MAR GREGORIOS COLLEGE OF ARTS & SCIENCE

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DEPARTMENT OF ELECTRONICS & COMMUNICATION SCIENCE

SUBJECT NAME: PRINCIPLES OF COMMUNICATION

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UNIT I

INTRODUCTION TO FOURIER TRANSFORM - properties of Fourier Transform - Sampling theorem – Natural Sampling & Flat-top Sampling (Qualitative analysis)

UNIT II

AMPLITUDE MODULATION & DEMODULATION – Block diagram of Communication System – Types ofCommunication Systems – Need for Modulation – Amplitude Modulation – Definition & Representation – Generation of Amplitude Modulation (Balanced modulator) – Generation of SSB-SC AM (Frequency discriminator method) – Generation of VSB – Detector – AM demodulator – FDM

AM TRANSMITTER – Block diagram of AM Transmitter – definition of low level &high level modulation – Superheterodyne receiver – General Characteristics of receiver.

UNIT III

FREQUENCY MODULATION – Representation of FM – Generation of FM – Direct method (Varactor diodemodulator) – indirect method (Armstrong method) – FM detection – slope detector – Foster seeley discriminator.

FM TRANSMITTER – Direct method & Armstrong method – FM super heterodyne receiver – Pre-emphasis &De-emphasis – Comparison of AM & FM -

UNIT IV

ANALOG PULSE CODE MODULATION - Generation & Detection of PAM, PWM & PPM.

DIGITAL PULSE MODULATION & DEMODULATION – PCM – Quantizing & Coding – Generation &Demodulation of PCM – Companding& encoding – Applications of PCM – Basic Concept of DM & ADM.

UNIT V

DIGITAL COMMUNICATION – TDM in PCM – Binary Systems – ASK – FSK and PSK – Detection of DigitalCommunication Signals. Introduction to FDM.

UNIT I

INTRODUCTION TO FOURIER TRANSFORM

Properties of Fourier Transform - Sampling theorem –Natural Sampling & Flat-top Sampling (Qualitative analysis)

1.0INTRODUCTION

Fourier series is a periodic function composed of harmonically related sinusoids, combined by a weighted summation. With appropriate weights, one cycle (or *period*) of the summation can be made to approximate an arbitrary function in that interval (or the entire function if it too is periodic). As such, the summation is a **synthesis** of another function. The discrete-time Fourier transform is an example of Fourier series. The process of deriving weights that describe a given function is a form of Fourier **analysis**. For functions on unbounded intervals, the analysis and synthesis analogies are Fourier transform and inverse transform.

1 Properties of Fourier Transform

	OriginalFunction	TransformedFunction
1.Linear	$af_1(t)+bf_2(t)$	$aF_1(j\omega)+bF_2(j\omega)$
2.Scaling	$f(at)$	$\frac{1}{a}F\left(\frac{j\omega}{a}\right)$
3.TimeShift	$f(t-t_0)$	$e^{-j\omega t_0}F(j\omega)$
4.FrequencyShift	$e^{at}f(t)$	$F(j(\omega-a))$
5.ReverseTime	$f(-t)$	$F(-j\omega)$
6.TimeDifferentiation	$\frac{df(t)}{dt}$, $\frac{d^nf(t)}{dt^n}$	$j\omega F(j\omega)$ $(j\omega)^n F(j\omega)$
7.TimeIntegral	$\int_{-\infty}^{t} f(\tau)d\tau$	$\frac{1}{j\omega}F(j\omega)$
8. FrequencyDifferentiation	$tf(t)$ $t^nf(t)$	$j\frac{dF(j\omega)}{d\omega}$

t-multiplication		$j^n \frac{d^n}{d\omega^n} F(j\omega)$
9. FrequencyIntegral t−division	$\frac{f(t)}{t}$	$\int^{\infty} \frac{F(j\omega)d\omega}{j\omega}$

LINEAR PROPERTY

$$ax_1(t)+bx_2(t) \longleftrightarrow aX_1(j\omega)+bX_2(j\omega)$$

Linear

Proof.

$$F\{ax_1(t)+bx_2(t)\} = \int_{-\infty}^{\infty} (ax_1(t)+bx_2(t))e^{-j\omega t}dt$$

$$= a\int_{-\infty}^{\infty} x_1(t)e^{-j\omega t}dt + b\int_{-\infty}^{\infty} x_2(t)e^{-j\omega t}dt = aX_1(j\omega) + bX_2(j\omega)$$

1. **Scaling**

$$x(at) \longleftrightarrow \frac{1}{a} X\left(\frac{j\omega}{a}\right)$$

***Proof*.**

$$\mathcal{F}\{x(at)\} = \int_{-\infty}^{\infty} x(at) e^{-j\omega t}\, dt$$

Let $at=\tau,\ t=\frac{\tau}{a},\ dt=\frac{1}{a}d\tau,\ t\in(-\infty,+\infty) \Rightarrow \tau\in(-\infty,+\infty)$

$$= \int_{-\infty}^{\infty} x(\tau)\exp\left(-j\frac{\omega}{a}\tau\right)\left(\frac{1}{a}d\tau\right) = \frac{1}{a}\int_{-\infty}^{\infty} x(\tau)\exp\left(-j\frac{\omega}{a}\tau\right)d\tau = \frac{1}{a}X\left(\frac{j\omega}{a}\right)$$

□

2. **Time Shift/Modulation in Time**

$$x(t-t_0) \longleftrightarrow e^{-j\omega t_0} X(j\omega)$$

***Proof*.**

$$\mathcal{F}\{x(t-t_0)\} = \int_{-\infty}^{\infty} x(t-t_0) e^{-j\omega t}\, dt$$

Let $\tau=t-t_0,\ t=\tau+t_0,\ dt=d\tau,\ t\in(-\infty,\infty) \Longrightarrow \tau\in(-\infty,+\infty)$

$$= \int_{-\infty}^{\infty} x(\tau) e^{-j\omega(\tau+t_0)} d\tau = e^{-j\omega t_0}\int_{-\infty}^{\infty} x(\tau) e^{-j\omega\tau} d\tau = e^{-j\omega t_0} X(j\omega)$$

□

***Remark*. Compare to causal, unilateral L, F is bilateral, Heaviside Unit Step Function is not required**

3. **Expoential Shift/Modulation in Frequency**

$$e^{jat}x(t) \longleftrightarrow X(j(\omega-a))$$

***Proof*.**

$$\mathcal{F}\left\{e^{jat}x(t)\right\} = \int_{-\infty}^{\infty} e^{jat}x(t)e^{-j\omega t}\,dt = \int_{-\infty}^{\infty} x(t)e^{-j(\omega-a)t}\,dt = X(j(\omega-a))$$

□

4. **TimeReverse**

$$x(-t) \longleftrightarrow X(-j\omega)$$

Proof.

$$F\{x(-t)\} = \int_{-\infty}^{\infty} x(-t)e^{-j\omega t}dt$$

Let $-t=\tau, dt=-d\tau, t=-\infty \Rightarrow \tau=\infty, t=\infty \Rightarrow \tau=-\infty$

$$= \int_{\infty}^{-\infty} x(\tau)e^{j\omega\tau}(-1)d\tau = \int_{-\infty}^{\infty} x(\tau)e^{-j(-\omega)\tau}d\tau = X(-j\omega)$$

DIFFERENTIAL PROPERTY

Time differentitation

Time intergral

Frequency differentiation

Frequency intergration

Time-Differentiation

$$\frac{dx(t)}{dt} \longleftrightarrow j\omega X(j\omega)$$

Proof.

$$F\left\{\frac{dx(t)}{dt}\right\} = \int_{-\infty}^{\infty}\frac{dx(t)}{dt}e^{-j\omega t}dt = \int_{-\infty}^{\infty} e^{-j\omega t}dx(t) = \left[e^{-j\omega t}x(t)\right]_{-\infty}^{\infty} - \int_{-\infty}^{\infty} x(t)\,de^{-j\omega t} = j\omega X(j\omega)$$

$0 \quad x(\pm\infty)=0$

***Remark*1.** $\lim_{t\to\pm\infty} x(t)=0$

***Remark*2. Repeat the process**

$$F\left\{\frac{d}{dt}\left[\frac{dx(t)}{dt}\right]\right\} = j\omega\, F\left[\left\{\frac{dx(t)}{dt}\right\}\right] = (j\omega)^2X(j\omega)$$

$$F\left\{\frac{d^nx(t)}{dt^n}\right\} = (j\omega)^n X(j\omega)$$

5. **Time Integral**

$$\int_{-\infty}^{t} f(\tau)d\tau \longleftrightarrow \frac{1}{j\omega}F(j\omega)$$

Proof. **Consider** $g(t)=\int_{-\infty}^{t} f(\tau)d\tau$, $\lim_{t\to\pm\infty} g(t)=0$

$$\frac{dg(t)}{dt} = f(t)$$

$$F\left\{\frac{dg(t)}{dt}\right\} \quad \frac{d}{d}$$

$$\mathcal{F}\left\{\frac{d}{dt}\int_{-\infty}^{t} f(\tau)\,d\tau\right\} = j\omega G(j\omega) \longleftrightarrow \mathcal{F}\{f(t)\} = F(j\omega)$$

$$G(j\omega) = \mathcal{F}\{g(t)\} = \mathcal{F}\left\{\int_{-\infty}^{t} f(\tau)\,d\tau\right\}$$

$$j\omega G(j\omega) = F(j\omega)$$

$$\mathcal{F}\left\{\int_{-\infty}^{t} f(\tau)\,d\tau\right\} = \frac{1}{j\omega}F(j\omega)$$

□

6. **Frequency Differentiation/t-multiplication**

$$t^n x(t) \longleftrightarrow j^n \frac{d^n}{d\omega^n} X(j\omega)$$

Proof. Consider RHS for $n=1$

$$\frac{dX(j\omega)}{d\omega} = \frac{d}{d\omega}\int_{-\infty}^{\infty} x(t)e^{-j\omega t}dt = \int_{-\infty}^{\infty} x(t)\frac{de^{-j\omega t}}{d\omega}dt = \int_{-\infty}^{\infty} -jtx(t)e^{-j\omega t}dt = j\int_{-\infty}^{\infty} tx(t)e^{-j\omega t}dt$$

$$\therefore j\frac{dX(j\omega)}{d\omega} = \int_{-\infty}^{\infty} tx(t)e^{-j\omega t}dt$$

Repeat the process, or using *Mathematical Induction*,

$$\mathcal{F}\{t^n x(t)\} = j^n \frac{d^n}{d\omega^n}X(j\omega)$$

□

7. **Frequency Integration/t-division**

$$\frac{x(t)}{t} \longleftrightarrow \int_{j\omega}^{\infty} X(j\mu)dj\mu$$

Proof. Consider RHS:

$$\int_{j\omega}^{\infty}[X(j\mu)]dj\mu = \int_{j\omega}^{\infty}\left[\int_{-\infty}^{\infty} x(t)e^{-j\mu t}dt\right]dj\mu = \int_{-\infty}^{\infty} x(t)\left[\int_{j\omega}^{\infty} e^{-j\mu t}dj\mu\right]dt$$

$$= \int_{-\infty}^{\infty} x(t)\left[\frac{e^{-j\mu t}}{-t}\right]_{j\omega}^{\infty}dt = \int_{-\infty}^{\infty}\frac{x(t)}{t}e^{-j\omega t}dt = \mathcal{F}\left\{\frac{x(t)}{t}\right\}$$

SAMPLINGTHEOREM

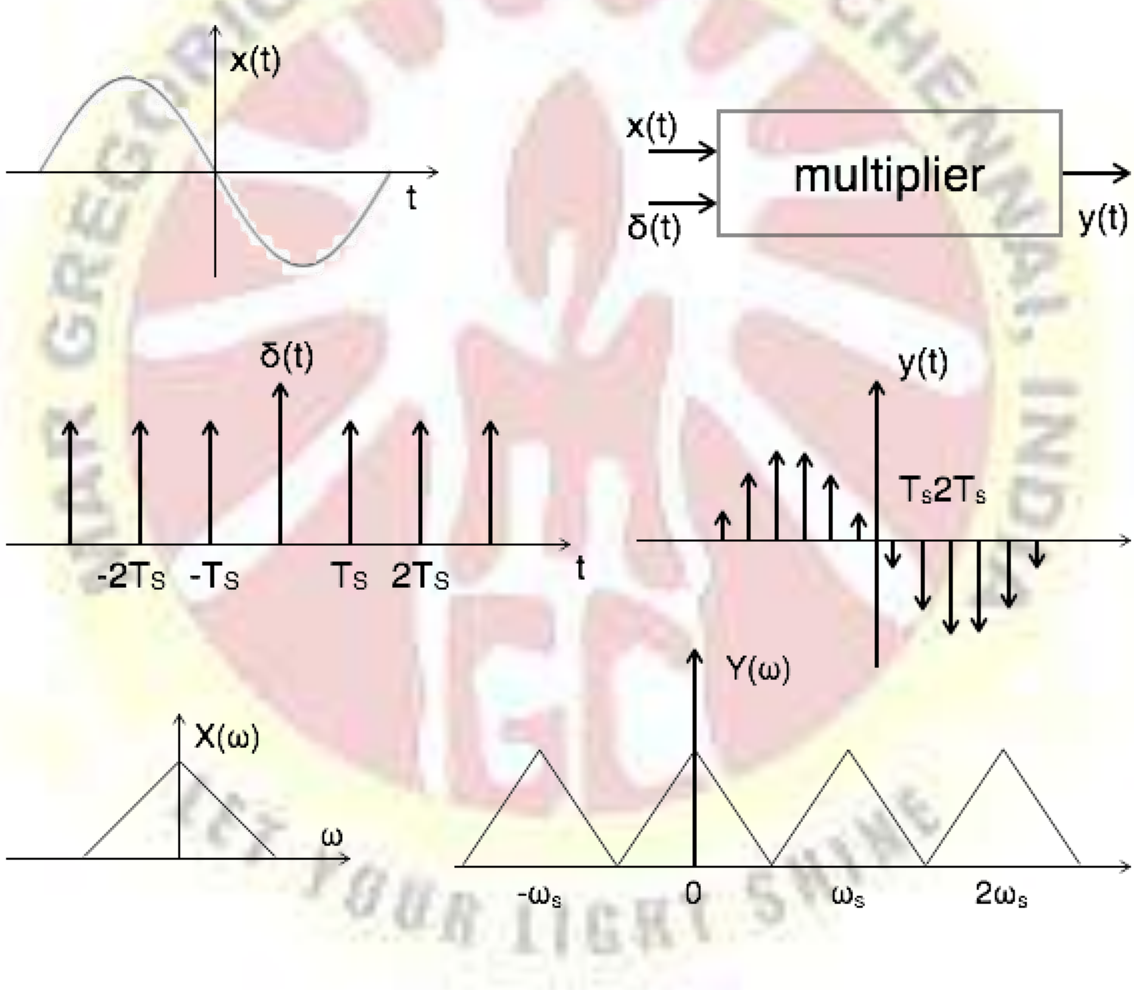
Statement: Acontinuoustimesignalcanberepresentedinitssamplesandcanberecoveredbackwhensamplingfrequency$f_s$isgreaterthanorequaltothetwicethehighestfrequencycomponentofmessagesignal. i.e.

$f_s \leq 2f_m$.

Proof: Consideracontinuoustime signalx$t$.The spectrumofx$t$isabandlimitedtof_mHzi.e.thespectrumofx$t$iszerofor$|\omega| > \omega_m$.

Samplingofinputsignalx$t$canbeobtainedbymultiplyingx$t$withanimpulsetrainδtofperiodT_s.The outputofmultiplierisadiscretesignalcalledsampledsignalwhichisrepresentedwithy$t$inthefollowingdiagrams:

Here,youcanobservethatthesample


x(t)
t
x(t)
δ(t)
multiplier
y(t)
δ(t)
-2TS
-TS
TS
2TS
t
y(t)
Ts2Ts
Y(ω)
X(ω)
ω
-ωs
0
ωs
2ωs

–

–

dsignaltakestheperiodofimpulse.Theprocessofsamplingcanbeexplainedbythefollowingmathematicalexpression:

Sampledsignal$y(t)=x(t).\delta(t)$......................................(1)

TrigonometricFourierseriesrepresentationofδtisgivenby

$\delta(t)=a_0+\sum_{n=1}^{\infty}(a_n\cos n\omega_s t+b_n\sin n\omega_s t)$....................................(2)

$$\text{Where}\ a_0=\frac{1}{T_s}\int_{-\frac{T}{2}}^{\frac{T}{2}}\delta(t)dt=\frac{1}{T_s}\delta(0)=\frac{1}{T_s}$$

$$a_n=\frac{2}{T_s}\int_{-\frac{T}{2}}^{\frac{T}{2}}\delta(t)\cos n\omega_s dt=\frac{2}{T_2}\delta(0)\cos n\omega 0=\frac{2}{T_s}$$

$$b_n = \frac{2}{T_s} \int_{-\frac{T}{2}}^{\frac{T}{2}} \delta(t) \sin n\omega_s t\, dt = \frac{2}{T_s} \delta(0) \sin n\omega 0 = 0$$

Substituteabovevaluesinequation2.

$$\therefore \delta(t) = \frac{1}{T_s} + \sum_{n=1}^{\infty} \left(\frac{2}{T_s} \cos n\omega_s t + 0\right)$$

Substitute δtinequation1.

$\rightarrow y(t) = x(t).\delta(t)$

$$= x(t)\left[\frac{1}{T_s} + \sum_{n=1}^{\infty} \left(\frac{2}{T_s} \cos n\omega_s t\right)\right]$$

$$= \frac{1}{T_S}\left[x(t) + 2\sum_{n=1}^{\infty} (\cos n\omega_s t) x(t)\right]$$

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

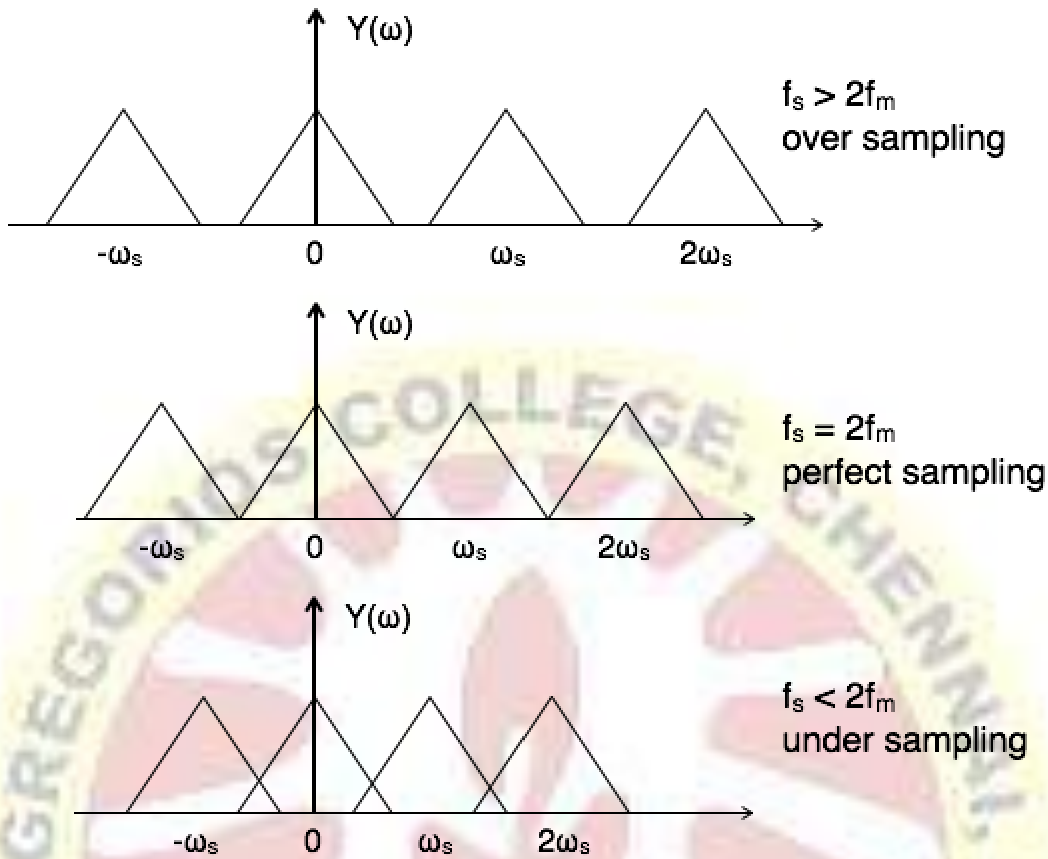
TakeFouriertransformonbothsides.

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

$$\therefore Y(\omega) = \frac{1}{T_s} \sum_{n=-\infty}^{\infty} X(\omega - n\omega_s) \quad where\, n = 0, \pm 1, \pm 2, \ldots$$

Toreconstructx$t$,youmustrecoverinputsignalspectrumXωfromsampledsignalspectrumYω,whichispossiblewhenthereisnooverlappingbetweenthecyclesofYω.

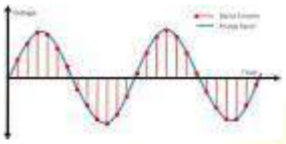
Possibilityofsampledfrequencyspectrumwithdifferentconditionsisgivenbythefollowingdiagrams:


Y(ω)
$f_s > 2f_m$
over sampling
-ωs
0
ωs
2ωs
Y(ω)
$f_s = 2f_m$
perfect sampling
-ωs
0
ωs
2ωs
Y(ω)
$f_s < 2f_m$
under sampling
-ωs
0
ωs
2ωs

Sampling is the process of converting analog signal into a discrete signal or making an analog or continuous signal to occur at a particular interval of time, this phenomena is known as sampling.

SAMPLING THEOREM:-

Sampling theorem states that a band limited signal having no frequency components higher than fm hertz can be sampled if its sampling freq is equal to or greater than Nyquist rate.



Analog Signal Representation

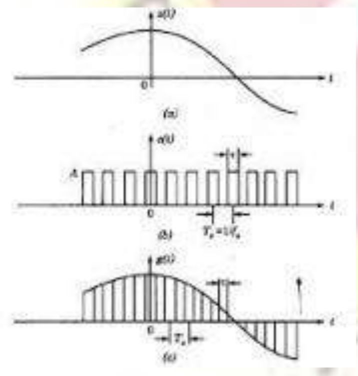
1.4 Sampling Techniques

Their are basically three types of Sampling techniques, namely:

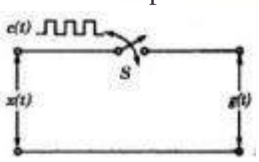
1. Natural Sampling
2. Flat top Sampling
3. Ideal Sampling

1. Natural Sampling:

Natural Sampling is a practical method of sampling in which pulse have finite width equal to τ. Sampling is done in accordance with the carrier signal which is digital in nature.



Natural Sampled Waveform



Functional Diagram of Natural Sampler

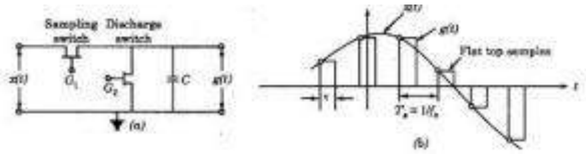
With the help of functional diagram of a Natural sampler, a sampled signal g(t) is obtained by multiplication of sampling function c(t) and the input signal x(t).

Spectrum of Natural Sampled Signal is given by:

$$\mathbf{G(f) = A\tau / T_s .[\Sigma \sin c(n f_s.\tau) X(f-n f_s)]}$$

2. <u>Flat Top Sampling:</u>

Flat top sampling is like natural sampling i.e; practical in nature. In comparison to natural sampling flat top sampling can be easily obtained. In this sampling techniques, the top of the samples remains constant and is equal to the instantaneous value of the message signal x(t) at the start of sampling process. Sample and hold circuit are used in this type of sampling.

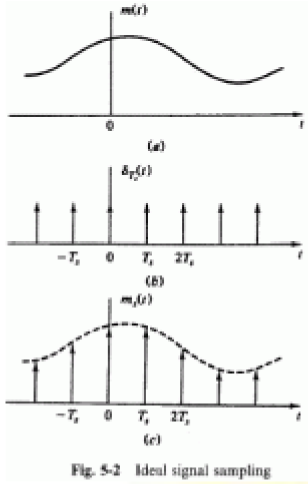


Block Diagram and Waveform

- **Figure(a),** shows functional diagram of a sample hold circuit which is used to generate fat top samples.
- **Figure(b),** shows the general waveform of the flat top samples. It can be observed that only starting edge of the pulse represent the instantaneous value of the message signal x(t).

Spectrum of Flat top Sampled Signal is given by: $G(f) = f_s .[\Sigma X(f\text{-}n f_s). H(f)]$

3. <u>Ideal Sampling:</u>



Ideal Sampling Wave form

Ideal Sampling is also known as Instantaneous sampling or Impulse Sampling.Train of impulse is used as a carrier signal for ideal sampling. In this sampling technique the sampling function is a train of impulses and the principle used is known as multiplication principle.

Here,
Figure (a), represent message signal or input signal or signal to be sampled.
Figure (b), represent the sampling function.
Figure (c), represent the resultant signal.
Spectrum of IdealSampled Signal is given by: $G(f) = f_s .[\Sigma X(f\text{-}n f_s)]$

<u>NYQUIST RATE:</u>

Nyquist rate is the rate at which sampling of a signal is done so that overlapping of frequency does not take place. When the sampling rate become exactly equal to $2f_m$ samples per second, then the specific rate is known as Nyquist rate. It is also knowaas the minimum sampling rate and given by: $\mathbf{f_s = 2f_m}$

Effect of Under sampling: ALIASING

It is the effect in which overlapping of a frequency components takes place at the frequency higher than Nyquist rate. Signal loss may occur due to aliasing effect. We can say that aliasing is the phenomena in which a high frequency component in the frequency spectrum of a signal takes identity of a lower frequency component in the same spectrum of the sampled signal.

Because of overlapping due to process of aliasing, sometimes it is not possible to overcome the sampled signal x(t) from the sampled signal g(t) by applying the process of low pass filtering since the spectral components in the overlap regions . hence this causes the signal to destroy.

The Effect of Aliasing can be reduced:

1) Pre alias filter must be used to limit band of frequency of the required signal f_m Hz.
2) Sampling frequency f_s must be selected such that $f_s > 2f_m$.

2.0 AMPLITUDE MODULATION

PREREQUISTING ABOUT MODULATION:

In this chapter we discussed about Modulation is the process of varying one or more properties of a periodic waveform, called the carrier signal (high frequency signal), with a modulating signal that typically contains information to be transmitted.

✓ **Need for modulation:**

- Antenna Height
- Narrow Banding
- Poor radiation and penetration
- Diffraction angle Multiplexing.

✓ **Functions of the Carrier Wave:**

The main function of the carrier wave is to carry the audio or video signal from the transmitter to the receiver. The wave that is resulted due to superimposition of audio signal and carrier wave is called the modulated wave.

✓ **Types of modulation:**

The sinusoidal carrier wave can be given by the equation, $v_c = V_c Sin(w_c t + \theta) = V_c\ Sin(2f_c t + \theta)$ V_c – Maximum Value f_c – