled at the rate of f_s Hz & f_s is higher than nyquest rate such that sampling theorem is satisfied.

A sampled signal $x_\delta(t)$ is obtained by multiplication of sampling function & signal $x(t)$. sampling function $c(t)$ is a train of periodic pulses of width τ & frequency equal to f_s Hz.

$$x_\delta(t) = x(t) \quad \text{when } c(t) = A$$
$$x_\delta(t) = 0 \quad \text{when } c(t) = 0$$

$A \rightarrow$ amplitude of $c(t)$

$$x_\delta(t) = x(t) \cdot c(t)$$

$$c(t) = \sum_{n=-\infty}^{\infty} C_n \cdot e^{j2\pi nt/T_0} \quad ; \quad C_n = \frac{TA}{T_0} \operatorname{sinc}(f_n T)$$

T = pulse width $= \tau$

— For the periodic pulse train of $c(t)$ we have

$$T_0 = T_s = \frac{1}{f_s} = \text{period of } c(t)$$

$$C_n = \frac{\tau A}{T_s} \operatorname{sinc}(f_n \tau)$$

$$f_0 = f_s = \frac{1}{T_0} = \frac{1}{T_s} = \text{frequency of } c(t)$$

$$f_n = n \cdot f_s = \frac{n}{T_0} = n f_0$$

~~$C_n = \frac{\tau A}{T_s} \operatorname{sinc}(f_n \tau)$~~

~~substituting C_n value in~~

$$c(t) = \sum_{n=-\infty}^{\infty} C_n \, e^{j2\pi f_s n t} \longrightarrow (1)$$

$c(t)$ is a rectangular pulse train. C_n for this waveform is given by

$$C_n = \frac{TA}{T_0} \operatorname{sinc}(f_n T)$$

Substituting C_n value in eq (1)

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

$$x_\delta(t) = c(t) \cdot x(t) \rightarrow (2)$$

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

which represents naturally sampled signals.

Applying fourier transform to the equation (2)

$$X_\delta(f) = FT\{x_\delta(t)\}$$

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

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

From frequency shifting property of fourier transform

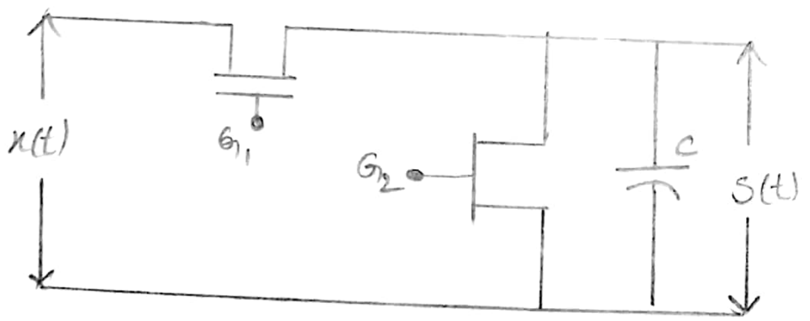
$$e^{j2\pi f_s n t} \cdot x(t) \longleftrightarrow X(f - nf_s)$$

$$X_\delta(f) = \frac{\tau A}{T_s} \sum_{n=-\infty}^{\infty} \text{sinc}(f_n\tau) \cdot X(f - nf_s)$$

Flat-top sampling:-

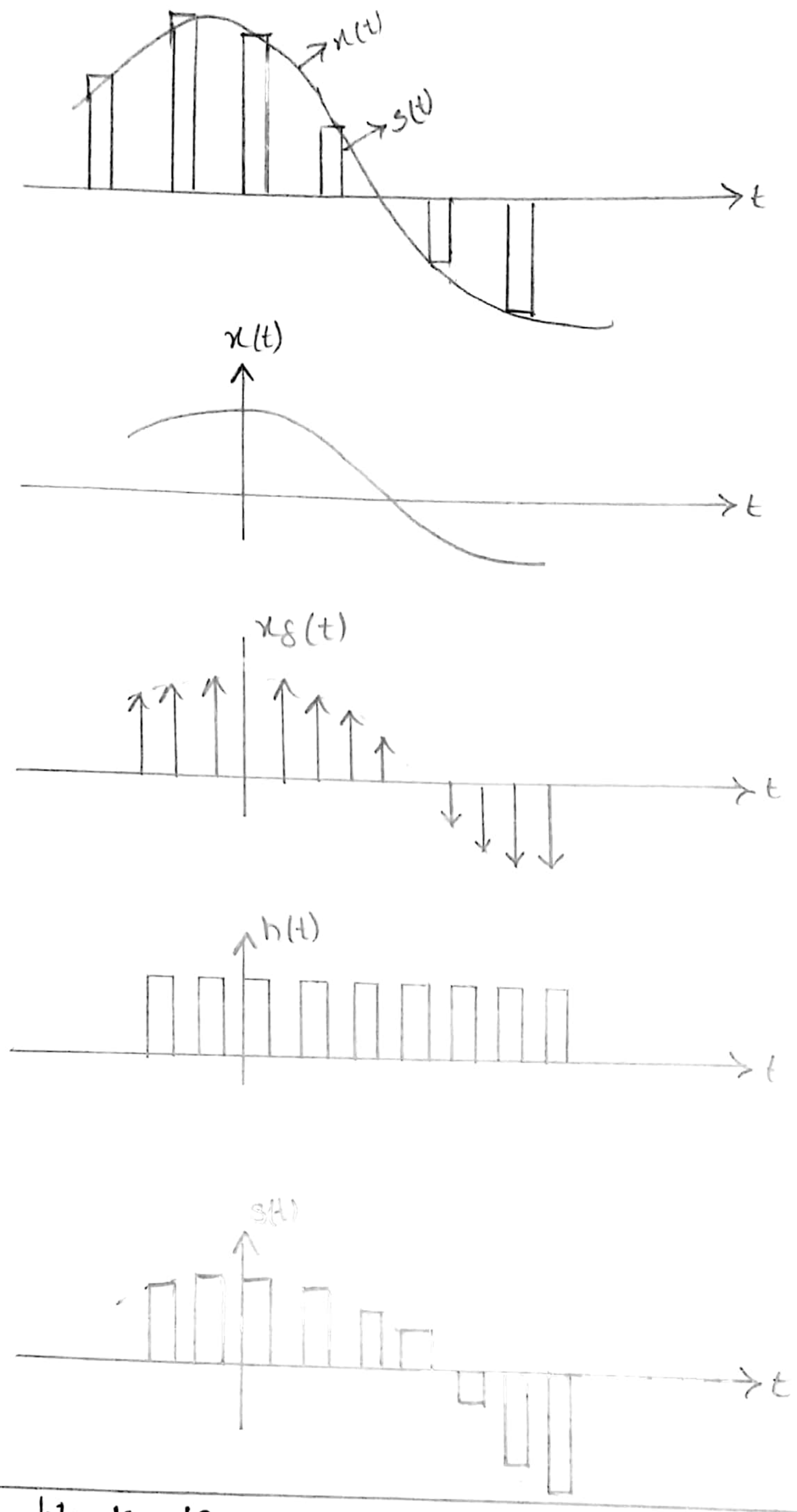
In this method the top of the samples remains constant & equal to instantaneous value of baseband signal $x(t)$ at the start of sampling. The duration of each sample τ and sampling rate is equal to f_s.

Generation of flat-top samples.



Spectrum of flat-top sampled signal $S(f)$ can be obtained as

$$S(f) = f_s \cdot \sum_{n=-\infty}^{\infty} X(f - nf_s) \cdot H(f)$$



Q) Draw the block diagram of PCM system and explain each block in detail?

