UNIT II

SIMPLE DESCRIPTION ON MODULATION

Analog Modulation

A message carrying a signal has to get transmitted over a distance and to establish a reliable communication, it needs to take the help of a high frequency sign. A high frequency signal can travel up to a longer distance, without getting affected by external disturbances. We take the help of such high frequency signal which is called as a **carrier signal** to transmit our message signal. Such a process is simply called as Modulation which should not affect the original characteristics of the message signal.

Signals in the Modulation Process:

Following are the three types of signals in the modulation process.

Message or Modulating Signal

The signal which contains a message to be transmitted, is called as a **message signal**. It is a baseband signal (the original signal which is generated by the source) which has to undergo the process of modulation, to get transmitted. Hence, it is also called as the **modulating signal**.

Carrier Signal

The high frequency signal, which has a certain amplitude, frequency and phase but contains no information is called as a **carrier signal**. It is an empty signal and is used to carry the signal to the receiver after modulation.

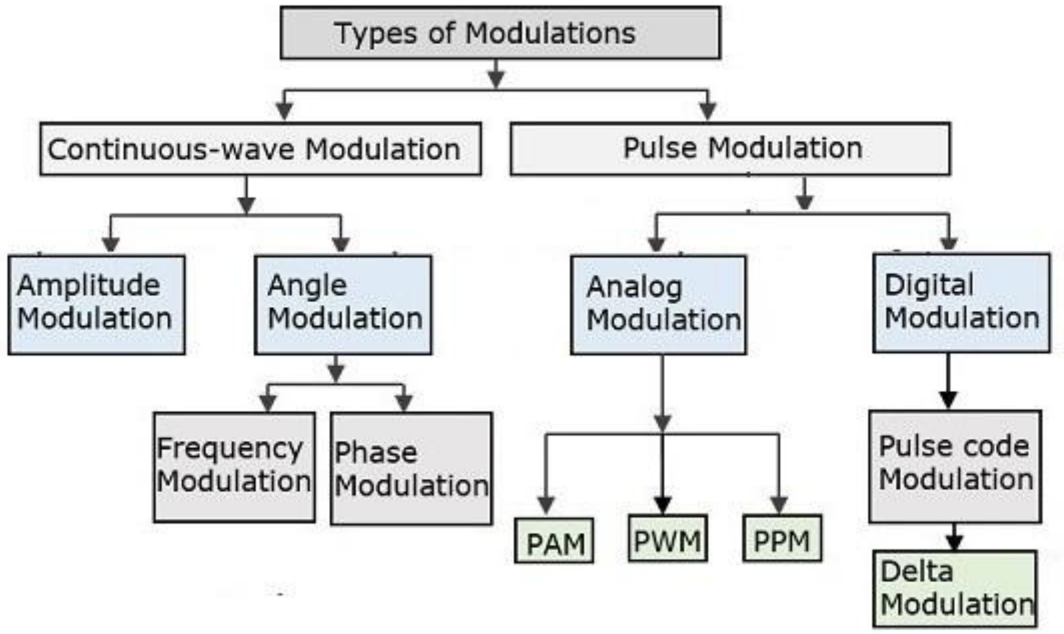
Modulated Signal

The resultant signal after the process of modulation is called as a **modulated signal**. This signal is a combination of modulating signal and carrier signal.

There are many types of modulations. Depending upon the modulation techniques used, they are classified as shown in the following figure.

Continuous-wave Modulation

In continuous-wave modulation, a high frequency sine wave is used as a carrier wave. This is further divided into amplitude and angle modulation



- If the amplitude of the high frequency carrier wave is varied in accordance with the instantaneous amplitude of the modulating signal, then such a technique is called as **Amplitude Modulation**.
- If the angle of the carrier wave is varied, in accordance with the instantaneous value of the modulating signal, then such a technique is called as **Angle Modulation**. Angle modulation is further divided into frequency modulation and phase modulation.
    - If the frequency of the carrier wave is varied, in accordance with the instantaneous value of the modulating signal, then such a technique is called as **Frequency Modulation**.
    - If the phase of the high frequency carrier wave is varied in accordance with the instantaneous value of the modulating signal, then such a technique is called as **Phase Modulation**.

Pulse Modulation

In Pulse modulation, a periodic sequence of rectangular pulses, is used as a carrier wave. This is further divided into analog and digital modulation.

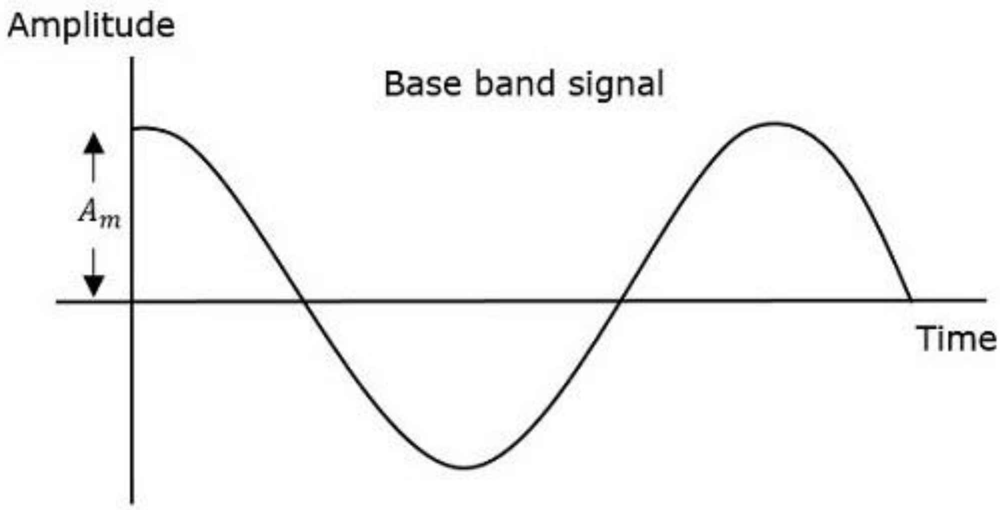
In analog modulation technique, if the amplitude or duration or position of a pulse is varied in accordance with the instantaneous values of the baseband modulating signal, then such a technique is called as Pulse Amplitude Modulation (PAM) or Pulse Duration/Width Modulation (PDM/PWM), or Pulse Position Modulation (PPM).

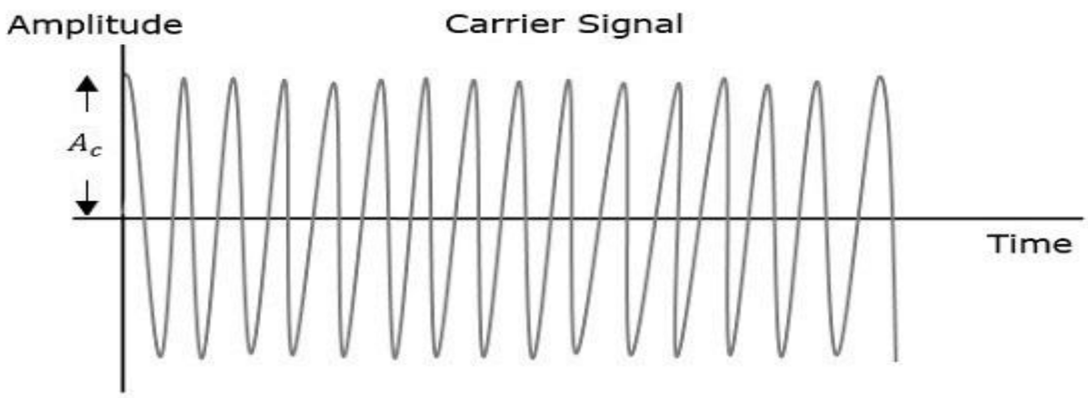
In digital modulation, the modulation technique used is Pulse Code Modulation (PCM) where the analog signal is converted into digital form of 1s and 0s. As the resultant is a coded pulse train, this is called as PCM. This is further developed as Delta Modulation (DM).

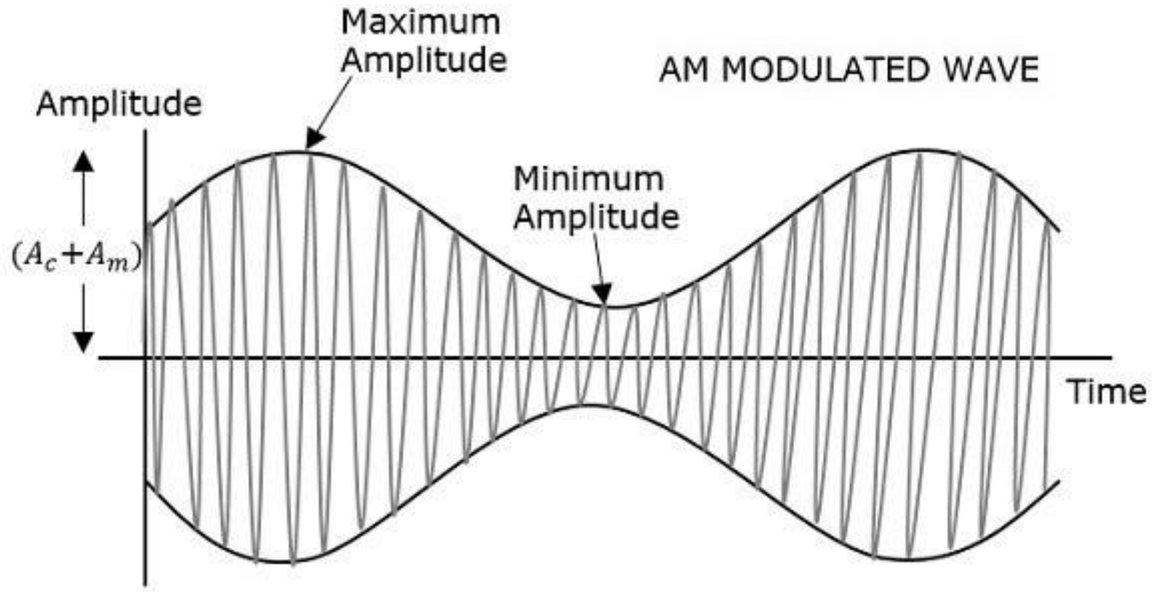
AMPLITUDE MODULATION

A continuous-wave goes on continuously without any intervals and it is the baseband message signal, which contains the information. This wave has to be modulated.

According to the standard definition, "The amplitude of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal." Which means, the amplitude of the carrier signal containing no information varies as per the amplitude of the signal containing information, at each instant. This is shown in the figures below.







The first figure shows the modulating wave, which is the message signal. The next one is the carrier wave, which is a high frequency signal and contains no information. While, the last one is the resultant modulated wave.

It can be observed that the positive and negative peaks of the carrier wave, are interconnected with an imaginary line. This line helps recreating the exact shape of the modulating signal. This imaginary line on the carrier wave is called as **Envelope**. It is the same as that of the message signal.

Time-domain Representation of the Waves

Let the modulating signal be,

$$m(t)=A_m \cos(2\pi f_m t)$$

and the carrier signal be,

$$c(t)=A_c \cos(2\pi f_c t)$$

Where, A_m and A_c are the amplitudes of the modulating signal and the carrier signal respectively and f_m and f_c are the frequency of the modulating signal and the carrier signal respectively.

Then, the equation of Amplitude Modulated wave will be

$$s(t)=[A_c+A_m\cos(2\pi f_m t)] \cos(2\pi f_c t) \text{------------------}(1)$$

Re arranging the above equation,

$$s(t) = A_c\,(1+(A_m / A_c)\cos(2\pi f_m t)]\cos(2\pi f_c t)$$

$$= A_c\,(1+\mu\cos(2\pi f_m t))\cos(2\pi f_c t) \text{---------------}(2)$$

Where μ is = A_m / A_c = modulation index which indicates the level of modulation that a carrier wave undergoes.----------------------(3)

If A_{max} and A_{min} are the maximum and minimum amplitudes of the modulated wave then the modulation index μ is given by

$$\mu = \frac{A_{max} - A_{min}}{A_{max} + A_{min}} \text{-----------------------}(4)$$

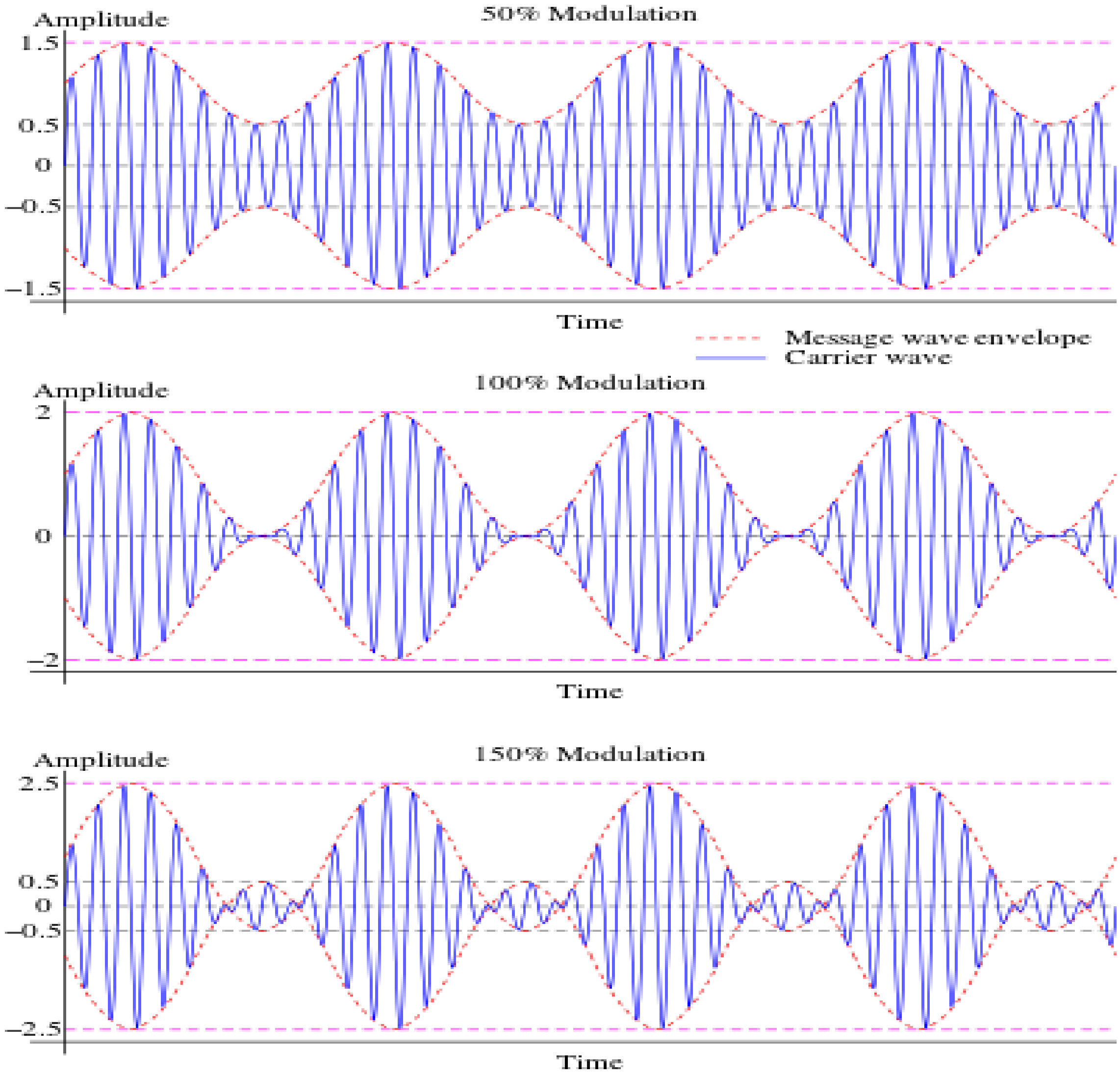
Equation 3 and Equation 4 are the two formulas for Modulation index. The modulation index or modulation depth is often denoted in percentage called as Percentage of Modulation. We will get the **percentage of modulation**, just by multiplying the modulation index value with 100.

For a perfect modulation, the value of modulation index should be 1, which implies the percentage of modulation should be 100%.

For instance, if this value is less than 1, i.e., the modulation index is 0.5, then the modulated output would look like the following figure . It is called as **under-modulation**. Such a wave is called as an **under-modulated wave**

If the value of the modulation index is equal to 1, then the wave will be an **critical-modulated wave**. It would look like the following figure .

If the value of the modulation index is greater than 1, i.e., 1.5 or so, then the wave will be an **over-modulated wave**. It would look like the following figure .



Bandwidth of AM Wave

The sine wave is the fundamental waveform and every other kind of waveform (triangular, rectangular etc) can be written as a combination of the fundamental sine wave. We get digital pulses when we superimpose sine waves of different harmonics.Bandwidth is a range of frequencies within a continuous set of frequencies. It is measured in Hertz.

Bandwidth (BW) is the difference between the highest and lowest frequencies of the signal. Mathematically, we can write it as

$$BW=f_{max} - f_{min}$$

Considering the equation (2)

$$s(t)= A_c (1+\mu \cos(2\pi f_m t)) \cos(2\pi f_c t)$$

$$= A_c \cos(2\pi f_c t) + A_c \mu \cos(2\pi f_m t) \cos(2\pi f_c t)$$

$$= A_c \cos(2\pi f_c t) + \frac{A_c \mu}{2} \cos[(2\pi(f_c + f_m)t)] + \frac{A_c \mu}{2} \cos[(2\pi(f_c - f_m)t)]$$

Hence, the amplitude modulated wave has three frequencies. Those are carrier frequency f_c , upper sideband frequency $f_c + f_m$ and lower sideband frequency $f_c - f_m$.

Here $f_{max} = f_c + f_m$ and $f_{min} = f_c - f_m$.

$BW = f_{max} - f_{min} = (f_c + f_m) - (f_c - f_m) = 2 f_m$.

Thus, it can be said that the bandwidth required for amplitude modulated wave is twice the frequency of the modulating signal.

Power Calculations of AM Wave

Consider the following equation of amplitude modulated wave,

$$= A_c \cos(2\pi f_c t) + \frac{A_c \mu}{2} \cos[(2\pi(f_c + f_m)t)] + \frac{A_c \mu}{2} \cos[(2\pi(f_c - f_m)t)]$$

From the above equation we can say that the first term represents the unmodulated carrier and two additional terms represent two side bands.The frequency of lower sideband (LSB) is $f_c - f_m$ and the frequency of the upper sideband is $f_c + f_m$.

Power of AM wave is equal to the sum of powers of carrier, upper side band, and lower side band frequency components.

$$P_t = P_c + P_{USB} + P_{LSB}$$

We know that the standard formula for power of cos signal is

$$P = \frac{V_{rms}^2}{R}$$ where V_{rms} is the rms value of cos signal

$$= \frac{\left(V_m/\sqrt{2}\right)^2}{R}$$ where V_m is the peak value of the cos signal

The carrier power is $P_C = \frac{\left(A_C/\sqrt{2}\right)^2}{R} = \frac{A_C{}^2}{2R}$

The upper side band power= $P_{USB} = \frac{\left(\mu A_C / 2\sqrt{2}\right)^2}{R} = \frac{\mu^2}{8R} {A_C}^2$

The lower side band power=$P_{LSB} = \frac{\left(\mu A_C / 2\sqrt{2}\right)^2}{R} = \frac{\mu^2}{8R} {A_C}^2$

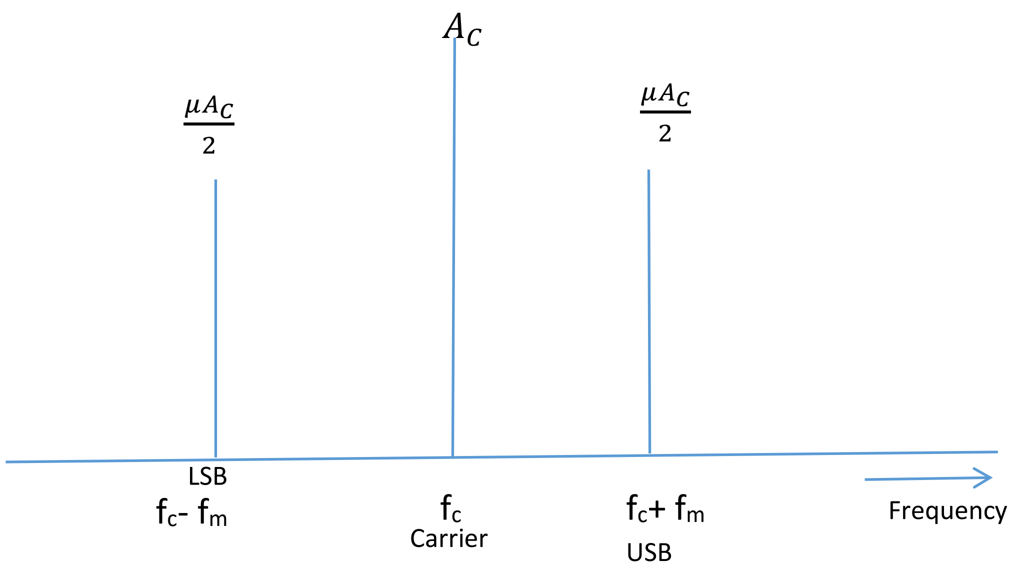
The total power of AM wave= $P_{Total} = P_c + P_{USB} + P_{LSB}$

$$= \frac{{A_C}^2}{2R} + \frac{\mu^2}{8R} {A_C}^2 + \frac{\mu^2}{8R} {A_C}^2$$

$$= \frac{{A_C}^2}{2R} \left(1 + \frac{\mu^2}{2}\right)$$

$$= P_C\left(1 + \frac{\mu^2}{2}\right)$$

Frequency domain representation of AM Signal:



The total AM current and unmodulated carrier current are related by the equation,

$$I_{Total} = I_C \sqrt{1 + \frac{\mu^2}{2}}$$

If there are two or more modulating signals then the total modulation index is given by

$$\mu_t = \sqrt{\mu_1^2 + \mu_2^2 + \dots\dots}$$

Ex: The output voltage of an AM transmitter is given by $400(1+0.4\cos 6280t)\cos 3.14*10^7t$. This voltage is fed to a load of 600Ωresistance. Determine carrier frequency, modulating frequency and total power output?

Sol: Given the output wave of transmitter is

$$s(t)= 400(1+0.4\cos 6280t) \cos 3.14*10^7t.$$

Comparing the above equation with equation(2)

$$s(t) = A_c (1+\mu \cos(2\pi f_m t)) \cos(2\pi f_c t)$$

A_{C}=400V; μ=0.4; R=600Ω

$2\Pi f_m=6280$; $2\Pi fc=3.14x10^7$;

$$f_m=\frac{6280}{2\Pi}=1000\text{Hz and } f_c=\frac{3.14x10^7}{2\Pi}=5\text{MHz}$$

The total power of AM wave=$P_{Total}= P_C(1+\frac{\mu^2}{2})$

$$P_C=\frac{{A_C}^2}{2R}=\frac{(400)^2}{2x600}= 133.33\text{W}$$

$$P_{Total}= 133.33\left(1+\frac{0.4^2}{2}\right)=144\text{W}$$

Ex: An amplitude modulated signal is given by

$s(t)=10\cos(2\Pi x10^6t)+5\cos(2\Pi x10^6t)\cos(2\Pi x10^3t)+2\cos(2\Pi x10^6t)\cos(4\Pi x10^3t)$

Volts. Find the total modulation index, sideband power and total modulated power.

Sol: Given $s(t)=10\cos(2\Pi x10^6t)+5\cos(2\Pi x10^6t)\cos(2\Pi x10^3t)+2\cos(2\Pi x10^6t)\cos(4\Pi x10^3t)$

$=10\cos(2\Pi x10^6t)[1+0.5\cos\cos(2\Pi x10^3t)+0.2\cos(4\Pi x10^3t)]$

$=10[1+0.5\cos(2\Pi x10^3t)+0.2\cos(4\Pi x10^3t)]\cos(2\Pi x10^6t)$

Comparing with the standard AM wave,

A_C= 10V; μ_1 =0.5; μ_2=0.2; $f_c= 10^6$=1MHz;

$f_{m1}= 10^3$Hz; $f_{m2}= 2x10^3$=2KHz; Let R=100Ω

The total modulation index is given by

$\mu_t = \sqrt{\mu_1^2+\mu_2^2+\ldots\ldots}$ $= \sqrt{0.5^2 + 0.2^2} = 0.5385$

The sideband power= $\frac{\mu_t^2}{4R} A_C^2 = \frac{0.5385^2}{4x100}$x $10^2 = 7.25$W

The total power of AM wave= $P_C(1+\frac{\mu^2}{2}) = \frac{A_C^2}{2R}(1+\frac{\mu^2}{2}) = \frac{10^2}{2x100}(1+\frac{0.5385^2}{2})$

=7.25W

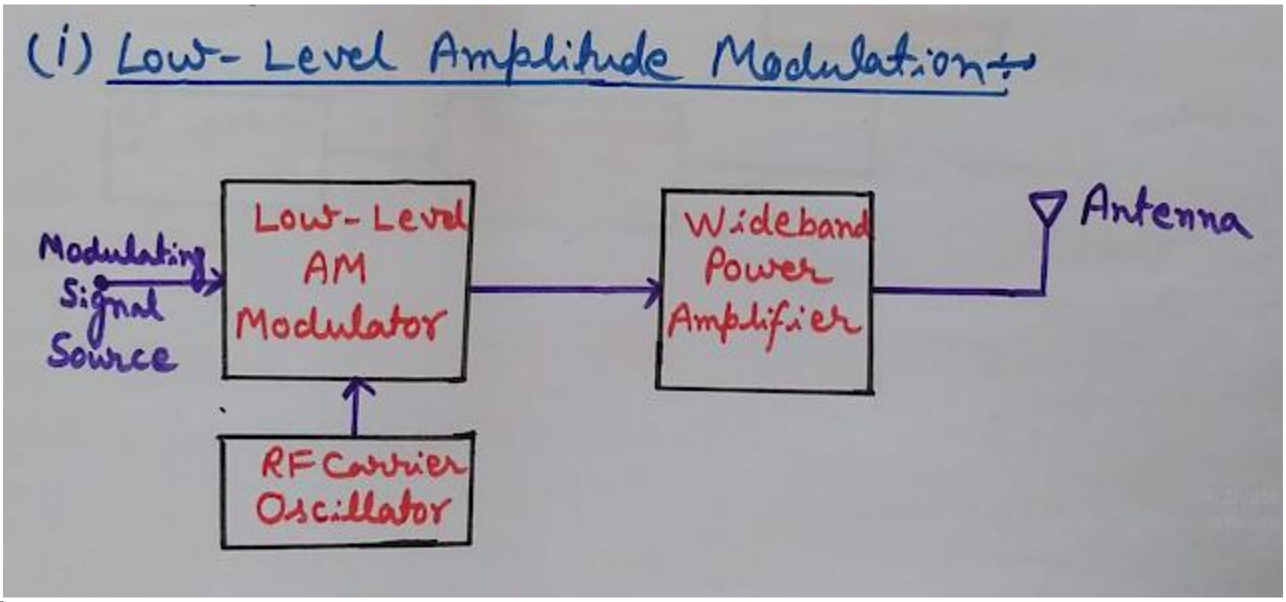
Generation of AM wave

The device which is used to generate an amplitude wave(AM) wave is known as Amplitude Modulator.

The methods of AM generation may be broadly classified as follows:

1) Low level AM Modulation
2) High level AM Modulation

Low level AM Modulation

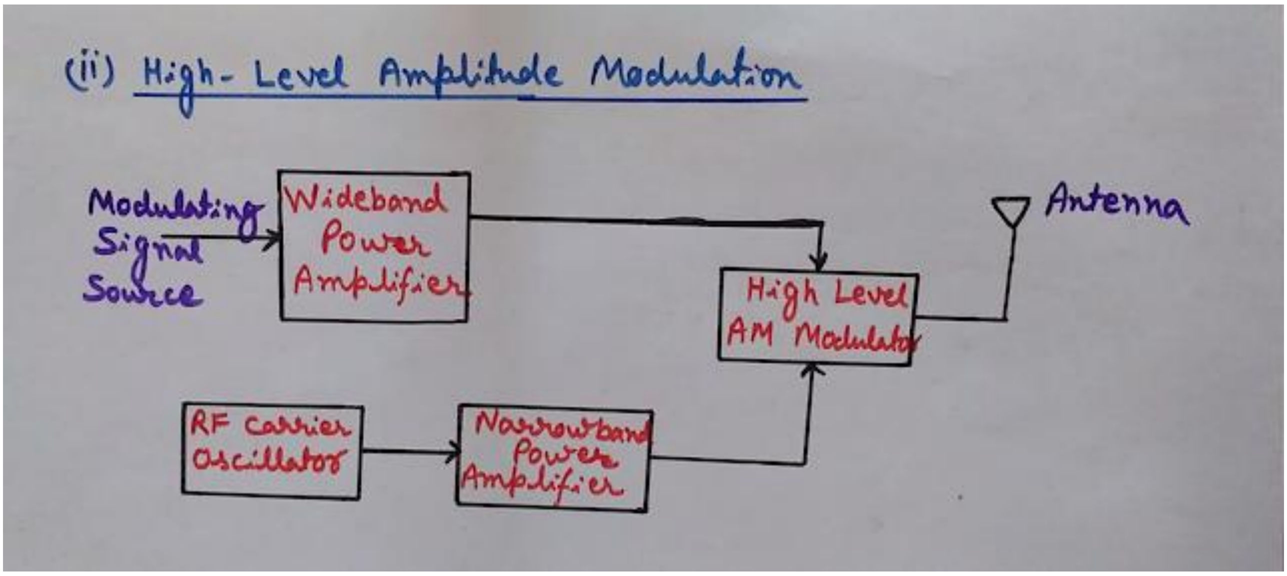


Here the modulation is done at low power level.At low power levels, a very small power is associated with the carrier and modulating signal. Because of this, the output power of modulation is low. Therefore the power amplifiers are required to boost the amplitude of the modulated signals upto the desired output level.From the figure it is clear that the modulation is done at low power levels. After this, the modulated signal is done is applied to a wideband power amplifier.

Square law diode modulation and switching modulation are examples of low level modulation.

High level AM Modulation

The block diagram of high level AM Modulation system is shown below



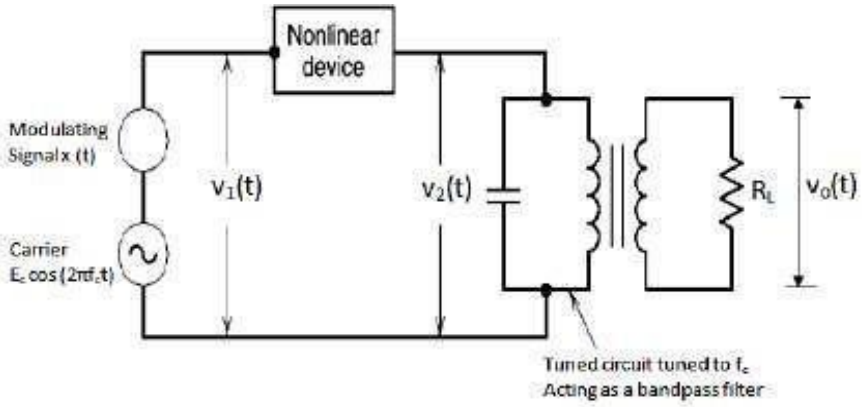
Here the modulation is done at high power levels.To produce amplitude modulation at these power levels, the baseband and the carrier signal must be a high power levels.Here the modulating and the carrier signals are first power amplified and then applied to high level AM modulator.For modulating signal, the wideband power amplifier is required just to preserve all the frequency components present in the modulating signal.For carrier signal, the narrow band power amplifier is required because it is a fixed frequency signal.

Square law modulator:

Let the modulating and carrier signals be denoted as m(t) and $A_C \cos(2\pi fct)$ respectively. These two signals are applied as inputs to the summer (adder) block. This summer block produces an output, which is the addition of the modulating and the carrier signal.

Mathematically, we can write it as

$$V_1(t)=m(t)+A_C \cos(2\pi f_C t)$$



Square law Modulator

This signal $V_1(t)$ is applied as an input to a nonlinear device like diode. The characteristics of the diode are closely related to square law.

$$V_2(t)=k_1\, V_1(t)+k_2V_1{}^2\,(t) \quad \text{Where, } k_1 \text{ and } k_2 \text{ are constants.}$$

Substituting $V_1(t)$ in Equation above

$V_2(t)=k_1[m(t)+A_c\cos(2\pi f_C t)]+k_2[m(t)+A_c\cos(2\pi f_c t)]^2$

$\Rightarrow V_2(t)=k_1m(t)+k_1A_c\cos(2\pi f_c t)+k_2m^2(t)+k_2A_c{}^2\cos^2 2\pi fct+2k_2m(t)A_c\cos(2\pi f_c t)$

$\Rightarrow V_2(t)=k_1m(t)+k_2m^2(t)+k_2A_C{}^2\cos^2(2\pi f_c t)+\mathbf{k_1Ac[1+(2k_2/k_1)m(t)]\cos(2\pi f_c t)}$

The last term of the above equation represents the desired AM wave and the first three terms of the above equation are unwanted. So, with the help of band pass filter, we can pass only AM wave and eliminate the first three terms.

Therefore, the output of square law modulator is

$$s(t)=k_1Ac[1+(2k_2/k_1)m(t)]\cos(2\pi f_c t)$$

The standard equation of AM wave is

$$s(t) = k_1A_c\,(1+\mu\cos(2\pi f_m t))\cos(2\pi f_c t)$$

where $\mu=(2k_2/k_1)=$ modulation index

AM DEMODULATION

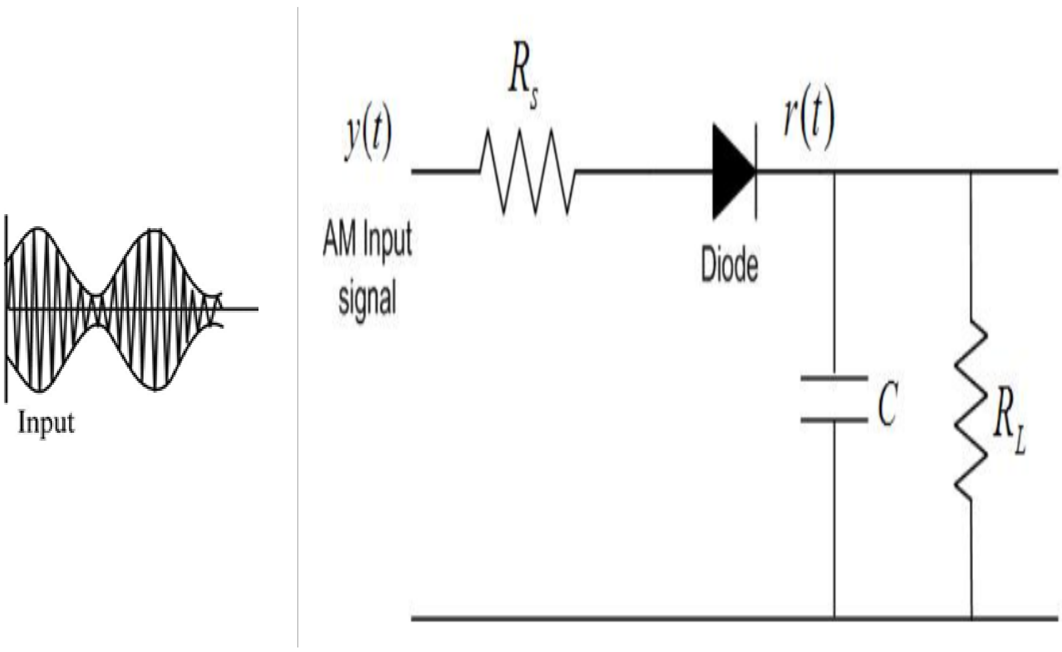
The process of extracting an original message signal from the modulated wave is known as detection or demodulation. The circuit, which demodulates the modulated wave is known as the demodulator. The following demodulators (detectors) are used for demodulating AM wave.

1) Square law detectors

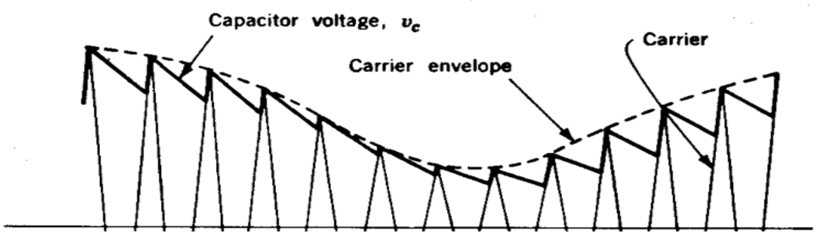
2) Linear diode or Envelope detectors

Linear Diode or Envelope Detector:

The diode operating in a linear region of its characteristics is used to detect the AM wave Envelope detector is used to detect (demodulate) high level AM wave. Following is theblock diagram of the envelope detector.



The output waveform developed across the resistor or capacitor (extracted message signal)is shown below



This envelope detector consists of a diode and low pass filter. Here, the diode is the maindetecting element. Hence, the envelope detector is also called as the diode detector. The low pass filter contains a parallel combination of the resistor and the capacitor.

In the positive half cycle of AM wave, the diode conducts and the capacitor charges to the peak value of AM wave through the resistor R_S. When the value of AM wave is less than this value, the diode will be reverse biased. Thus, the capacitor will discharge through resistor R_L till thenext positive half cycle of AM wave. When the value of AM wave is greater than the capacitor voltage, the diode conducts and the process will be repeated. We should select the component values in such a way that the capacitor charges very quickly and discharges very slowly. As a result, we will get the capacitor voltage waveform same as that of the envelope of AM wave, which is almost similar to the modulating signal.

To rapidly charge the capacitor to the peak value of iput signal,the charging time constant R_SC must be short compared to the carrier time period $1/f_c$, that is,

$$R_SC << 1/f_c$$

The discharging time constant R_LC must be long enough to ensure that the capacitor discharges slowly through the load resistance R_L between the positive peaks of the carrier wave that is,

$$1/f_c << R_LC << \frac{1}{BW}$$

The other type of modulation in continuous-wave modulation is **Angle Modulation**. Angle Modulation is the process in which the frequency or the phase of the carrier signal varies according to the message signal.

The standard equation of the angle modulated wave is

$$s(t)=A_C \cos\Phi_i(t)$$

where,Ac is the amplitude of the modulated wave, which is the same as the amplitude of the carrier signal and $\theta_i(t)$ is the angle of the modulated wave

Angle modulation is further divided into frequency modulation and phase modulation.

1) **Frequency Modulation** is the process of varying the frequency of the carrier signal linearly with the message signal.

2) **Phase Modulation** is the process of varying the phase of the carrier signal linearly with the message signal.

ANGLE MODULATION:

It is defined as the process in which the total phase angle of the carrier wave is varied in accordance with the instantaneous value of the modulating or message signal while keeping the amplitude of the carrier constant.

Let us consider that an unmodulated carrier signal is expressed as

$c(t)=Ac\cos(w_c t+\theta_0)$ where θ_0 is some phase angle

$w_c t+\theta_0=\Phi$ where Φ is the total phase angle of the carrier signal

Now, $c(t)=Ac\cos(w_c t+\theta_0)=Ac\cos(\Phi)$

If this phase angle $\Phi$is varied according to the instantaneous value of the modulating signal then the carrier signal is said to be angle modulated.

We can vary the phase angle in two ways and thus there are two types of modulation

1) Phase Modulation

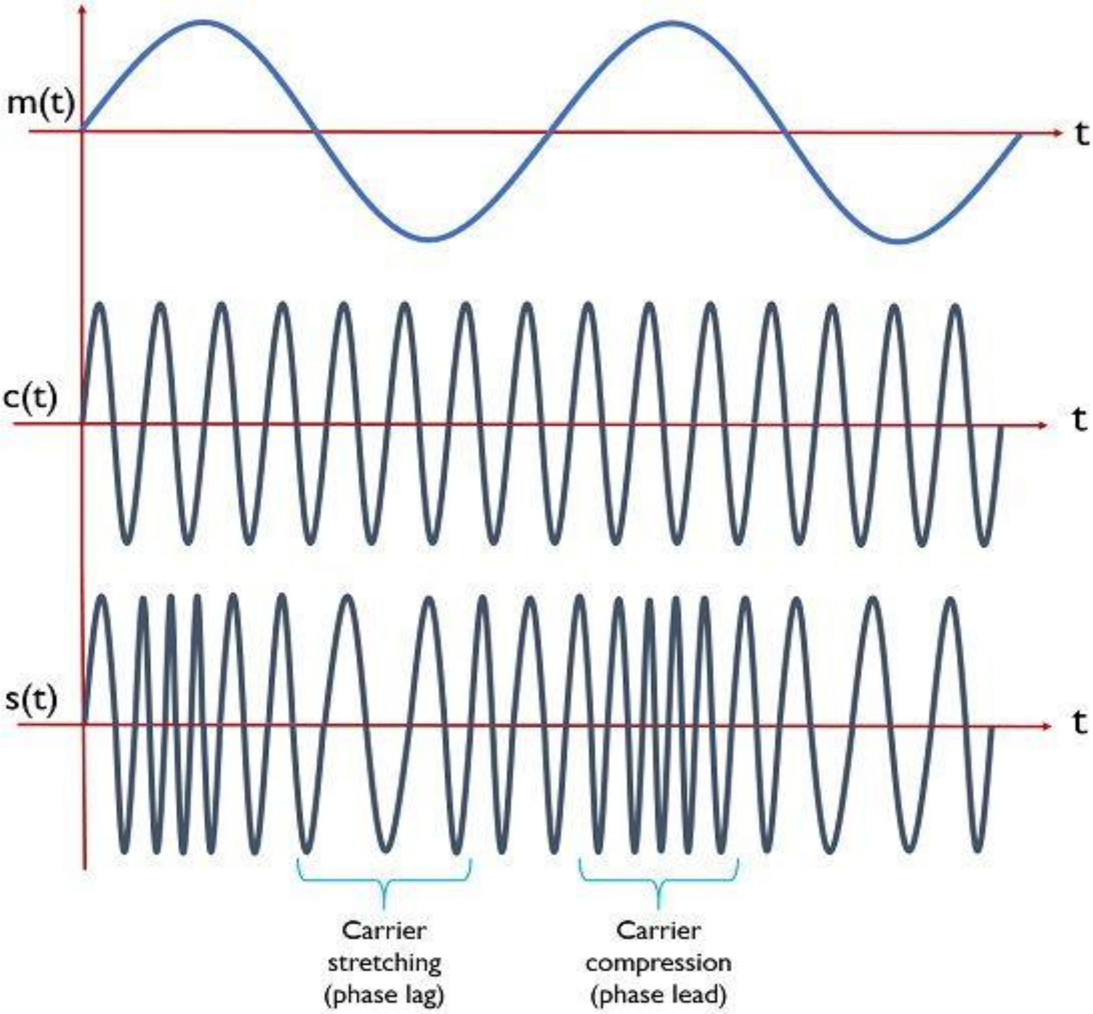
2) Frequency Modulation

Phase Modulation

Phase Modulation is that type of angle modulation in which the phase angle $\Phi$is varied linearly with the modulating signal m(t) about an unmodulated phase angle$(w_c t+\theta_0)$ .This means that in Phase Modulation, the instantaneous value of the phase angle denoted by φ_i is equal to the phase angle of the unmodulated carrier $(w_c t + \theta_0)$ plus a time varying component which is proportional to modulating signal m(t)

Instantaneous phase angle, $\varphi_i = w_c t + k_p\, m(t)$ (neglecting θ_0)

$s(t) = Ac \cos(\varphi_i) = Ac \cos(w_c t + k_p\, m(t))$ where k_p is the proportionality constant and is known as phase sensitivity of the modulator expressed in radians/volts.



Waveform for Phase Modulation

Here, it is clear from the above figure that when the amplitude of the sinusoidal signal starts to increase and reaches the maximum value, then the phase lead of the carrier signal gets increased.

Due to this a compression in the carrier signal is noticed. This resultantly increases the frequency of the signal.

However, when the amplitude of the modulating signal starts falling and attains a minimum value, then the phase lag of the carrier wave occurs. Thereby resultantly causing stretching of the signal. Due to this, the frequency of the signal gets increased.

So, in this way, we can say that with the change in phase of the signal during phase modulation, the frequency of the signal also shows some variation.

For a sinusoidal signal, the modulated signal is somewhat similar in case of both frequency and phase modulation

Frequency modulation (FM):

Frequency Modulation is that type of angle modulation in which the instantaneous frequency ω_i is varied linearly with the modulating signal about an unmodulated carrier frequency ω_c.This means that in Frequency Modulation, the instantaneous value of angular frequency ω_i denoted is equal to the carrier frequency ω_c plus a time varying component which is proportional to modulating signal m(t).

Instantaneous carrier frequency, $\omega_i = \omega_c + k_f\, m(t)$ where k_f is the proportionality constant and is known as frequency sensitivity of the modulator expressed in Hz/volt.

$$w_c t + \theta_0 = \Phi$$

Differentiating both the the sides of the above equation wrt to t,

$$w_c = \frac{d\varphi}{dt}$$

This time dependent angular velocity or angular frequency is known as instantaneous angular velocity or instantaneous angular frequency denoted by w_i.

The above equation becomes $w_i = \frac{d\varphi i}{dt}$ where w_i time dependent

From the above we get $\varphi i = \int w_i dt$

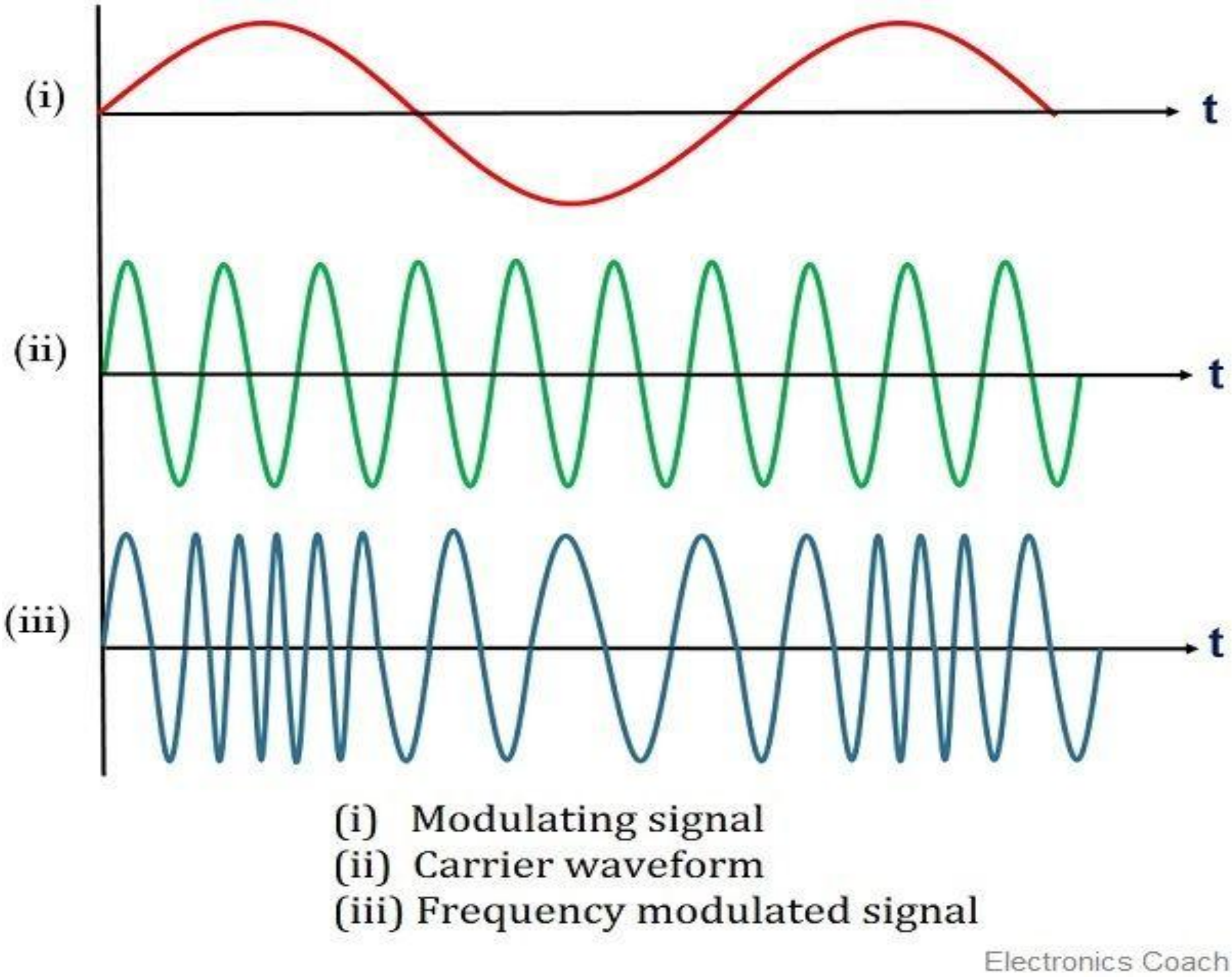
$$\text{-->}\ \varphi i = \int \left(\omega_c + k_f\, m(t)\right) dt$$

$$= \omega_c t + k_f \int m(t) dt$$

The expression for frequency modulated wave will be

$$S(t) = A_c \cos \Phi i$$

$$s(t) = A_C \cos[w_C t + k_f \int m(t)\, dt]$$



The frequency of the modulated wave increases, when the amplitude of the modulating or message signal increases. Similarly, the frequency of the modulated wave decreases, when the amplitude of the modulating signal decreases. Note that, the frequency of the modulated wave remains constant and it is equal to the frequency of the carrier signal, when the amplitude of the modulating signal is zero.

FREQUENCY DEVIATION OF FM WAVE:

The maximum change in instantaneous frequency from the carrier frequency w_c is called **frequency deviation** denoted by Δω which is given by

$$\Delta\omega = \left| k_f m(t) \right|_{max}$$

The total variation in frequency from lowest to highest point is called **carrier swing.**

The highest frequency attained by the modulated signal is equal to the resting or carrier frequency plus the frequency deviation.

$$f_H = f_c + \Delta f$$

The lowest frequency attained by the modulated signal is equal to the resting or carrier frequency minus the frequency deviation.

$$f_L = f_c - \Delta f$$

Carrier swing = 2x frequency deviation

Modulation index $m_f\ (\beta) = \frac{\Delta\omega}{w_m} = \frac{\Delta f}{f_m}$

Power of FM wave = $\frac{{A_C}^2}{2R}$

AM, FM and PM Waveforms:

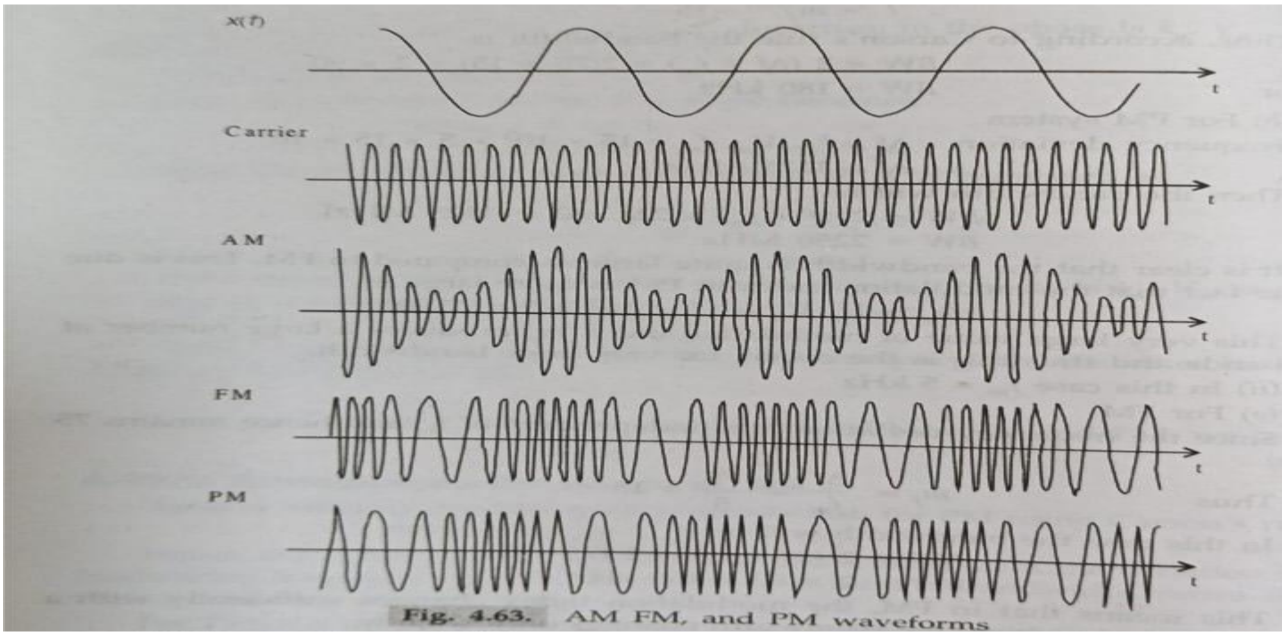


Fig. 4.63. AM FM, and PM waveforms

EQUATION OF SINGLE TONE FM

When **β<<1** then it is called as **narrowband FM** consisting essentially of acarrier, an upper side-frequency component or a lower side-frequency component.

When **β>>1** radian then it is called as **wideband FM** which contains a carrier and aninfinite number of side-frequency components located symmetrically around the carrier. The envelope of an FM wave is constant, so that the average power of such a wavedissipated in a 1-ohm resistor is also constant.

BANDWIDTH OF FM SIGNAL

Carson's rule provides a thumb formula to calculate the bandwidth of single tone wideband FM.According to this rule, the FM bandwidth is given as twice the sum of the frequency deviation and the highest modulating frequency i.e.,

$$\mathbf{BW=2(\Delta f+f_m)}$$

SINGLE TONE FREQUENCY MODULATION:

Let the modulating signal be $m(t)=A_m \cos 2\Pi f_m t=A_m \cos\omega_m t$

From the equation of FM Wave,

$$s(t)=A_C \cos[w_C t+ k_f\int m(t)\, dt]$$

$$s(t)=A_C \cos(\omega_c t+k_f\int Am \cos\omega m t)$$

$$=A_C \cos(\omega_c t+k_f A_m\frac{\sin \omega_m t}{w_m}$$

$$=A_C \cos(\omega_c t+\frac{\Delta w}{w_m}\sin w_m t)$$

$$=A_C \cos(\omega_c t+\beta\sin w_m t)$$

PROBLEMS

1) A single tone FM is represented by the following equation

$$S(t)=12\cos(6x10^8t+5\sin 1250t)$$

Find carrier frequency, modulating frequency, modulation index, maximum deviation and power delivered to a load of 10Ω.

Sol: Given equation of FM is

$$S(t)=12\cos(6x10^8t+5\sin 1250t)$$

Comparing it with standard equation of single tone FM

$$S(t)=A_C(2\Pi f_Ct+\beta\sin2\Pi f_mt)$$

A_C= 12V; $2\Pi f_C=6x10^8$; $\beta=5$; $2\Pi f_m=1250$

Carrier frequency= $f_C=\frac{6x10^8}{2\Pi}$=95.5MHz

Modulating frequency= $f_m=\frac{1250}{2\Pi}$= 199Hz

Modulation index=β=5

Maximum deviation(Δf)=?

$$\beta=\frac{\Delta f}{f_m}$$

$\Delta f=\beta x f_m=$ 5x199= 995Hz

Power of FM wave = $\frac{A_C{}^2}{2R}=\frac{12^2}{2x10}$= 7.2W

2) A 107.6MHz signal is frequency modulated by a 7kHz sine wave. The resultant FM signal has a frequency deviation of 50kHz. Determine the following

(i) The carrier swing of the FM signal

(ii) The highest and lowest frequencies attained by the modulated signal

(iii) The modulation index of FM wave

Sol: Given that f_c= 107.6MHz; f_m=7kHz; Δf=50kHz

(i) The carrier swing of the FM signal= 2xΔf=2x50k=100kHz

(ii) The highest and lowest frequencies attained by the modulated signal

The highest frequency= $f_H=f_c+\Delta f=107.6k+50k$

$=107.6x10^6+50x10^3=$ 107.65MHz

The lowest frequency= $f_c-\Delta f=107.6k-50k$

$=107.6x10^6-50x10^3=$ 107.55MHz

(iv) The modulation index of FM wave

$$\beta=\frac{\Delta f}{f_m}=\frac{50k}{7k}=7.143$$

3) A sinusoidal modulating waveform of amplitude 5 V and a frequency of 2 kHz is applied to FM generator, which has a frequency sensitivity of 40 Hz/volt. Calculate the frequency deviation, modulation index, and bandwidth?

Sol: Given $A_m=m(t)_{max}=5V$; $f_m=2kHz$; $k_f=40Hz/volt$

Frequency deviation ,

$$\Delta f=\left|\boldsymbol{k_f m(t)}\right|_{max}$$

$$= 40x5= 200Hz$$

Modulation index=$\beta=\frac{\Delta f}{f_m}=\frac{200}{2x1000}=0.1$

Bandwidth=$2(\Delta f+f_m)$= 2(200+2k)=2x2200=4400=4.4kHz

RELATION BETWEEN FM AND PM

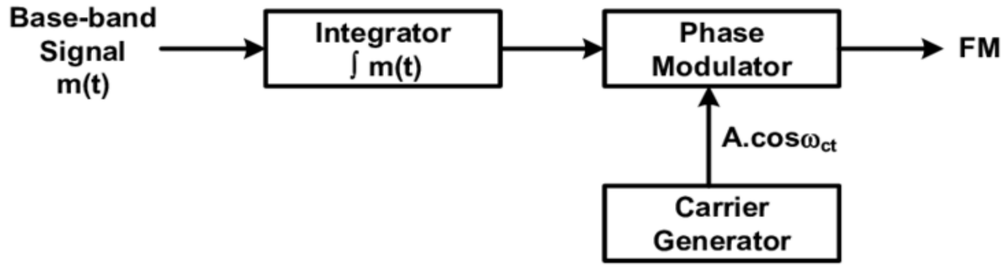
The expression for phase modulated wave (PM) is given by

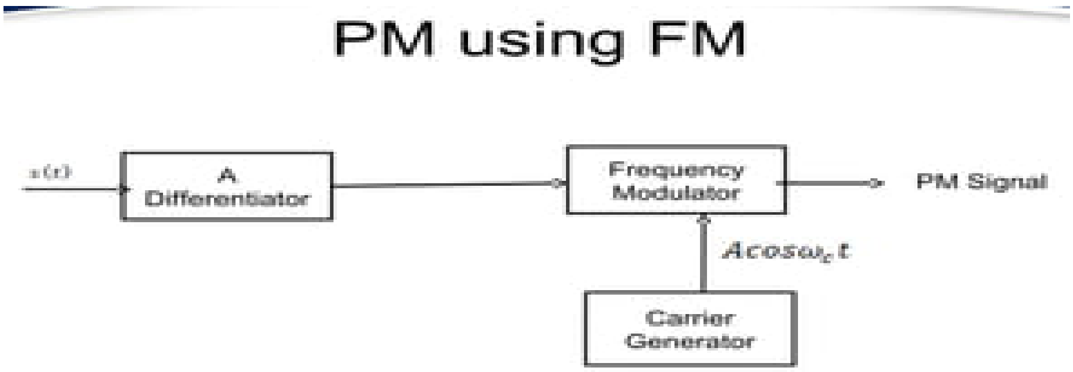
$$s(t) = Ac\cos(w_c t + k_p\, m(t))$$

The expression for frequency modulated wave (FM) is given by

$$s(t)=A_C\cos[w_C t+ k_f\int m(t)\, dt]$$

It may be observed from the above equations that PM and FM are closely related. In PM, the phase angle varies linearly with the message signal m(t) where as in FM, the phase angle varies linearly with the integral of baseband signal m(t). This means that FM wave may be obtained by using PM and PM wave may be obtained using FM.





Methods to generate FM Wave

The FM modulator circuits used for generating FM signals may be put into two categories as under

(i) The direct method or parameter variation method

(ii) The indirect method or the Armstrong method

The direct method or parameter variation method:

1. The Direct Method or Parameter Variation Method

In direct method or parameter variation method, the baseband or the modulating signal directly modulates the carrier. The carrier signal is generated with the help of an oscillator circuit. This oscillator circuit uses a parallel tuned L-C circuit. Thus the frequency of oscillation of the carrier generation is governed by the expression

$$\omega_c = \frac{1}{2\pi\sqrt{LC}}$$

Now, we can make the carrier frequency ω_c to vary in accordance with the baseband or modulating signal $x(t)$ or $m(t)$ if L & C is varied according to $m(t)$.

Varactor diode method for FM Generation

The varactor diode is a semiconductor diode whose junction capacitance changes with d.c. bias voltage. This varactor diode is connected with the tuned circuit of the carrier oscillator. This arrangement is shown in fig., below

Here the varactor diode C_d is connected in shunt (parallel) with the tank circuit.

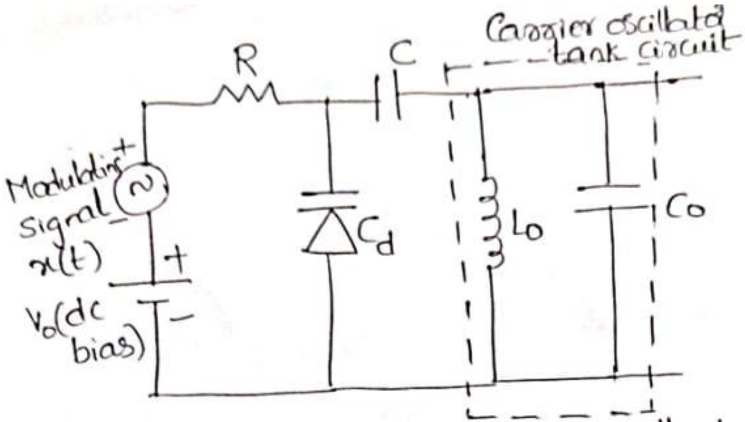


Fig: Varactor diode method of FM generation

In varactor diode FM generation arrangement, the capacitor C is made much smaller than the varactor diode capacitance C_d so that the radio frequency (RF) voltage from the oscillator across the diode is small as compared to reverse bias d.c. voltage across the varactor diode. In addition to this, the reactance of the capacitor C at the highest modulating frequency is made large enough compared to resistor R.

Mathematical Analysis

The capacitance C_d of the varactor diode is expressed as

$$C_d = \frac{K}{\sqrt{v_D}} = K v_D^{-1/2}$$

where v_D = total instantaneous voltage across the varactor diode and is given by $\quad K$ = constant of proportionality

$$v_D = V_0 + x(t)$$

The oscillation frequency is given as

$$\omega_C = \frac{1}{\sqrt{LC}}$$

Now, the total capacitance of the oscillator tank circuit will be $C_0 + C_d$ and thus the instantaneous frequency of oscillation ω_i is expressed as

$$\omega_i = \frac{1}{\sqrt{L_0(C_0+C_d)}} \qquad (\because C_d = K v_D^{-1/2})$$

$$= \frac{1}{\sqrt{L_0(C_0+K v_D^{-1/2})}}$$

We conclude that the instantaneous frequency ω_i of FM Signal depends upon v_D which in turn depends upon the value of the modulating signal $x(t)$. Thus the instantaneous oscillator frequency ω_i also depends on the base band or modulating signal $x(t)$ and hence frequency modulation is generated.

Drawbacks of Direct Method for FM Generation:

1. In direct method, it is not easy to get high order stability in carrier frequency. This is due to the fact that the generation of carrier signal is directly affected by

the modulating signal $x(t)$. The baseband signal directly controls the tank circuit of the carrier generator and thus a stable oscillator circuit cannot be used.

(ii) The non linearity of the varactor diode produces a frequency variation due to harmonics of the modulating or baseband signal and therefore the FM signal is distorted.

Despite all the above drawbacks, the direct method is utilized for high power FM generation in several applications

FM DEMODULATION:

Demodulation of FM waves

There are two methods of frequency demodulation:

1) Direct method
2) Indirect method

The direct method includes frequency discriminators and zero crossing detectors.

The phase locked loops (PLL) is an example of indirect method of frequency demodulation.

PLL FM Demodulator

The Phase locked loop (PLL) is a negative feedback system. It is primarily used to track the phase and frequency of the carrier component of an incoming FM signal. It consists of three major components:

1. A multiplier serving as phase detector or a phase comparator
2. A loop filter which is a low pass filter
3. A Voltage Controlled Oscillator (VCO)

A VCO is a sine wave generator whose frequency is determined by the voltage applied to it from an external source. In a VCO, the oscillation frequency varies linearly with its input voltage.

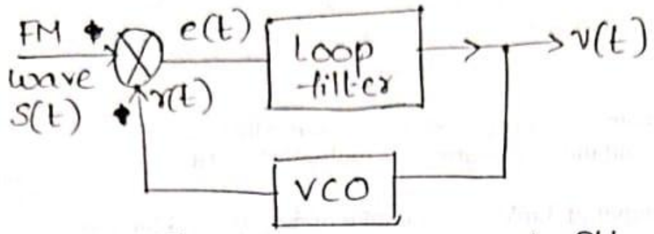


Fig: The block diagram of PLL

Operation:

The operation of a PLL is similar to any other feedback system. In any feedback system, the feedback signal tends to follow the input signal. If the signal fedback is not equal to the input signal, the error signal will change the value of fedback signal until it is equal to the input signal. The difference between s(t) and r(t) is called an error signal e(t). A PLL operates on the similar principle except for the fact that the quantity fedback is not the amplitude but a generalized phase $\phi(t)$.

Here e(t) = output of multiplier

This output e(t) is passed through a loop filter which is a low pass filter and then applied to the VCO as input. This error signal e(t) is utilized to adjust the VCO frequency in such a way that the instantaneous phase angle comes close to the angle of the incoming signal $s(t)$. At this point, the two signals $s(t)$ and $r(t)$ are in synchronism and the PLL is locked to the incoming signal $s(t)$.

The following mathematical steps helps us to understand how FM demodulation can be performed by using PLL.

Initially, the control voltage to VCO is zero, then it is adjusted so that

(i) The frequency of VCO is exactly made equal to the unmodulated carrier frequency f_c.

(ii) The VCO output has a phase shift of 90° with respect to the unmodulated carrier wave.

Mathematical Analysis

Suppose that the i/p signal $s(t)$ is applied to PLL which is an FM wave

$$s(t) = A_c \, \text{cosine} \left[2\pi f_c t + 2\pi K_f \int m(t)\,dt\right]$$

sine

$$= A_c \sin\left[2\pi f_c t + 2\pi K_f \int m(t)\,dt\right]$$

Let $\phi_1(t) = 2\pi K_f \int m(t)\,dt$

$$\therefore \; s(t) = A_c \sin\left[2\pi f_c t + \phi_1(t)\right]$$

If $r(t)$ denotes the output of VCO then

$$r(t) = A_v \cos\left[2\pi f_c t + 2\pi K_v \int v(t)\,dt\right]$$

where K_v is the frequency sensitivity of VCO measured in Hz/Volt

Let $\phi_2(t) = 2\pi K_v \int v(t)\,dt$

The incoming FM $s(t)$ and the VCO output $r(t)$ are the two inputs to multiplier.

The output $e(t)$ which is input to the loop filter will be $e(t) = K_m\, s(t) \cdot r(t)$ where K_m is multiplier gain

$$\therefore e(t) = K_m \left\{ A_c \sin\left[2\pi f_c t + \phi_1(t)\right] \right\} \left\{ A_v \cos\left[2\pi f_c t + \phi_2(t)\right] \right\}$$

$$= K_m A_c A_v \left\{ \sin\left[\underbrace{2\pi f_c t + \phi_1(t)}_{A}\right] \right\} \left\{ \cos\left(\underbrace{2\pi f_c t + \phi_2(t)}_{B}\right) \right\}$$

$$\sin A \cos B = \frac{1}{2}\left[\sin(A+B) + \sin(A-B)\right]$$

$$\Rightarrow e(t) = \frac{1}{2} K_m A_c A_v \left\{ \sin\left[4\pi f_c t + \phi_1(t) + \phi_2(t)\right] + \sin\left[\phi_1(t) - \phi_2(t)\right] \right\}$$

$$= \frac{K_m A_c A_v}{2} \left[\sin\left(4\pi f_c t + \phi_1(t) + \phi_2(t)\right) + \sin\left(\phi_1(t) - \phi_2(t)\right) \right]$$

This is the input to the loop filter which is a low pass filter which will attenuate the first term in the above equation which is a high frequency component

$\therefore$ Considering only the second term,

$$e(t) = \frac{1}{2} K_m A_c A_v \sin\left[\phi_1(t) - \phi_2(t)\right]$$

$$= \frac{1}{2} K_m A_c A_v \sin\left[\phi_e(t)\right]$$

where $\phi_e(t)$ is the phase error

The phase locked loop is said to be in phase lock when the phase error $\phi_e(t)$ is zero. When phase error $\phi_e(t)$ is less than one radian, we may use the approximation.,

$$\sin[\phi_e(t)] \approx \phi_e(t)$$

Now $\phi_e(t) = \phi_1(t) - \phi_2(t)$

Assuming again small error $\phi_e(t)$,

$$\Rightarrow \phi_1(t) \approx \phi_2(t)$$

$$\Rightarrow 2\pi K_f \int m(t)dt = 2\pi K_v \int v(t)\,dt$$

$$\Rightarrow K_f \int m(t)dt = K_v \int v(t)\,dt$$

Differentiating both sides w.r.t time

$$K_f\, m(t) = K_v\, v(t)$$

$$\Rightarrow v(t) = \frac{K_f}{K_v} m(t)$$

$$\therefore v(t) \propto m(t)$$

Thus the output $v(t)$ of low pass filter is proportional to the original modulating signal or in the other words, the input FM signal is demodulated and at the output of loop filter we get the modulating signal.

ADVANTAGES OF FM OVER AM

1) FM receivers may be fitted with amplitude limiters to remove the variations caused by noise.This makes FM reception more immune to noise than AM reception.

2) It is possible to reduce noise still further by increasing the frequency deviation. This is a feature which AM does not have because it is not possible to exceed 100 percent modulation without causing severe distortion.

3) Standard Frequency Allocations provide a guard band between commercial FM stations.Due to this, there is less adjacent channel interference than in AM.

4) FM broadcasts operate in the upper VHF and UHF frequency ranges at which there happens to be less noise than in MF and HF ranges occupied by AM broadcasts .

5) In FM, all the transmitted power is useful where as in AM, most of the power is carrier power which does not contain any information

PULSE MODULATION:

Pulse modulation is a technique in which the signal is transmitted in the form of pulses. Here, the carrier signal is a pulse wave. In Pulse modulation ,one or more characteristics of a pulse train is varied in accordance with the message signal.

Pulse modulation is divided into two types

1) Analog Pulse Modulation

2) Digital Pulse Modulation

ANALOG PULSE MODULATION:

In analog pulse modulation, a periodic pulse train is used as carrier signal to modulate a continuous-time message signal. One or more properties of this pulse train is varied continuously according to the corresponding sample of message signal. The information is transmitted in analog form but transmission is done at discrete times.

There are basically three types of analog pulse modulation. They are

1) Pulse Amplitude Modulation

2) Pulse Time Modulation

The two main types of Pulse Time Modulation are

1) Pulse Width(PWM) or Pulse Duration Modulation(PDM)

2) Pulse Position Modulation(PPM)

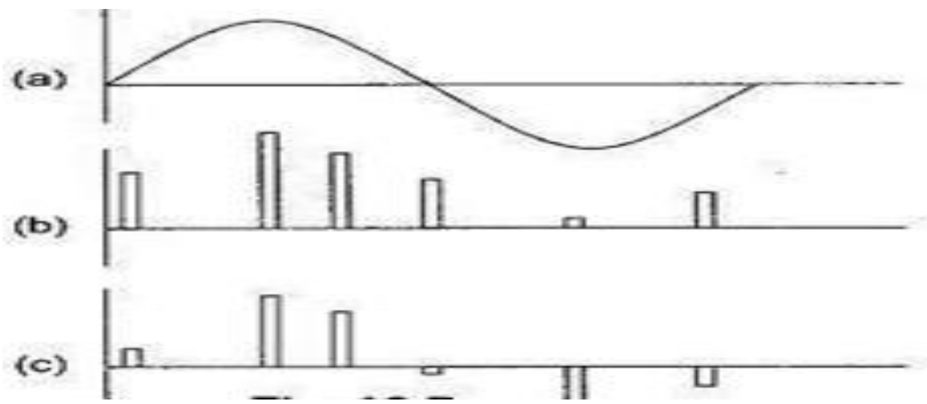
PULSE AMPLITUDE MODULATION (PAM):

In PAM, the amplitude of a periodic pulse train is varied in accordance with corresponding sample values of a continuous message signal. Pulse amplitude modulation is categorized into two types:

1) Single Polarity PAM: Single polarity PAM is a situation where a fixed DC level is added to the signal to ensure that all the pulses are positive.

2) Double Polarity PAM: Double polarity PAM is a situation where the pulses are both positive and negative.

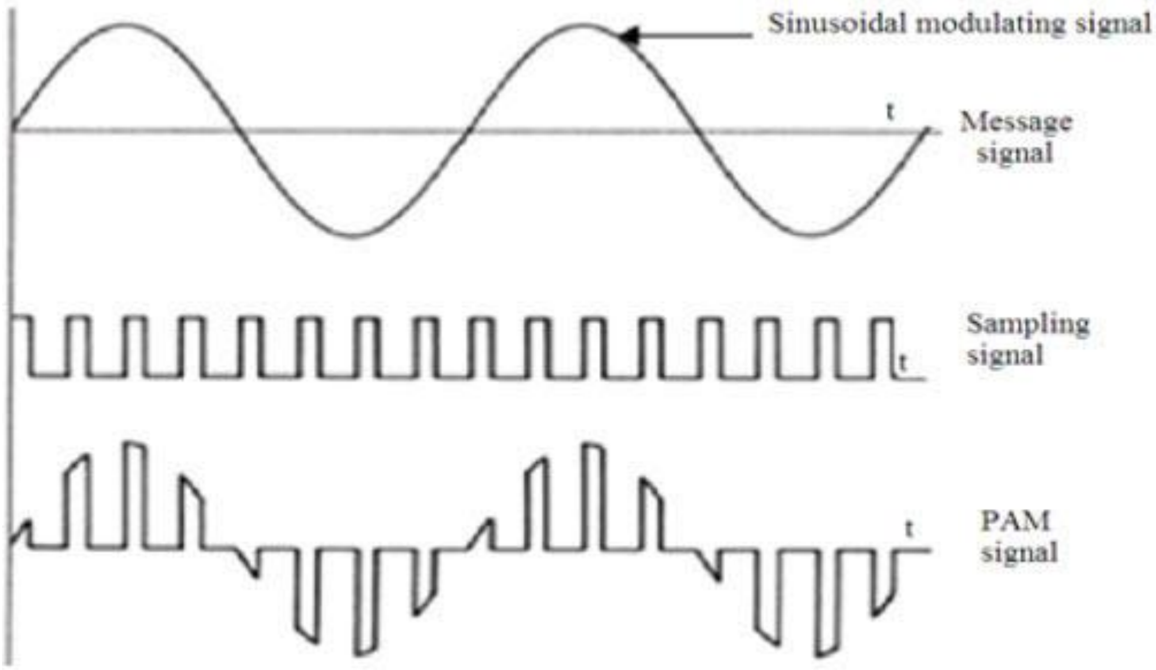
PAM wave with single polarity and double polarity are shown in the figure



below:

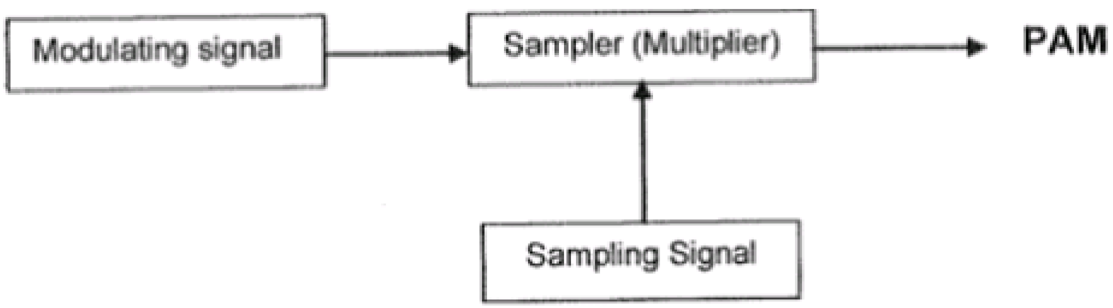
Fig: Single & Double Polarity PAM Signal

Pulse amplitude modulation is a technique in which the amplitude of each pulse is controlled by the instantaneous amplitude of the modulation signal. It is a modulation system in which the signal is sampled at regular intervals and each sample is made proportional to the amplitude of the signal at the instant of sampling.



Generation of PAM:

The following figure shows the generation of PAM signal from the sampler which has two inputs i.e. modulating signal and sampling signal or carrier pulse.



Nyquist Shannon Sampling Theorem:

Sampling theorem states that a continuous time signal can be represented in its samples and can be recovered back when sampling frequency f_s is greater than or equal to the twice the highest frequency component of message signal f_m.i.e,

$$f_s \geq 2f_m$$

Sampling rate or sampling frequency defines the number of samples per second (or per other unit) taken from a continuous signal to make a discrete or digital signal.

The minimum sampling rate required to accurately capture an analog signal in digital form without information loss is known as **Nyquist Frequency** or **Nyquist Limit.** It is defined as twice the maximum frequency component present in the analog signal. Mathematically it can be represented as $\boldsymbol{f_s = 2f_m}$

The Nyquist interval, also known as the Nyquist period, is the time interval between consecutive samples in a digital signal or digital sampling system. It is the reciprocal of the Nyquist rate, which is the smallest sampling rate required to accurately capture an analog signal in digital form without information loss. Mathematically it can be represented as:

$$T = \frac{1}{Nyquist\ Frequency}$$

Types of Sampling

1. Ideal Sampling: Ideal sampling, also known as impulse or Dirac sampling, is a theoretical notion in which samples of a continuous signal are taken at specific time intervals, often at the Dirac delta function impulse points.

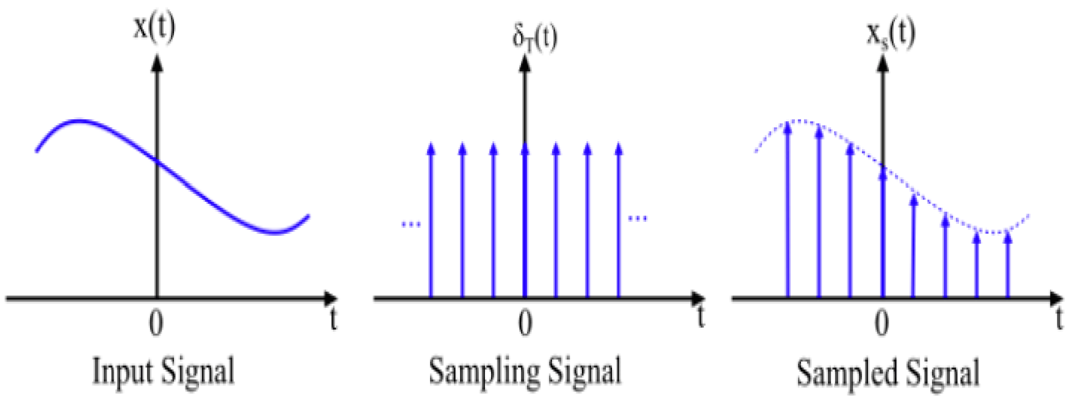


Fig: Ideal Sampling

2)Natural Sampling:

In Natural sampling, also known as zero-order hold sampling, each sample is taken in natural sampling by retaining the value of the continuous signal constant for the duration of the sampling period.

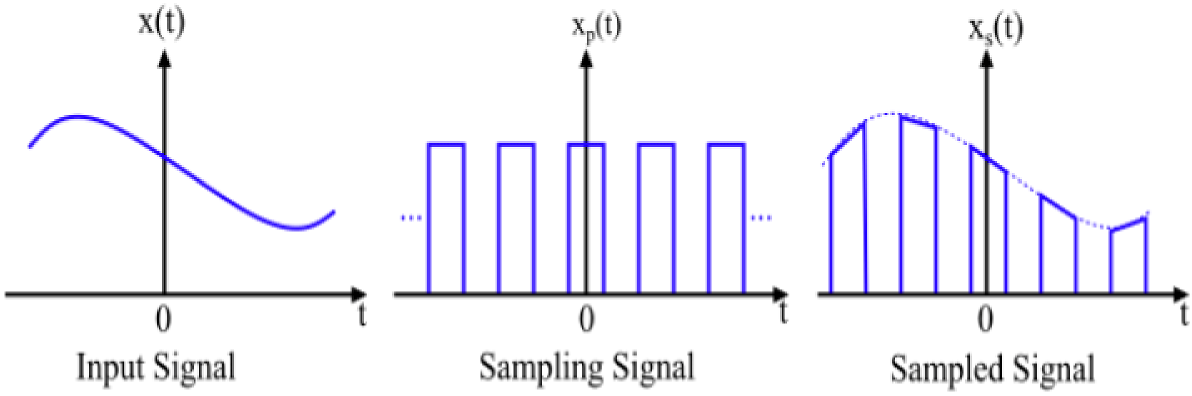


Fig: Natural Sampling

3) FLAT TOP SAMPLING:

Flat-top sampling is a type of natural sampling in which each sample is obtained by maintaining the value of the continuous signal constant for a set period of time, resulting in a flat-top waveform. Instead of retaining the value for the whole sample interval, flat-top sampling holds it only for a portion of the interval while allowing it to change at the beginning and end.

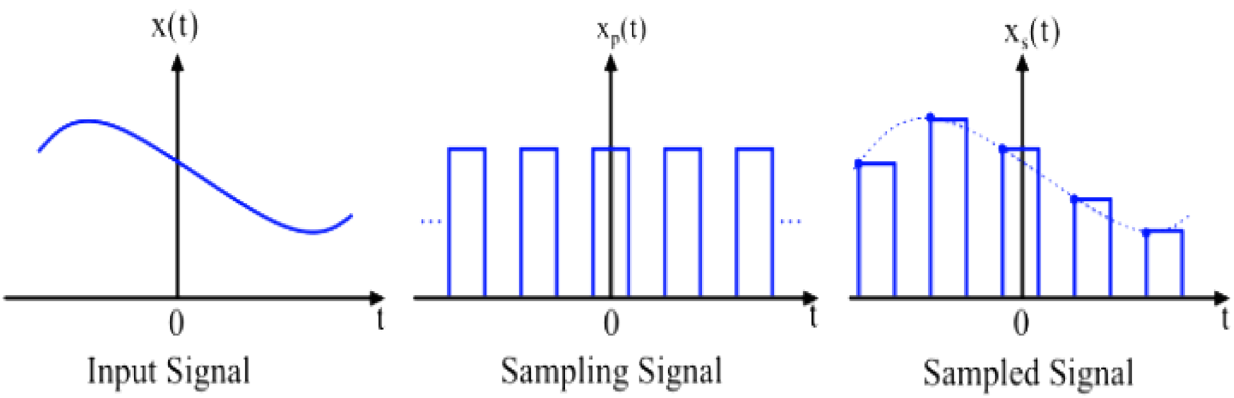


Fig: Flat Top Sampling

There are two types of sampling techniques for transmitting a signal using PAM. They are:

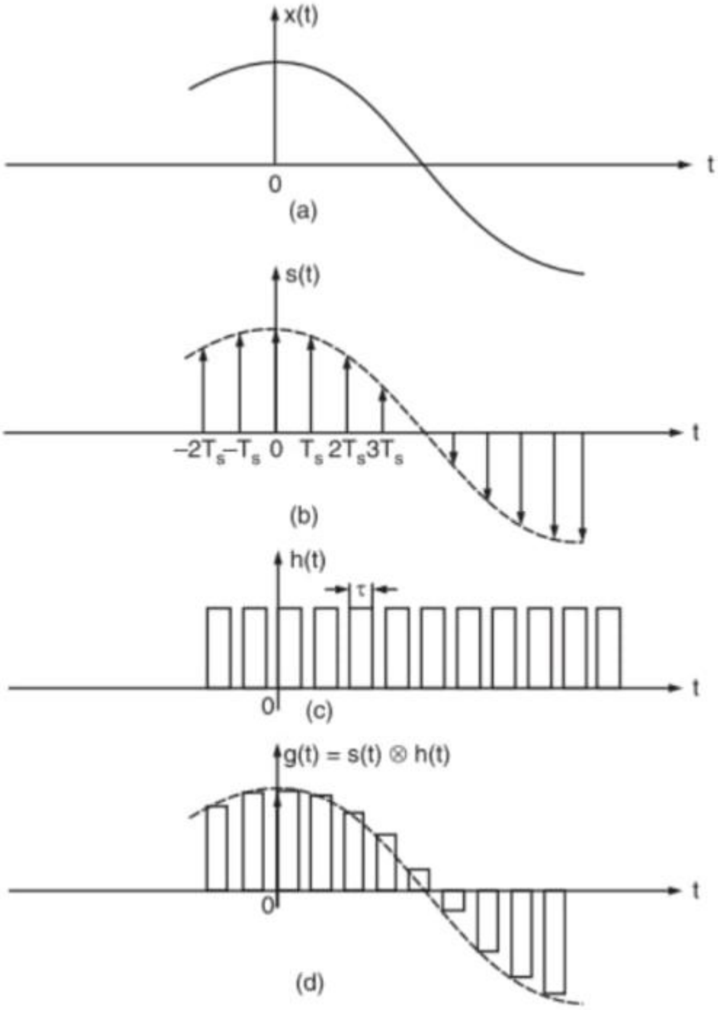
1) Flat Top PAM

2) Natural PAM

Flat Top PAM Generation:

In case of natural samples PAM signal, the pulse has varying top in accordance with the signal variation. Now, when such type of pulse is received at the receiver, it is always contaminated by noise.

Then it becomes quite difficult to determine the shape of the top of the pulse and thus amplitude detection of the pulse is not exact.Due to this, errors are introduced in the received signal.Therefore, flat top sampled PAM is widely used.



The amplitude of each pulse is directly proportional to modulating signal amplitude at the time of pulse occurrence. The amplitude of the signal cannot be changed with respect to the analog signal to be sampled. The tops of the amplitude remain flat.The fig shown below shows the sample and hold circuit to produce flat top sampled PAM and the waveform for flat top sampled PAM.

A sample and hold circuit shown in fig (a) is used to produce Flat top sampled PAM. The working principle of this circuit is quite easy.

The sample and Hold (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 G1 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 G2 of the other transistor.

Due to this, the capacitor ‘C’ is discharged to zero volts. The discharges 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 samples as shown in fig(b) below

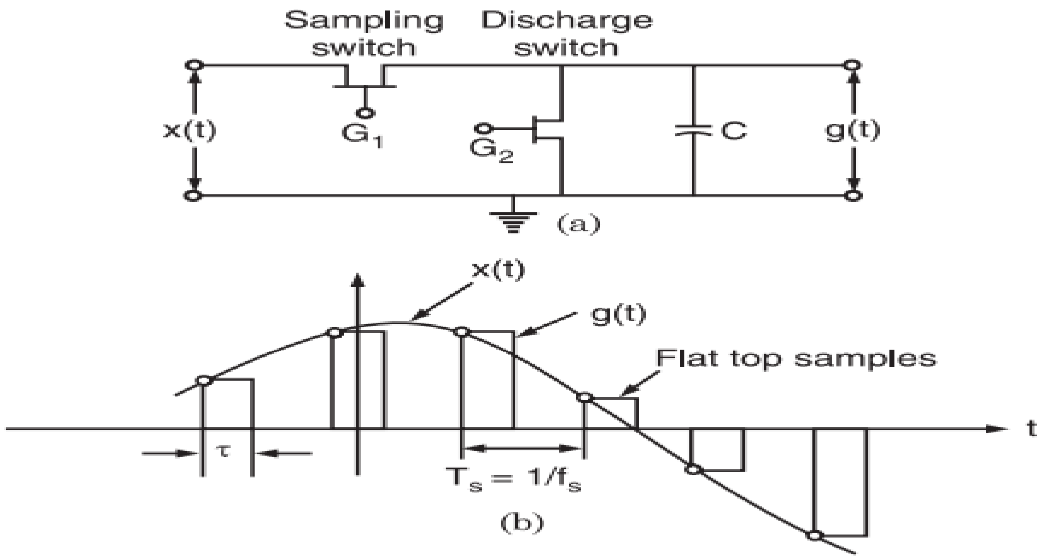


Fig:Flat Top PAM Modulator and its output waveform

PAM DEMODULATOR

For pulse amplitude modulated (PAM) signals, the demodulation is done using a Holding circuit. The fig shows the block diagram of a PAM demodulator.

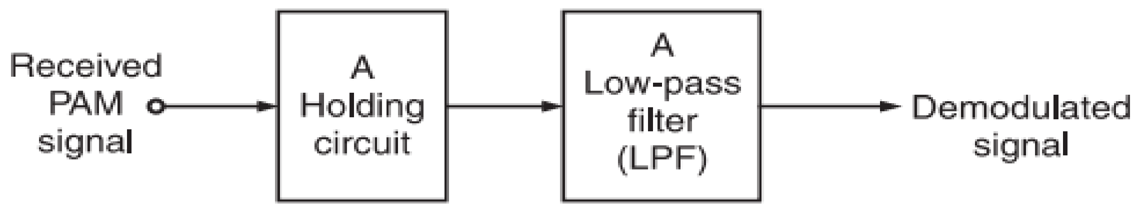


Fig: Flat Top PAM Demodulator

In this method, the received PAM signal is allowed to pass through a Holding circuit and a low pass filter (LPF) as shown in fig..

Now, the fig shown below illustrates a very simple holding circuit.

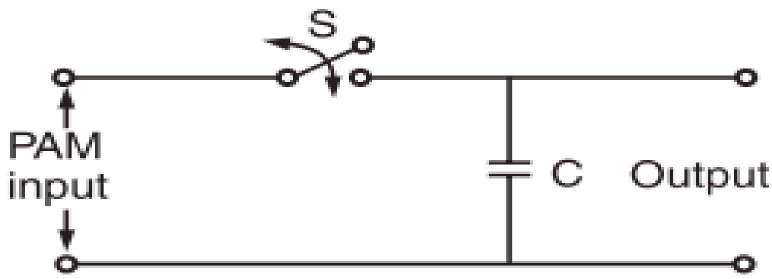
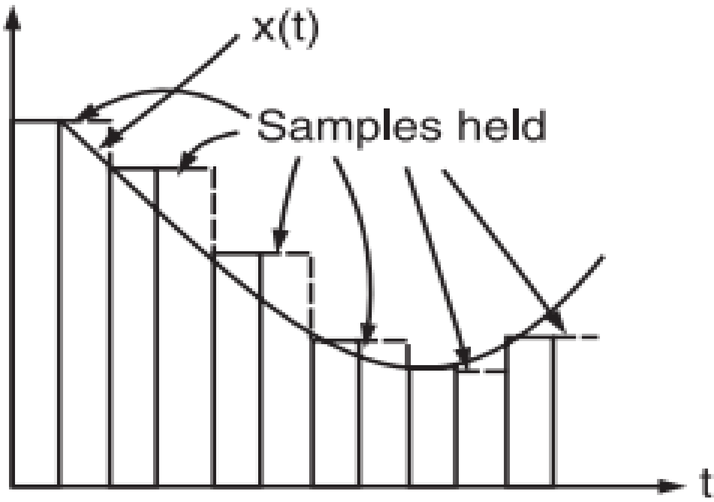


Fig: Holding Circuit

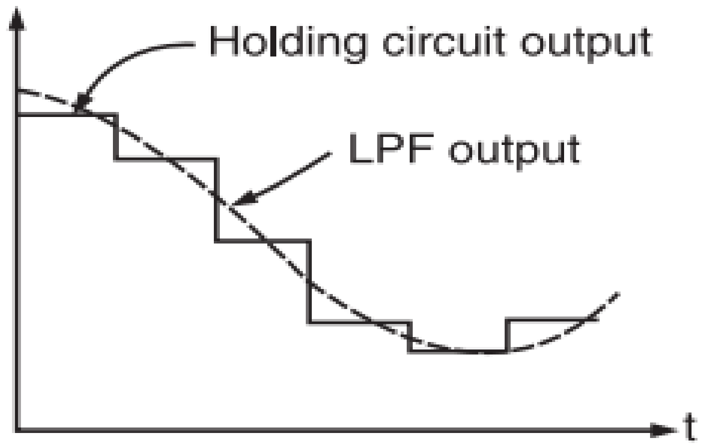
Here the switch S is closed after the arrival of pulse and it is opened at the end of the pulse.In this way the capacitor C is charged to the pulse amplitude value and holds this value during the interval between the two pulses.

After this the holding circuit output is smoothened in Low pass filter as shown in fig below

It may be observed that some kind of distortion is introduced due to the holding circuit. In fact the circuit of fig.shown above 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 which is shown below



After this the holding circuit output is smoothened in Low Pass filter as shown in fig below

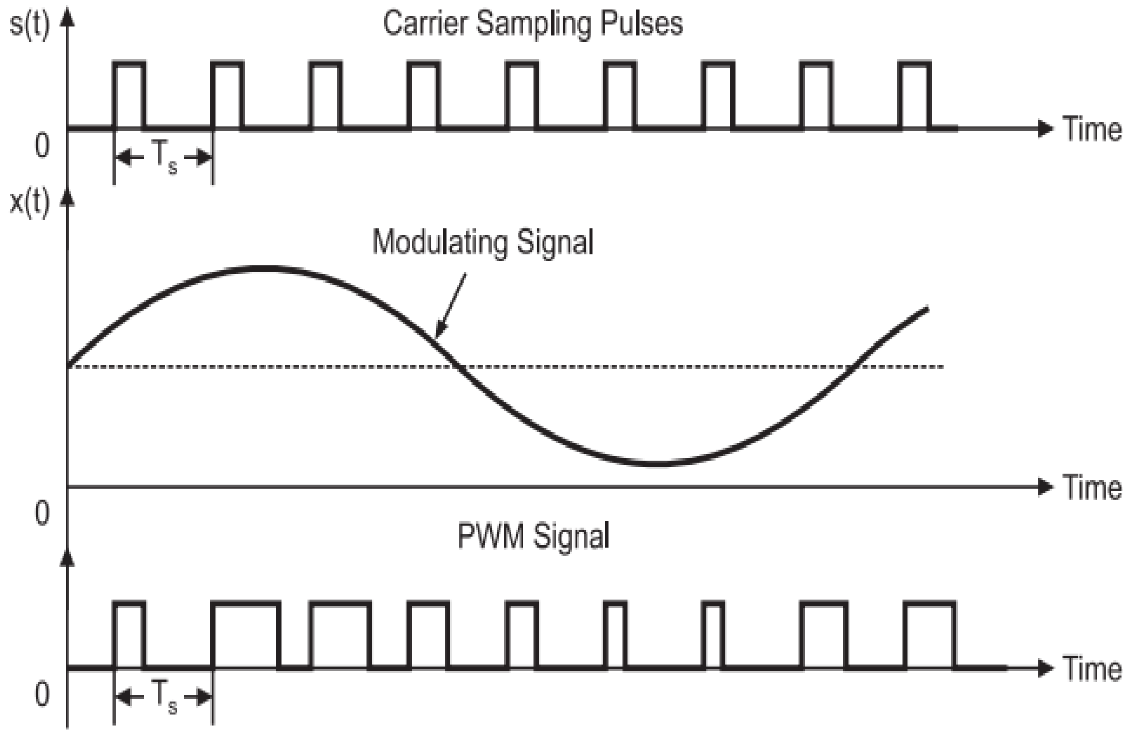


Pulse Width Modulation

In **Pulse Width Modulation (PWM)** or Pulse Duration Modulation (PDM) or Pulse Time Modulation (PTM) technique, the width or the duration or the time of the pulse carrier varies, which is proportional to the instantaneous amplitude of the message signal.

The width of the pulse varies in this method, but the amplitude of the signal remains constant. Amplitude limiters are used to make the amplitude of the signal constant. These circuits clip off the amplitude to a desired level, and hence the noise is limited.Thus, the PWM system is more immune to noise than the PAM signal.

The waveform of PWM is shown in fig below:



Generation of PWM Signal:

The block diagram of a PWM signal generator is shown in fig. below. This circuit can also be used for the generation of PPM signal.

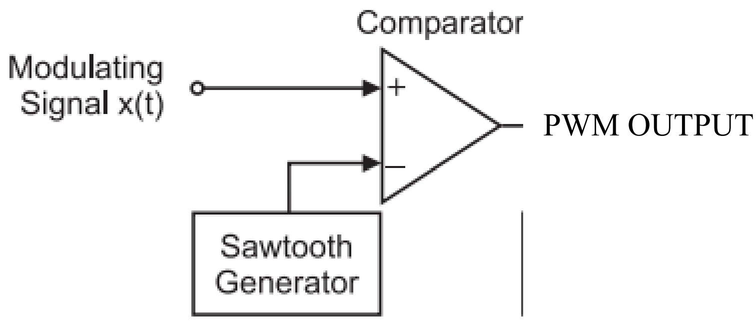
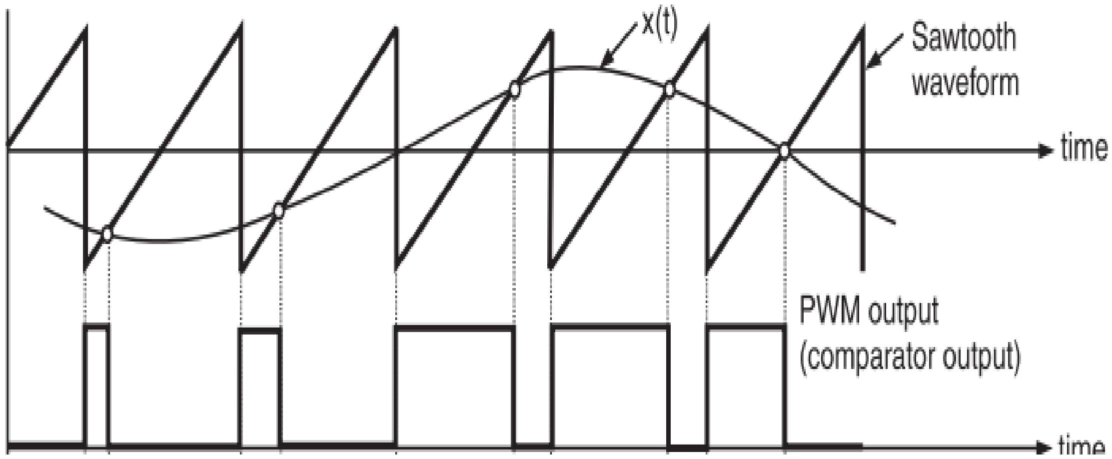


Fig: PWM generator

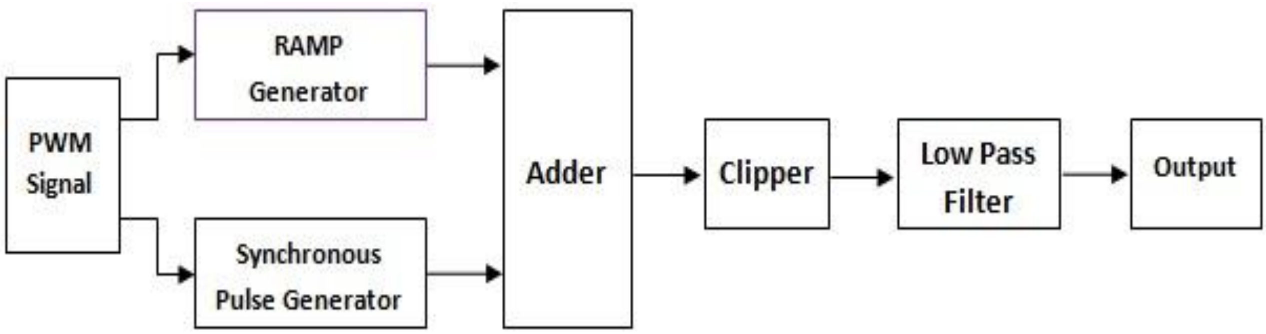
- A sawtooth generator generates a sawtooth signal of frequency fs, and this sawtooth signal in this case is used as a sampling signal.
- It is applied to the inverting terminal of a comparator.
- The modulating signal x (t) is applied to the non-inverting terminal of the same comparator.

- The comparator output will remain high as long as the instantaneous amplitude of x (t) is higher than that of the ramp signal.
- This gives rise to a PWM signal at the comparator output as shown in fig.



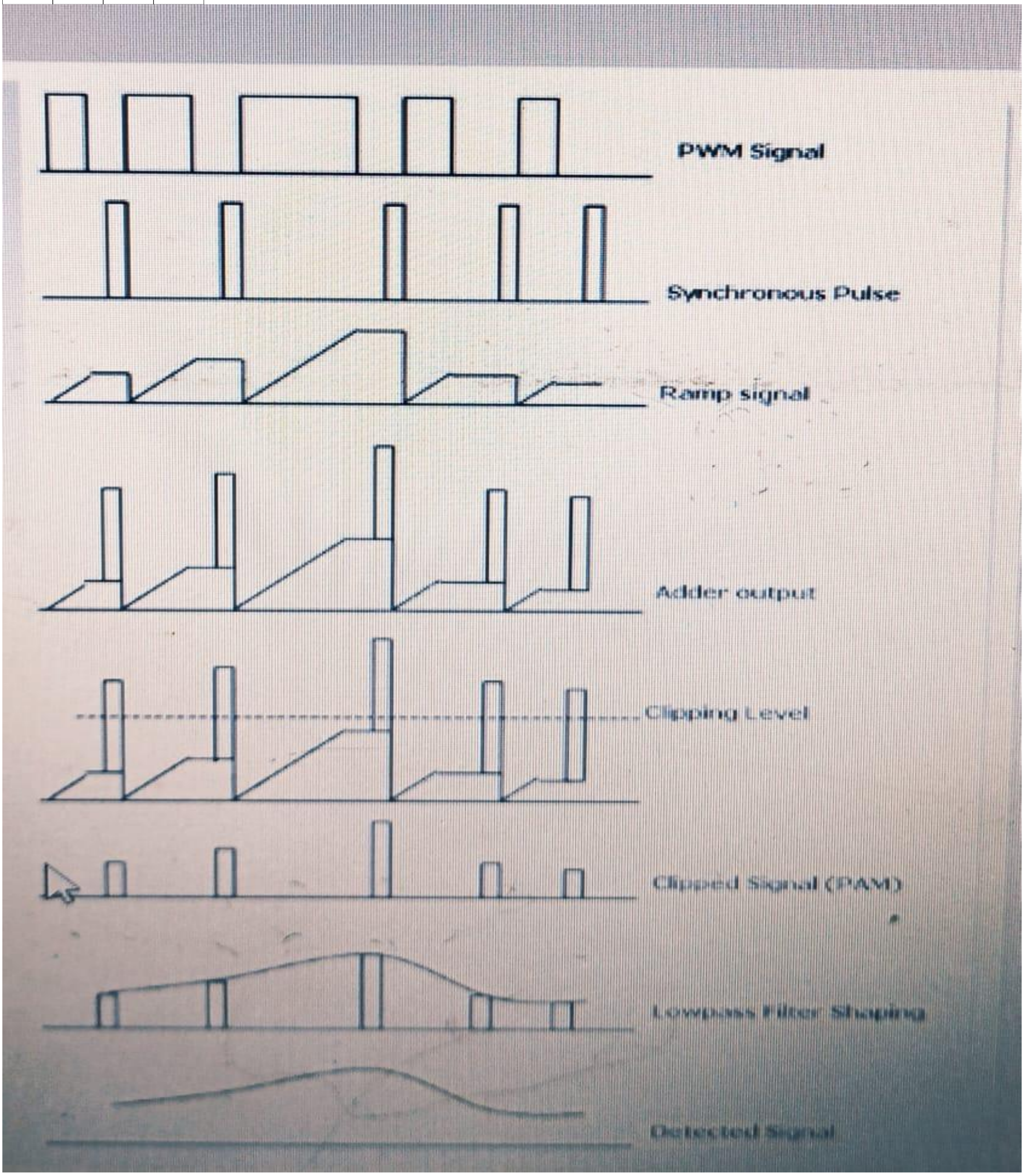
PWM DEMODULATOR

The PWM demodulator circuit is shown below:



The input PWM wave is applied to Ramp generator and pulse generator. Synchronous pulse generator will generate a pulse waveform such that the pulse will end at the beginning of each PWM pulse.Ramp generator will produce a ramp signal whose amplitude is proportional to width of the PWM signal.Apply these Ramp and Synchronous pulse to an Adder circuit which adds these signals together.The next block is a positive Clipper with a specific voltage; Clipper clips the waveform at a particular level.The output of clipper will be PAM signal, now the PWM signal gets converted to PAM signal.

Ramp Generator + Synchronous Pulse + Adder + Clipper = PWM to PAM Converter. The PAM can be demodulated by Low Pass filtering method.



Introduction to Digital Communication

The communication that occurs in our day-to-day life is in the form of signals. These signals, such as sound signals, generally, are analog in nature. When the communication needs to be established over a distance, then the analog signals are sent through wire, using different techniques for effective transmission.

The conventional methods of communication used analog signals for long distance communications, which suffer from many losses such as distortion, interference, and other losses including security breach.

In order to overcome these problems, the signals are digitized using different techniques. The digitized signals allow the communication to be more clear and accurate without losses.

The following figure indicates the difference between analog and digital signals. The digital signals consist of 1s and 0s which indicate High and Low values respectively.

Advantages of Digital Communication

As the signals are digitized, there are many advantages of digital communication over analog communication, such as

1) The effect of distortion, noise, and interference is much less in digital signals as they are less affected.

2) Digital circuits are more reliable.

3) Digital circuits are easy to design and cheaper than analog circuits.

4) The hardware implementation in digital circuits, is more flexible than analog.

5) The occurrence of cross-talk is very rare in digital communication.

6) The signal is unaltered as the pulse needs a high disturbance to alter its properties, which is very difficult.

7) Signal processing functions such as encryption and compression are employed in digital circuits to maintain the secrecy of the information.

8) The probability of error occurrence is reduced by employing error detecting and error correcting codes.

9) Spread spectrum technique is used to avoid signal jamming.

10) Combining digital signals using Time Division Multiplexing TDM is easier than combining analog signals using Frequency Division Multiplexing FDM.

11) The configuring process of digital signals is easier than analog signals.

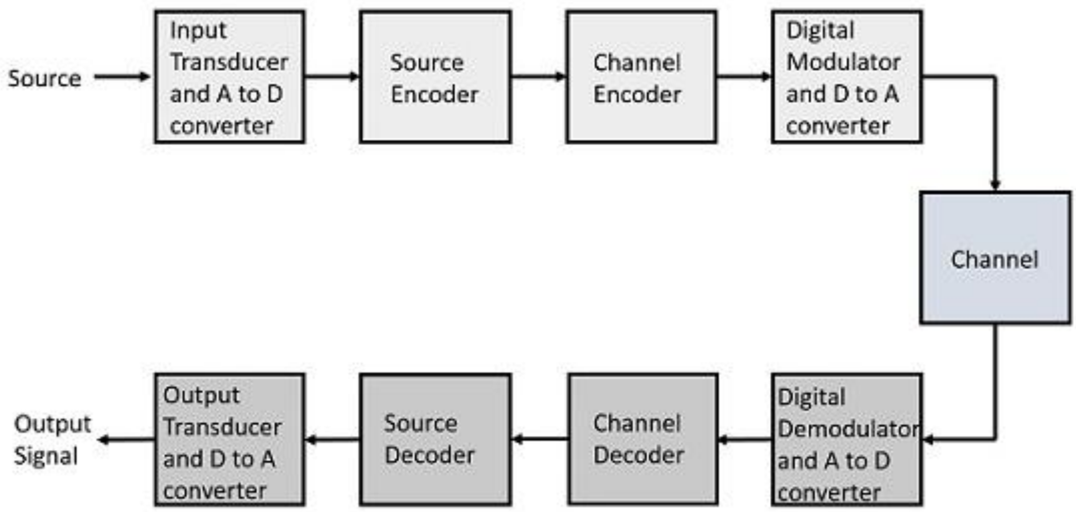
12) Digital signals can be saved and retrieved more conveniently than analog signals.

13) Many of the digital circuits have almost common encoding techniques and hence similar devices can be used for a number of purposes.

14) The capacity of the channel is effectively utilized by digital signals.

Elements of Digital Communication

The elements which form a digital communication system is represented by the following block diagram for the ease of understanding.



Basic Elements of a Digital Communication System

Source

The source output may be either an analog signal, such as an audio or video signal, or a discrete signal, such as the output of a teletype machine, that is discrete in time and has a finite number of output characters.

Input Transducer

This is a transducer which takes a physical input and converts it to an electrical signal (Example: microphone). This block also consists of an analog to digital converter where a digital signal is needed for further processes.

A digital signal is generally represented by a binary sequence.

Source Encoder

The source encoder compresses the data into minimum number of bits. This process helps in effective utilization of the bandwidth. It removes the redundant bits and unnecessary excess bits.

Channel Encoder

The channel encoder, does the coding for error correction. During the transmission of the signal, due to the noise in the channel, the signal may get altered and hence to avoid this, the channel encoder adds some redundant bits to the transmitted data. These are the error correcting bits.

Digital Modulator

In analog modulation, the amplitude, frequency and phase of carrier wave changes with respect to amplitude of message signal whereas in Digital Modulation a process called as Shift Keying is used.

Shift Keying means that the amplitude, frequency or phase of the carrier wave is shifted between two or more discrete values rather than varying continuously like Analog Modulation. Binary data requires two discrete levels of amplitude, frequency or phase for modulation called as Binary Shift Keying.

There are mainly three types of Digital Modulation techniques. They are :

1) Amplitude Shift Keying

2) Frequency Shift Keying

3) Phase Shift Keying

The received signal is demodulated as well as converted again from analog to digital. The signal gets reconstructed here.

Channel Decoder

The channel decoder, after detecting the sequence, does some error corrections. The distortions which might occur during the transmission, are corrected by adding some redundant bits. This addition of bits helps in the complete recovery of the original signal.

Source Decoder

The resultant signal is once again digitized by sampling and quantizing so that the pure digital output is obtained without the loss of information. The source decoder recreates the source output.

Output Transducer

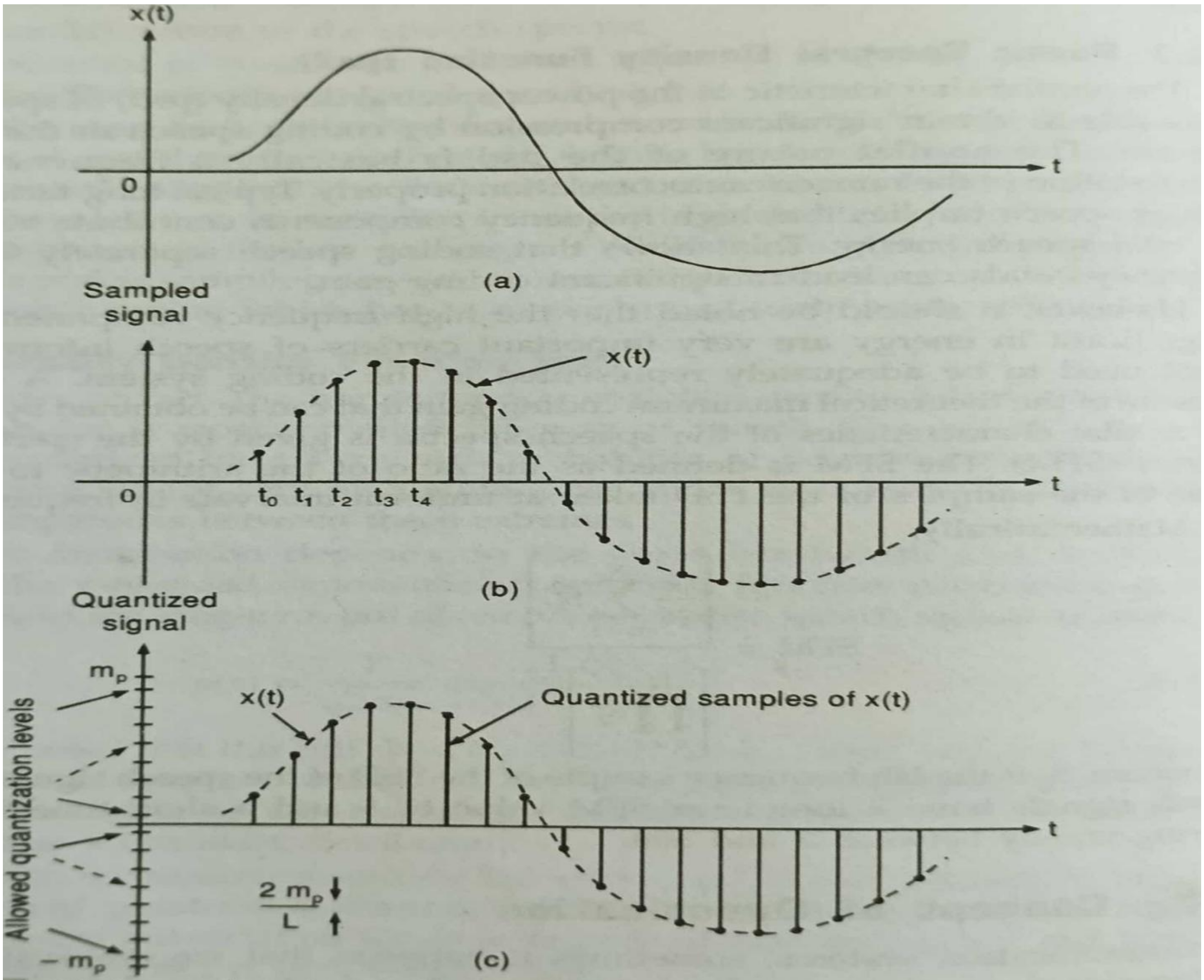
This is the last block which converts the signal into the original physical form, which was at the input of the transmitter. It converts the electrical signal into physical output (Example: loud speaker).

Output Signal

This is the output which is produced after the whole process. Example − The sound signal received.

Concept of Quantization

In communication it is happen that we are available with the analog signal but we have to transmit a digital signal for a particular application. In such cases, we have to convert an analog signal into digital signal. This means that we have to convert a continuous time signal in the form of digits. To see how a signal can be converted from analog to digital form, let us consider an analog signal as shown in figure(a) below



(*a*) An analog signal, (*b*) Samples of analog signal, (*c*) Quantization.

First of all, we get samples of this signal according to sampling theorem. For this purpose, we mark the time-instants t_0, t_1, t_2 and so on, at equal time-intervals along the time axis. At each of these time-instants, the magnitude of the signal is measured and thus samples of the signal are taken. Figure (b) shows a representation of the signal of figure 6.3(a) in terms of its samples.

Now, we can say that the signal in figure (b) is defined only at the sampling instants. This means that it no longer is a continuous function of time, but rather, it is a discrete-time signal. However, since the magnitude of each sample can take any value in a continuous range, the signal in figure (b) is still an analog signal.

This difficulty is neatly resolved by a process known as **quantization**. In quantization, the total amplitude range which the signal may occupy is divided into a number of standard levels.

As shown in figure (c), amplitudes of the signal x(t) lie in the range ($-m_p, m_p$) which is partitioned into L intervals, each of magnitude

$\Delta v = \frac{2m_p}{L}$ Now, each sample is approximated or rounded off to the nearest quantized level as shown in figure 6.3. Since each sample is now approximated to one of the L numbers therefore the information is digitized.

The quantized signal is an approximation of the original one. We can improve the accuracy of the quantized signal to any desired degree simply by increasing the number of levels L.

Quantization Techniques

As a matter of fact, a q-level quantizer compares the discrete-time input $x(nT_S)$ with its fixed digital levels. It assigns any one of the digital level to $x(nT_S)$ with its fixed digital levels. It then assigns any one of the digital level to $x(nT_S)$ which results in minimum distortion or error. This error is called **quantization error**. Thus, the output of a quantizer is a digital level called x $x_q(nT_s)$. Basically, the quantizers are of four types as under:

(i) Uniform quantization

(ii) Non-uniform quantization

(iii) Adaptive quantization

(iv) Vector quantization.

(i) Uniform Quantizer

A uniform quantizer is that type of quantizer in which the 'step size' remains same throughout the input range.

(ii) Nonuniform Quantizer

A non-uniform quantizer is that type of quantizer in which the 'step-size' varies according to the input signal values.

PULSE CODE MODULATION

Pulse-code modulation is known as a digital pulse modulation technique. In fact, the pulse-code modulation (PCM) is quite complex compared to the analog pulse modulation techniques (i.e., PAM, PWM and PPM) in the sense that the message signal is subjected to a great number of operations. The following figure shows the basic elements of a PCM system.

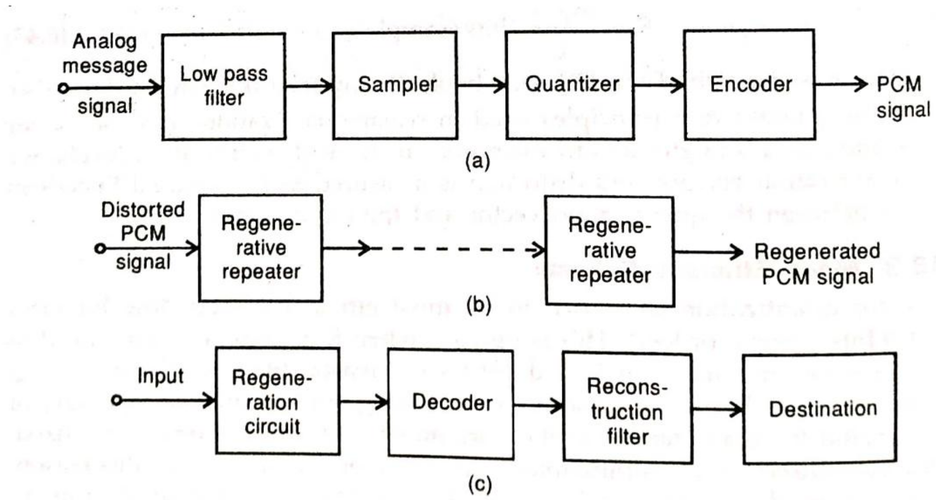


Fig (a) PCM Transmitter (b) Transmission path (c) PCM Receiver

It consists of three main parts i.e., transmitter, transmission path and receiver. The essential operations in the transmitter of a PCM system are sampling, quantizing and encoding as shown in figure 6.14. As discussed earlier, sampling is the operation in which an analog (i.e., continuous- time) signal is sampled according to the sampling theorem resulting in a discrete- time signal. The quantizing and encoding operations are usually performed in the same circuit which is known as an analog-to-digital converter (ADC).

Also, the essential operations in the receiver are regeneration of impaired signals, decoding and demodulation of the train of quantized samples. These operations are usually performed in the same circuit which is known as a digital-to-analog converter (DAC).

Further, at intermediate points, along the transmission route from the transmitter to the receiver, regenerative repeaters are used to reconstruct (i.e., regenerate) the transmitted sequence of coded pulses in order to combat the accumulated effects of signal distortion and noise.

As discussed in article 6.3, the quantization refers to the use of a finite set of amplitude levels and the selection of a level nearest to a particular sample value of the message signal as the representation for it. In fact, this operation combined with sampling, permits the use of coded pulses for representing the message signal. Thus, it is the combined use of quantizing and coding that distinguishes pulse code modulation form analog modulation techniques.

Now, let us summarize PCM in the form of few points as under:

(i) PCM is a type of pulse modulation like PAM, PWM or PPM but there is an important difference between them PAM, PWM or PPM are "analog" pulse modulation systems whereas PCM is a "digital" pulse modulation system.

(ii) This means that the PCM output is in the coded digital form. It is in the form of digital pulses of constant amplitude, width and position. (iii) The information is transmitted in the form of "code words". A PCM system consists of a PCM encoder (transmitter) and a PCM decoder (receiver).

(iv) The essential operations in the PCM transmitter are sampling, quantizing and encoding.

(v) All the operations are usually performed in the same circuit called as analog-to-digital converter.

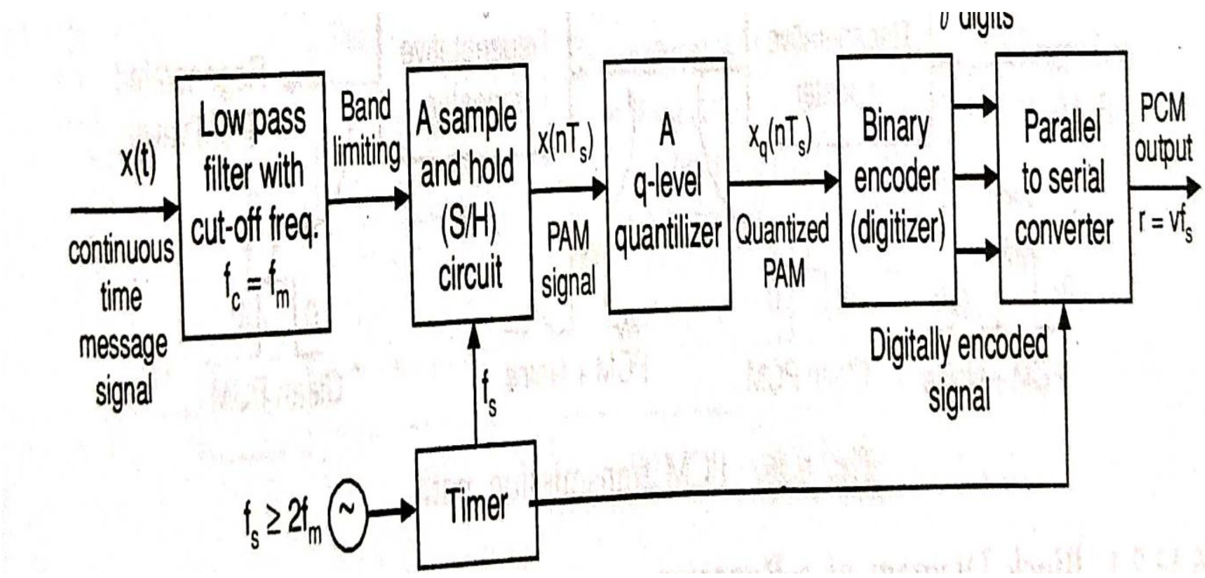
(vi) It should be understood that the PCM is not modulation in the conventional

sense.

(vii) Because in modulation, one of the characteristics of the carrier is varied in proportion with the amplitude of the modulating signal. Nothing of that sort happen in PCM.

A PCM Generator or Transmitter

The following fig shows a practical block diagram of a PCM generator.



The signal x(t) is first passed through the low-pass filter of cutoff frequency f_m Hz. This low-pass filter blocks all the frequency components which are lying above f_m Hz.

This means that now the signal x(t) is bandlimited to f, Hz. The sample and hold circuit then samples this signal at the rate of fs. Sampling frequency f, is selected sufficiently above nyquist rate to avoid aliasing i.e., fs≥ 2fm

In figure shown above, the output of sample and hold circuit is denoted by $x(nT_s)$. This signal $x(nT_s)$ is discrete in time and continuous in amplitude. A q-level quantizer compares input $x(nT_{s,})$ with its fixed digital levels. It assigns any one of the digital level to $x(nT_{s,})$ with its fixed digital levels. It then assigns any one of the digital level to $x(nT_s)$ which results in minimum distortion or error. This error is called quantization error. Thus, output of quantizer is a digital level called $x_q(nT_s)$.

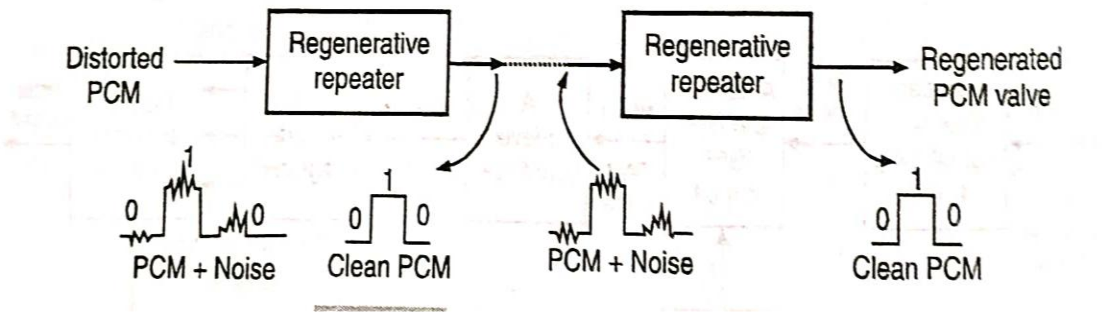
Now, the quantized signal level $x_q(nT_s)$ is given to binary encoder. This encoder converts input signal to 'v' digits binary word. Thus $x_q(nT_s)$ is converted to 'v' binary bits. This encoder is also known as digitizer.

It is not possible to transmit each bit of the binary word separately on transmission line. Therefore 'v' binary digits are converted to serial bit stream to generate single baseband signal. In a parallel to serial converter, usually a shift register does this job. The output of PCM generator is thus a single baseband signal of binary bits.

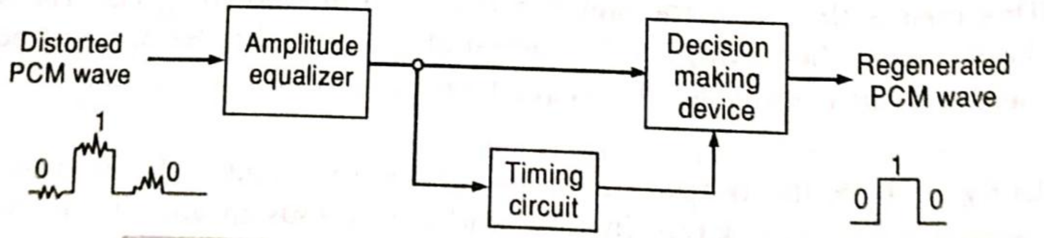
Also, an oscillator generates the clocks for sample and hold circuit and parallel to serial converter. In the pulse code modulation generator discussed above, sample and hold, quantizer and encoder combinely form an analog to digital converter (ADC).

PCM TRANSMISSION PATH

The Path between the PCM transmitter and PCM receiver over which the PCM signal travel, is called as PCM transmission path and it is as shown in figure below



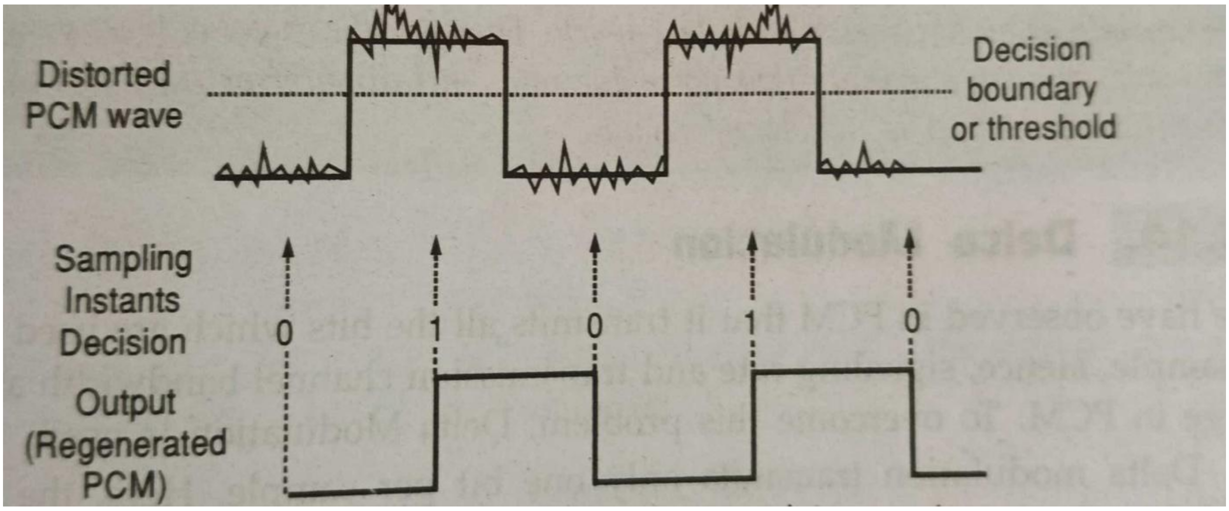
The most important feature of PCM system lies in its ability to control the effects of distortion and noise when the PCM wave travels on the channel. PCM accomplishes this capacity by means of using a chain of regenerative repeaters as shown in figure below.



Such repeaters are spaced close enough to each other on the transmission path. The regenerative performs three basic operations namely equalization, timing and decision making. Hence, each repeater actually reproduces the clean noise free PCM signal from the PCM signal distorted by the channel noise. This improves the performance of PCM in presence of noise.

The amplitude equalizer shapes the distorted PCM wave so as to compensate for the effects of amplitude and phase distortions. The timing circuit produces a periodic pulse train which is derived from the input PCM pulses. This pulse train is then applied to the decision making device. The decision making device uses this pulse train for sampling the equalized PCM pulses. The sampling is carried out at the instants where the signal to noise ratio is maximum.

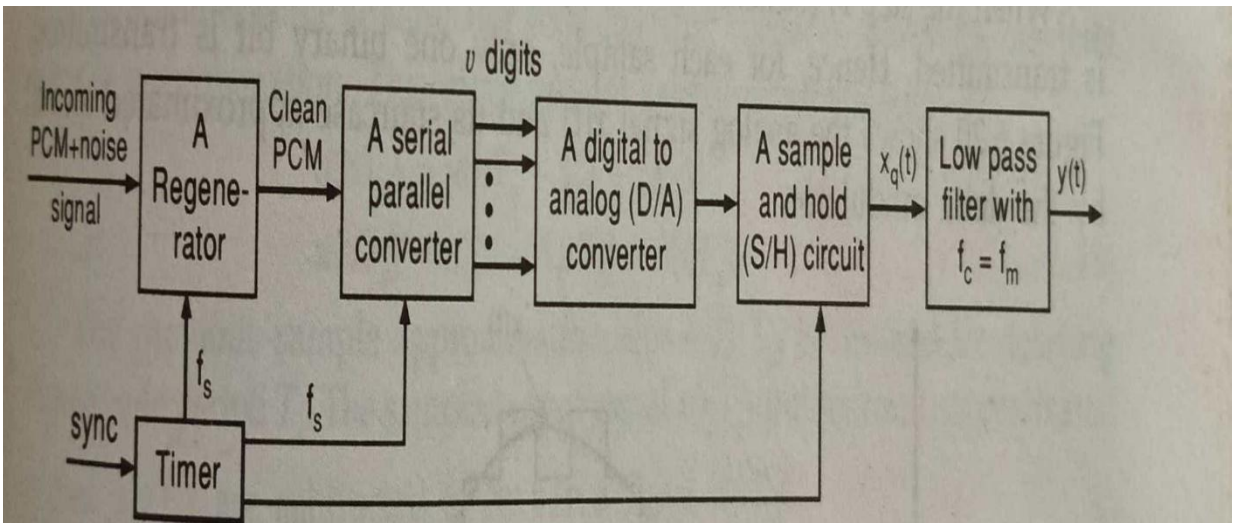
The decision device makes a decision about whether the equalized PCM wave at its input has a 0 value or 1 value at the instant of sampling. Such a decision is made by comparing equalized PCM with a reference level called decision threshold as illustrated in figure below



At the output of the decision device, we get a clean PCM signal without any trace of noise.

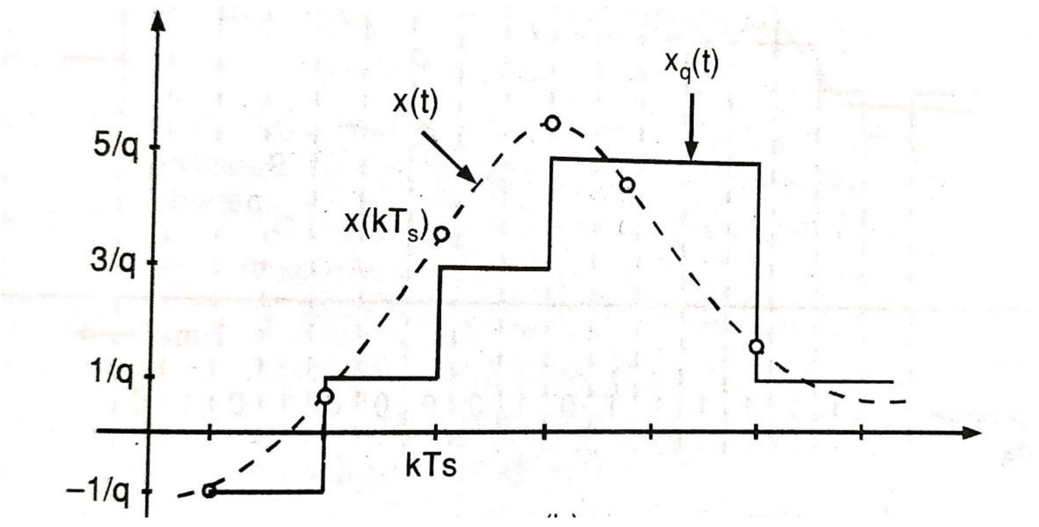
<u>PCM RECEIVER</u>

The following figure shows the block diagram of PCM receiver.



The regenerator at the start of PCM receiver reshapes the pulse and removes the noise. This signal is then converted to parallel digital words for each sample.Now, the digital word is converted to its analog value denoted as $x_q(t)$ with the help of a sample and hold circuit. This signal, at the output of sample and hold circuit, is allowed to pass through a lowpass reconstruction filter to get the appropriate original message signal denoted as y(t).

The reconstructed signal is shown in figure below



It is impossible to reconstruct exact original signal x(t) because of permanent quantization error introduced during quantization at the transmitter. In fact, this quantization error can be reduced by increasing the binary levels. This is equivalent to increasing binary digits (bits) per sample. But increasing bits 'v,' increases transmission bandwidth.Therefore the choice of these parameters is made, in such a manner that noise due to quantization error (i.e., also called as quantization noise) is in tolerable limits