- Pulse Code Modulation (PCM) system include the following basic elements.
    1. Transmitter or generator
    2. Transmission path
    3. Receiver.] X

Not required

Pulse Modulation System:-

In the actual sense, pulse modulation should not be considered as modulation. The following points are important for designing pulse modulation system.

(1) pulse modulated waves are which in DC and low frequency components.

(2) Care must be taken to prevent the overlapping of pulses during transmission of a pulse modulated wave.

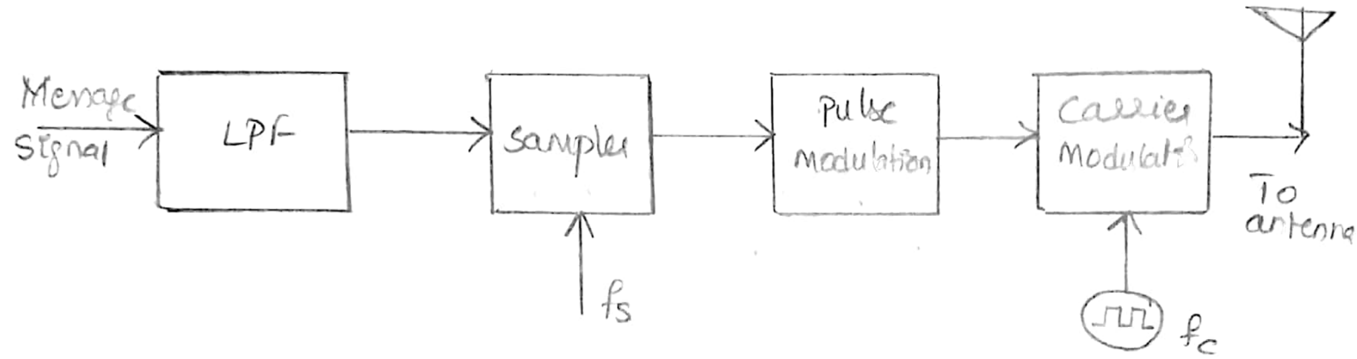


fig: Transmitter

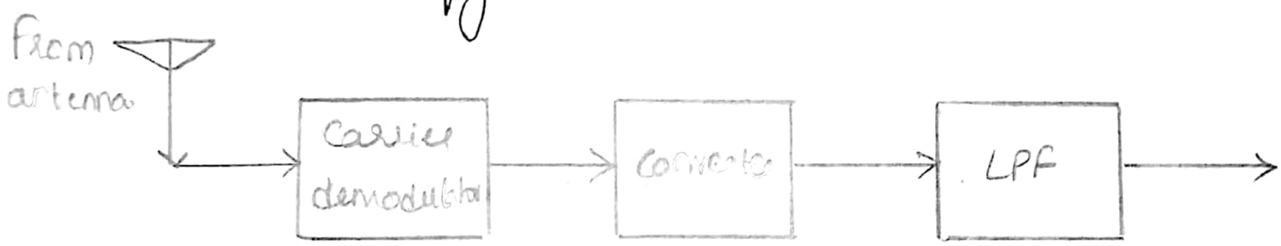


fig: Receiver.

There are two types of pulse modulation systems as under:

(i) pulse Amplitude Modulation (PAM)

(ii) Pulse Time Modulation (PTM)

Further there are two types of PTM. They are

(i) pulse width/Duration Modulation

(ii) Pulse Position Modulation.

Pulse Amplitude Modulation:-

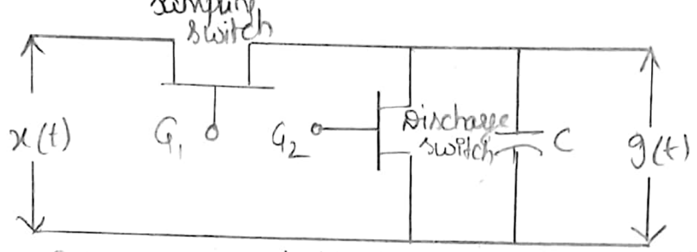
It is defined as the type of modulation in which the amplitudes of regularly spaced rectangular pulses vary according to instantaneous value of the modulating (or) message signal.

Out of the three sampling techniques (or) PAM methods, the flat-top PAM is most popular and is widely used. The reason for using flat-top PAM is that during the transmission, the noise interferes with the top of the transmitted pulses and this noise can be easily removed if the PAM pulse has flat-top. Where as in natural PAM & ideal PAM it is quite difficult to determine the shape of the top of the pulse and thus amplitude detection of the pulse is not exact.

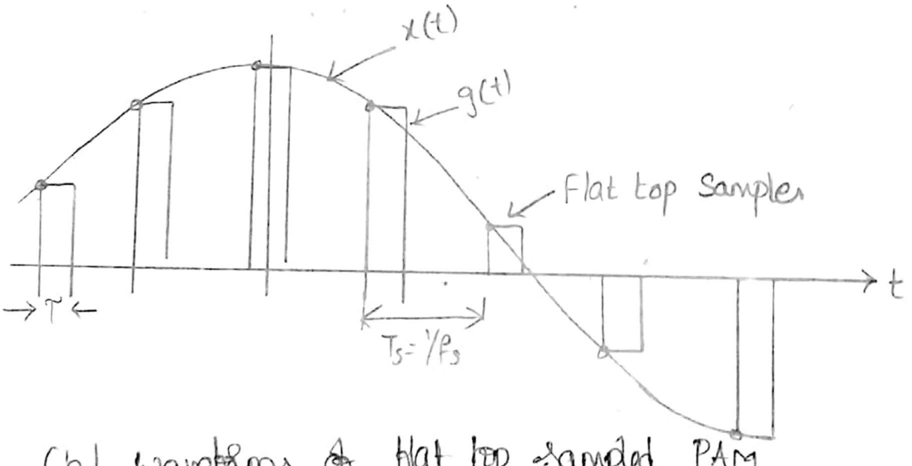
Working Principle:-

The flat-top PAM signal could be generated by using a sample and hold type electronic circuit. There is some high frequency loss in the recovered analog waveform due to filtering effect $H(f)$ caused by the flat top pulse shape. This can be compensated (Equalized) at the receiver by making the transfer function of the LPF to $1/H(f)$. This is a very common practice called "Equalization". The pulse width τ is called the "aperture" since τ/T_s determines the gain of the recovered analog signal.

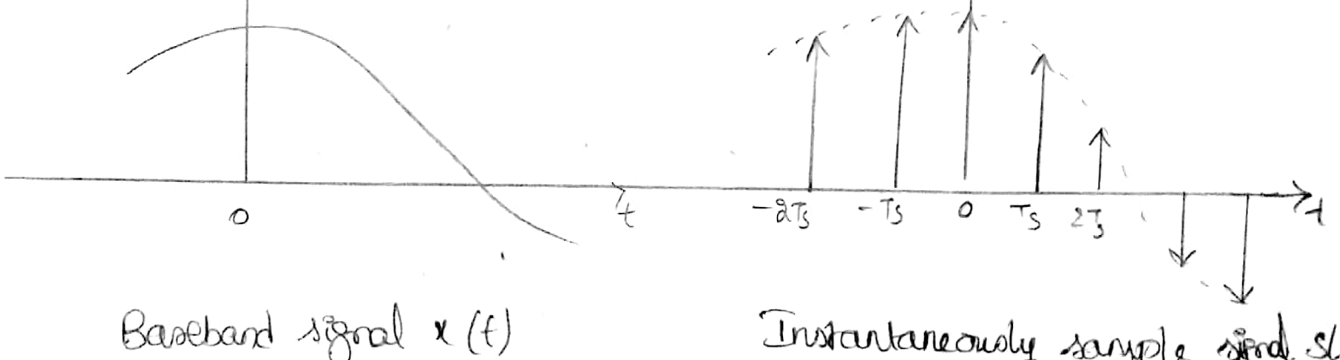
A sample and hold circuit shown in the figure is used to produce flat-top sampled PAM. The working principle of this circuit is quite easy. The sample & hold circuit (S&H circuit) consists of two field effect transistors (FET) switches and a capacitor. The sampling switch is closed for a short duration by a short pulse applied to the gate G_1 of the transistor. During this period, the capacitor C is quickly charged upto a voltage equal to the instantaneous sample value of the incoming signal x(t). Now the sampling switch is opened and the capacitor 'C' holds the charge. The discharge switch is then closed by a pulse applied to gate G_2 of the other transistor. Due to this, the capacitor C is discharged to zero volts. The discharged switch is then opened and thus capacitor has no voltage. Hence the output of the sample and hold circuit consists of a sequence of flat top sample as shown in figure below (b)



(a) sample & hold circuit generating flat-top sampled PAM

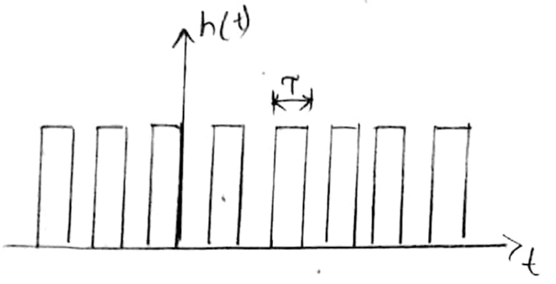


(b) waveforms of flat top sampled PAM

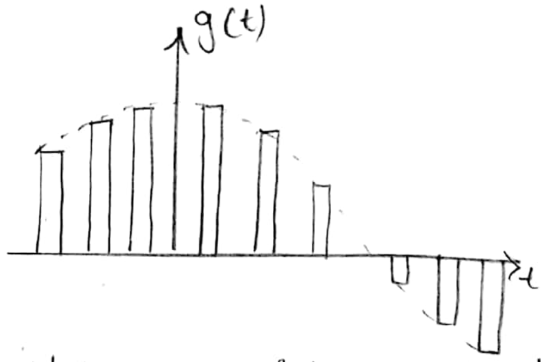


Baseband signal x(t)

Instantaneously sample signal s(t)



Constant pulse width function $h(t)$



Flat-top sampled PAM signal $g(t)$ obtained through convolution of $h(t)$ and $s(t)$

Mathematical Analysis:-

We know that the train of impulses may be represented mathematically as

$$\delta_{T_s}(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s) \longrightarrow (1)$$

The signal $s(t)$ is obtained by multiplication of baseband signal $x(t)$ & $\delta_{T_s}(t)$. Thus

$$s(t) = x(t) \cdot \delta_{T_s}(t) \longrightarrow (2)$$

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

$$= \sum_{n=-\infty}^{\infty} x(nT_s)\, \delta(t - nT_s) \longrightarrow (3)$$

Now, sampled signal $g(t)$ is given as equation

$$g(t) = s(t) \otimes h(t) \longrightarrow (4)$$

$$g(t) = \int_{-\infty}^{\infty} s(\tau)\, h(t-\tau)\, d\tau \longrightarrow (5)$$

$$g(t) = \int_{-\infty}^{\infty} \sum_{n=-\infty}^{\infty} x(nT_s)\, \delta(t - nT_s)\, h(t-\tau)\, d\tau$$

$$g(t) = \sum_{n=-\infty}^{\infty} x(nT_s) \int_{-\infty}^{\infty} \delta(\tau - nT_s)\, h(t-\tau)\, d\tau \longrightarrow (6)$$

According to shifting property of delta function, we know that

$$\int_{-\infty}^{\infty} f(t)\, \delta(t - t_0) = f(t_0) \longrightarrow (7)$$

using equations (6) & (7), we get

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

This equation represents value of $g(t)$ in terms of sampled value $x(nT_s)$ and function $h(t-nT_s)$ for flat top sampled signal. Now again from equation (4), by taking fourier transform on both sides of equation (4), we get

$$G(f) = S(f) \cdot H(f) \longrightarrow (8)$$

we know that $S(f)$ is given as

$$S(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s) \longrightarrow (9)$$

$$\therefore \quad G(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s) \cdot H(f) \longrightarrow (10)$$

Thus spectrum of flat-top PAM signal

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

Thus according to ~~spectrum~~ above equation, we can plot the spectrum $G(f)$ as shown in figure below

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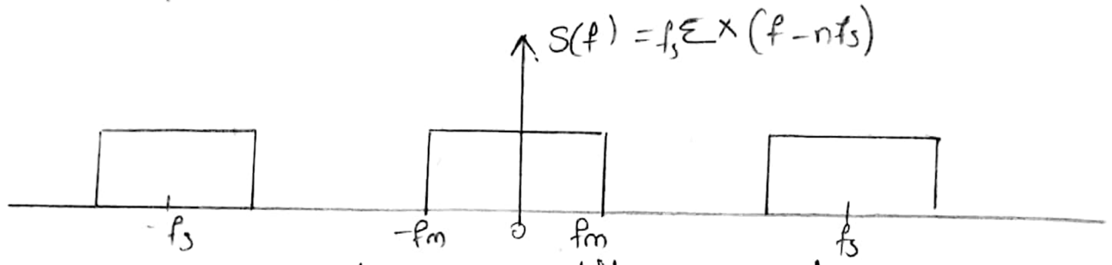
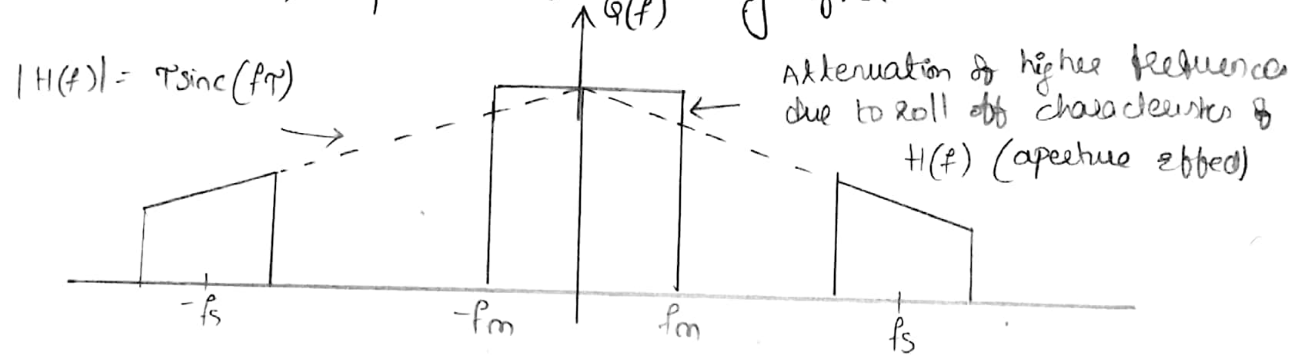
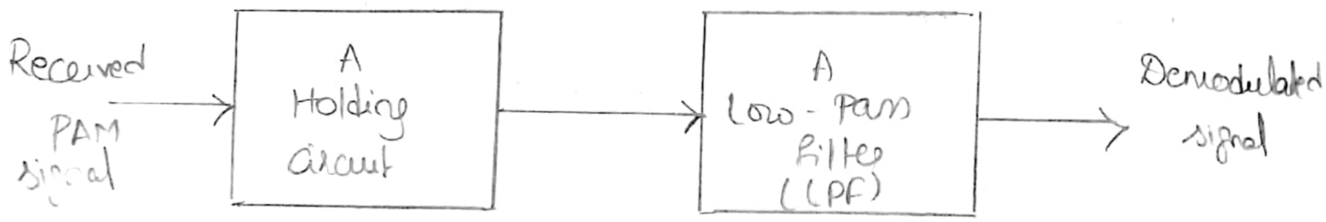


(a) spectrum of arbitrary signal

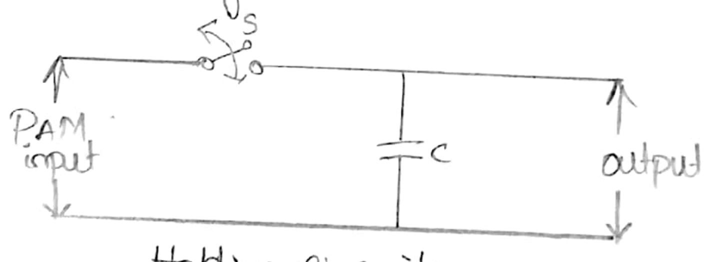


(b) spectrum of flat top signal

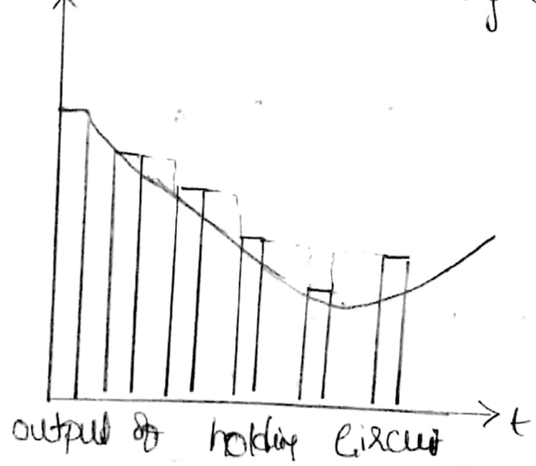
Demodulation of PAM Signals:-



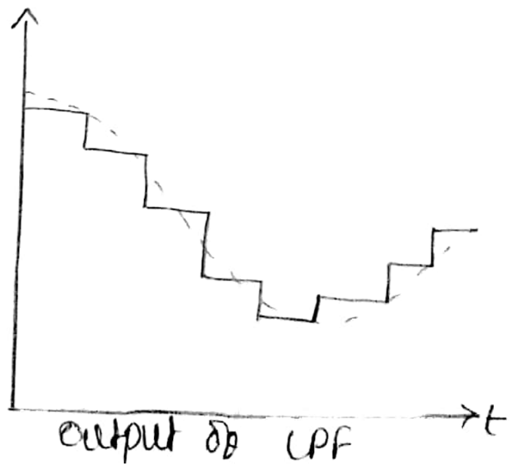
Block diagram of PAM demodulator.



Holding circuit.



output of holding circuit



output of LPF

For pulse amplitude modulated (PAM) signals, the demodulation is done using a holding circuit. In this method, the received PAM signal is allowed to pass through a Holding Circuit and a low pass filter (LPF) as shown in the above figure. In the figure holding circuit switch 'S' is closed after the arrival of the pulse and it is opened at the end of the pulse. In this way, the capacitor C is charged to the pulse amplitude value and it holds this value during the interval between the two pulses. Hence the sampled values are held as shown in figure of the output of LPF. It may be observed that some kind of distortion is introduced due to holding circuit. The circuit is known as zero-order holding circuit. This zero-order holding circuit considers only the previous sample to decide the value between the two pulses.

Transmission Bandwidth in PAM :-

In PAM signal the pulse duration 'τ' is considered to be very small in comparison to time period T_s between any two samples i.e $\tau << T_s$

Now, if the maximum frequency in the modulating signal $x(t)$ is f_m, then according to sampling theorem, the sampling frequency f_s must be equal to higher than the Nyquist rate i.e

$$f_s \geq 2 f_m$$

$$\frac{1}{T_s} \geq 2 f_m \text{ (or) } T_s \leq \frac{1}{2 f_m}$$

But according to equation $\tau << T_s$ $\therefore$ $\tau << T_s \leq \frac{1}{2f_m}$

Now, if the 'ON' and 'OFF' time of the pulse amplitude modulated (PAM) pulse is same as shown in figure below. then maximum frequency of the PAM pulse will be

$$f_{max} = \frac{1}{\tau + \tau} = \frac{1}{2\tau}$$

Therefore, the bandwidth required for the transmission of a PAM signal would be equal to the maximum frequency f_{max} given by the above equation. Thus, we have

Transmission bandwidth

$$BW \geq f_{max}$$

but $f_{max} = \frac{1}{2\tau}$

hence $BW \geq \frac{1}{2\tau}$

again, since $\tau << \frac{1}{2f_m}$

$$\boxed{\begin{aligned} \therefore \quad & BW \geq \frac{1}{2\tau} >> f_m \\ & BW >> f_m. \end{aligned}}$$

Pulse Time Modulation:-

In pulse Time modulation, amplitude of pulse is held constant, whereas position of pulse & width of pulse is made proportional to the amplitude of signal at the sampling instant. There are two types of pulse time modulation. They are (1) Pulse Width Modulation (PWM)

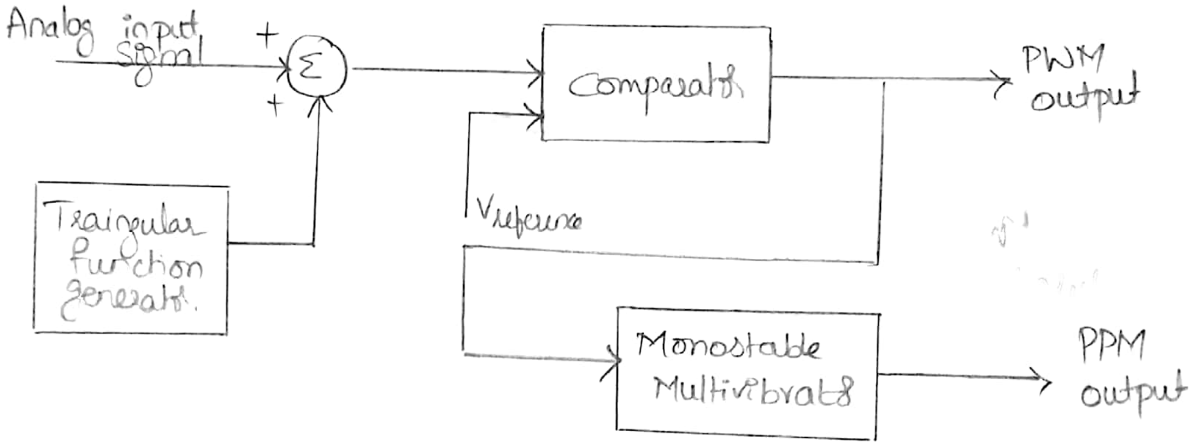
(2) Pulse Position Modulation (PPM)

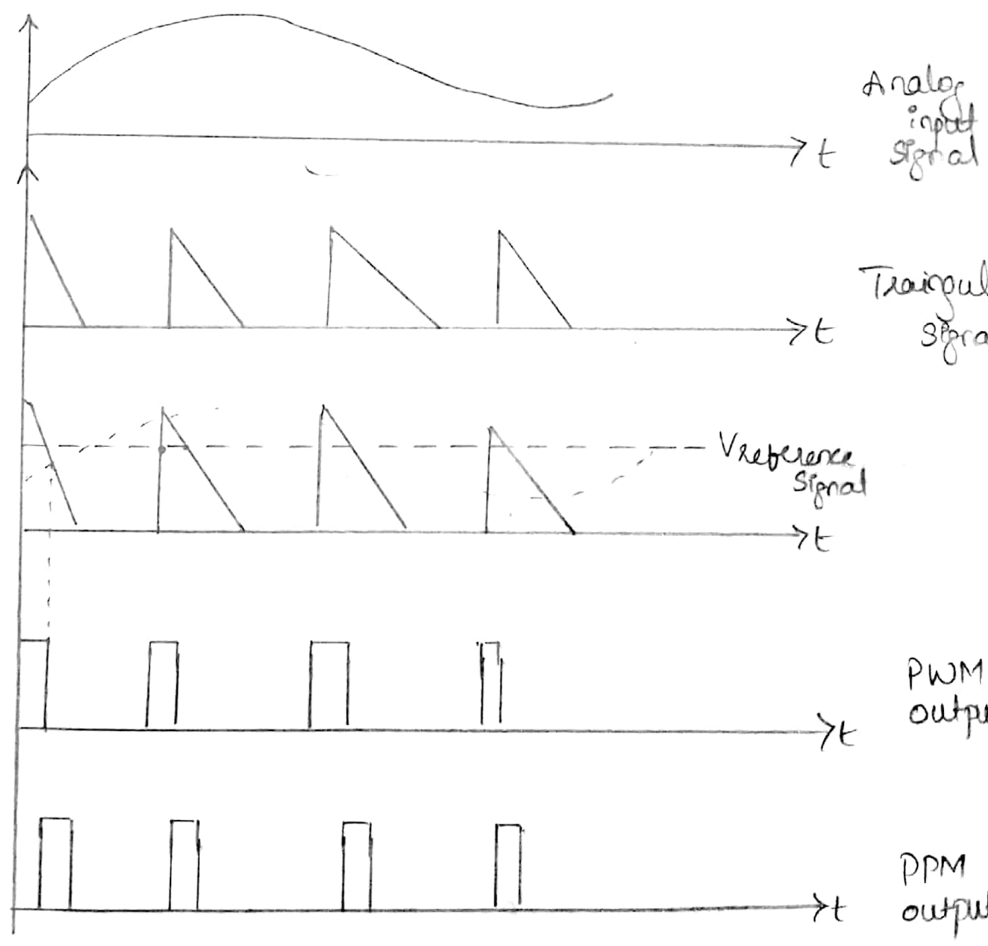
Because in both PWM and PPM, amplitude is held constant and does not carry any information, therefore amplitude limiters can be used. The amplitude limiters, similar to those used in FM, will clip off the portion of the signal corrupted by noise and hence provide a good degree of noise immunity.

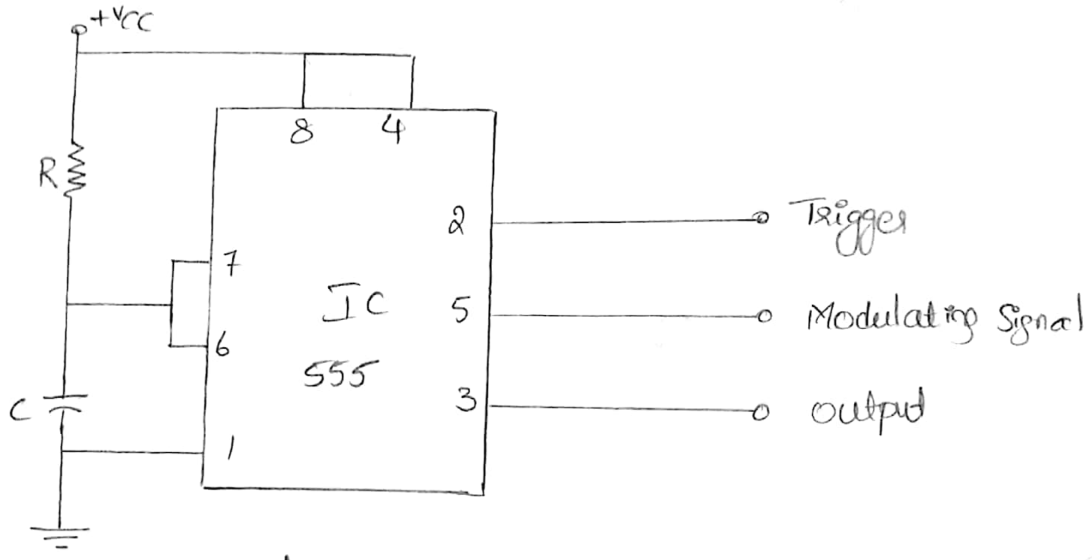
Pulse Width Modulation (PWM):-

In PWM the width of the carrier signal consisting of periodic train of pulses is varied linearly with amplitude of the message signal.

Generation of PWM:- (& PPM also)







Practical generation method.

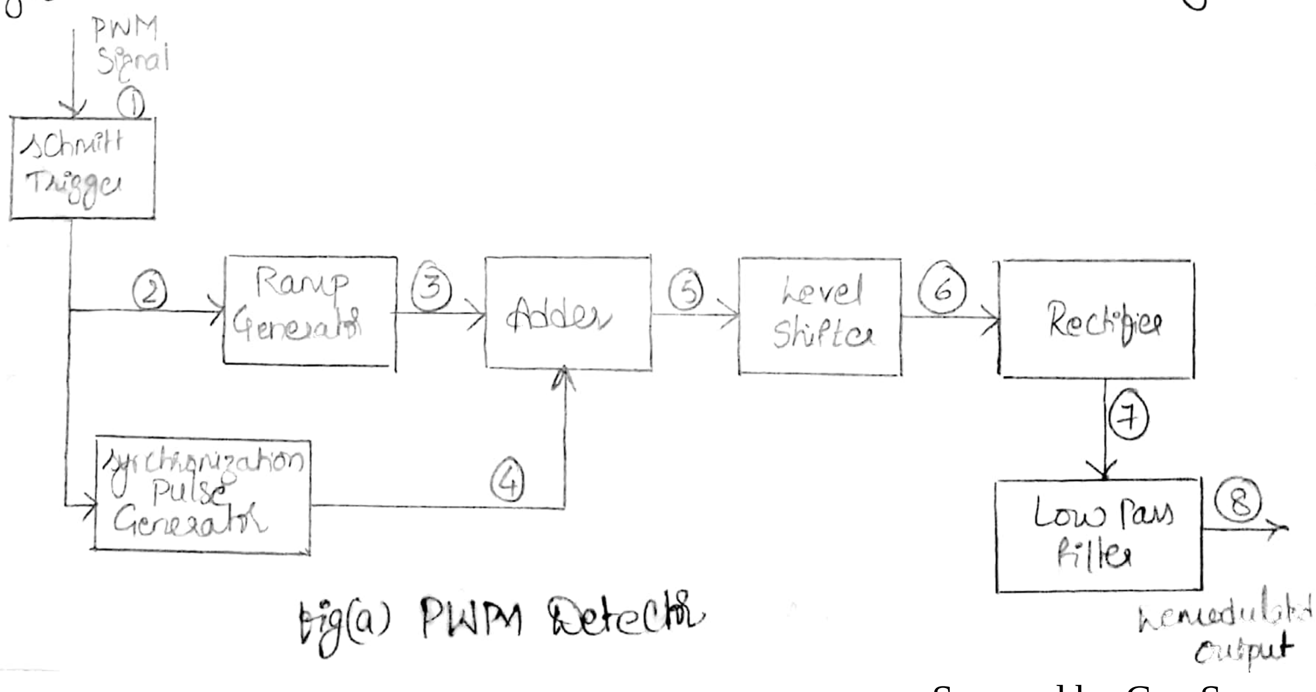
PWM signal can be generated by using a comparator, where modulating signal and triangular signal form the input of the Comparator. It is the simplest method for PWM generation. One input of the comparator is fed by the input message & modulating signal and the other input by a sawtooth signal which operates at carrier frequency considering both ±ve sides, the maximum of the input signal should be less than that of sawtooth signal. The comparator will compare two signals together to generate the PWM signal at its output as shown in the above diagrams. The rising edges of the PWM signal coincides with the falling edge of the triangular signal.

Generation of PWM using IC operation:-

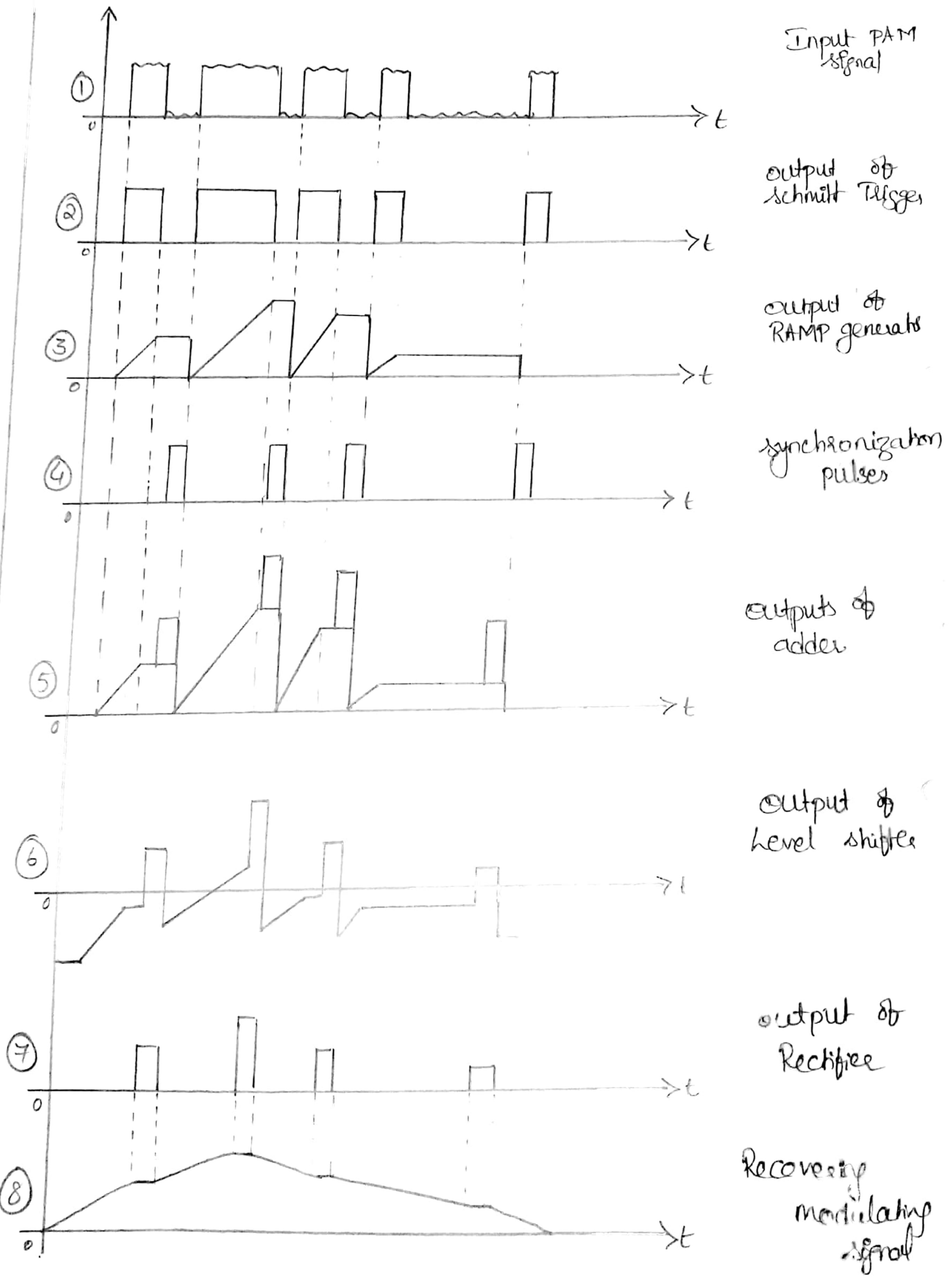
Generation of PWM using IC basically includes a monostable multivibrator with a modulating input signal applied at the control voltage input. Internally, the control voltage is adjusted to the 2/3 Vcc. Externally applied modulating signal changes the control voltage, and hence the threshold voltage level. As a result, the time period required to charge the capacitor up to threshold voltage level changes, giving pulse modulated signal at the output

Demodulation of PWM signal:-

The below block diagram represents the PWM detector. The received PWM signal is applied to the schmitt trigger circuit. This schmitt trigger circuit removes the noise in the PWM waveform. The regenerated PWM is then applied to the ramp generator and the synchronization pulse detector. The ramp generator produces ramps for the duration of pulses such that height of ramps are proportional to the widths of PWM pulses. The maximum ramp voltage is retained till the next pulse. on the other hand, synchronous pulse detector produces reference pulses with constant amplitude and pulse width. These pulses are delayed by specific amount of delay as shown in waveform figures. The delayed reference pulses and the output of ramp generator is added with the help of adder. The output of adder is given to level shifter. Here, negative offset shifts the waveform as shown in figure (b) Then the negative part of the waveform is clipped by rectifier. Finally, the output of rectifier is passed through LPF to recover the modulating signal



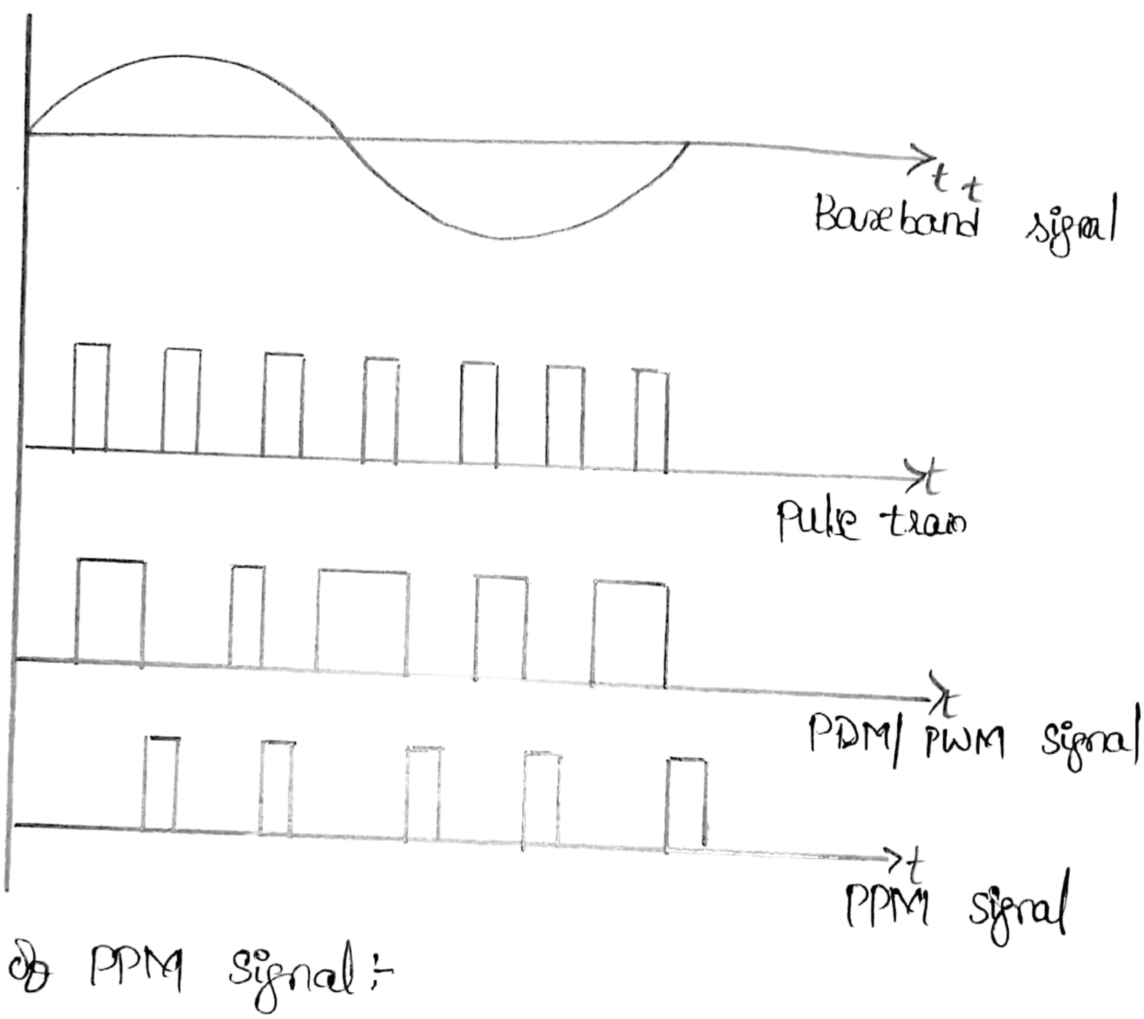
fig(a) PWM Detector



(b) Waveforms of PWM detection circuit

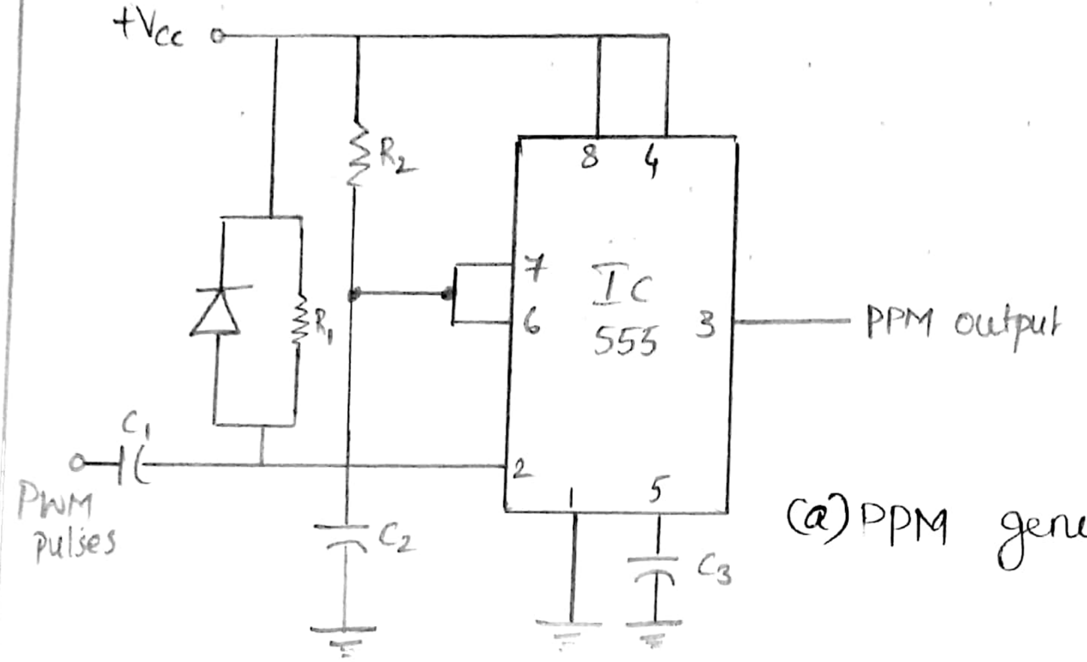
Pulse Position Modulation:-

In this system, the amplitude and width of the Pulses are kept constant, while the position of each pulse, with reference to the position of a reference pulse, is changed according to the instantaneous sampled value of the modulating signal. Thus, the transmitter has to send synchronizing pulses to keep the transmitter and receiver in synchronism. PPM is obtained from PWM Each trailing edge of the PWM pulse is a starting Point of the pulse in the PPM.

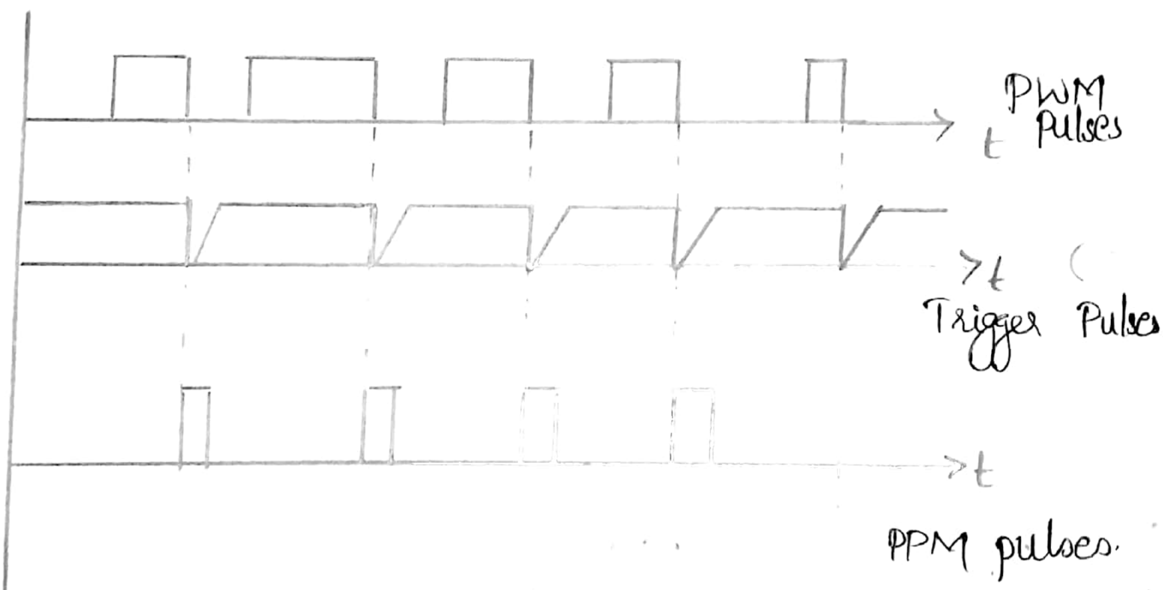


Generation of PPM Signal:-

The PPM generator consists of differentiator and a monostable multivibrator. The input to the differentiator is a PWM waveform. The differentiator generates positive and negative spikes corresponding to leading & trailing edges of the PWM waveform. Diode D_1 is used to bypass the positive spikes. The negative spikes are used to the trigger monostable multivibrator. The monostable multivibrator then generates the pulses of same width and amplitude with reference to trigger to give pulse position modulated waveform

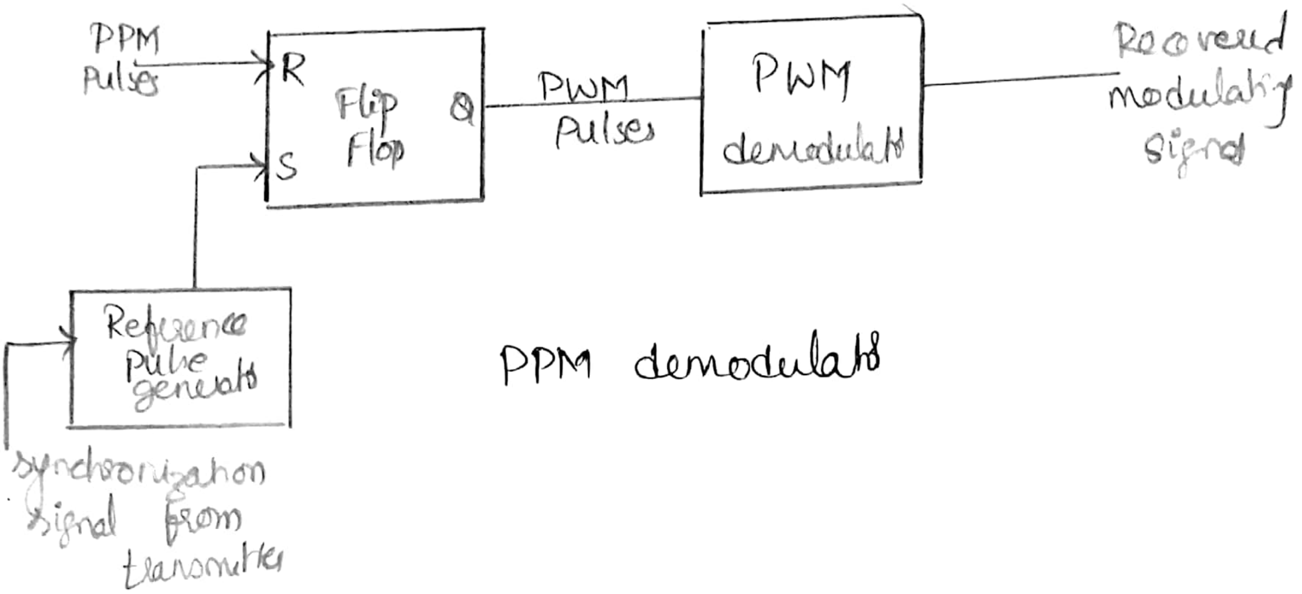


(a) PPM generator



(b) waveforms of PPM generator

Demodulation of PPM :-



PPM demodulator

Flip Flop circuit is set & turned 'ON' when the reference pulse arrives. This reference pulse is generated by reference pulse generator of the receiver with the synchronization signal from the transmitter. The flip-flop circuit reset & turned OFF (giving low output) at the leading edge of the position modulated pulse. This repeats and we get PWM pulses at the output of the flip flop. The PWM pulses are then demodulated by PWM demodulator to get original modulating signal.

Problem: For a PAM transmission of voice signal having maximum frequency equal to $f_m = 3KHz$, calculate the transmission bandwidth. It is given that the sampling frequency $f_s = 8KHz$ and the pulse duration $\tau = 0.1 T_s$.

$$T_s = \frac{1}{f_s} = \frac{1}{8 \times 10^3} = 125 \mu sec$$

given $\tau = 0.1 T_s$

$$= 0.1 \times 125 \mu sec$$

$$= 12.5 \mu sec$$

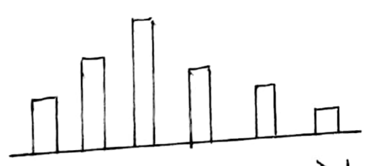
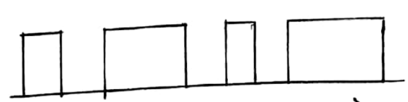
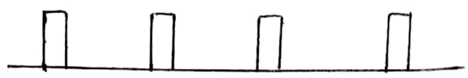
The transmission bandwidth for PAM signal is expressed as $BW \geq \frac{1}{2\tau}$

$$\therefore BW \geq \frac{1}{2 \times (12.5) \times 10^{-6}}$$

$$BW \geq \frac{1 \times 10^6}{25}$$

$$\boxed{BW \geq 40KHz}$$

Comparison of Pulse Analog Modulation Techni

PAM	PWM	PPM
1. Amplitude of the pulse is proportional to amplitude of the modulating signal	1. Width of the pulse is proportional to amplitude of the modulating signal	1. The relative position of the pulse is proportional to the amplitude of modulat signal.
2. The bandwidth of the transmission channel depends on width of the pulse	2. Bandwidth of transmission channel depends on rise time of the pulse	2. Bandwidth of the transmission channel depends on rising time of the pulse
3. The instantaneous power of the transmitter varies	3. The instantaneous power of the transmitter varies	3. The instantaneous power of the transmitter remains constant
4. Noise interference is high	4. Noise interference is minimum	4. Noise interference is minimum
5. Similar to AM	5. Similar to FM	5. Similar to PM
6. System is complex	6. Simple to implement	6. Simple to implement
7. Waveform representation  $\rightarrow t$	7. Waveform representation  $\rightarrow t$	7. Waveform representation  $\rightarrow t$

Advantages, Disadvantages & applications of PAM, PWM & PPM

PAM System:-

Merits:

→ Bandwidth depends on the width of the pulse

→ BW = $n f_m$ where n represents no of signals.

→ Instantaneous power of T^x varies

Demerits:

→ complex

→ high noise interference

→ average power is very less ie $A^2/2$

Application:

→ Flat-top PAM signal is used for the generation of different signals as it contains very less noise (nill).

PWM System:

Merits:

→ Bandwidth = BHz

→ noise Interference is minimum

→ simple to implement

→ bandwidth depends on rise of the pulse

Demerits:

→ inefficient in the sense of transmission power

→ frequency cannot be used for signal generation

Application:

→ It can be used for generation of PPM signal

→ used for generation of PAM signal

PPM system:

Merit

→ more efficient than PWM system in the sense of transmitted power utilization

→ instantaneous power remains constant

→ noise interference is minimum

→ simple to implement

Demerits

→ not used for various signal generations

→ PPM suffers from threshold effect

Application

→ PPM is used to generate PWM signal.

→ used in PAM signal generation

→ PPM signals are used in telemetry & instrumentation.

Drawbacks of PAM signal:-

(1) BW required for the transmission of a PAM signal is very large in comparison to the maximum frequency present in the modulating signal

(2) Interference of noise is maximum and it cannot be removed easily.

(3) As amplitude of the PAM signal is varies, the peak power required by the transmitter with modulating signal also varies.

=X=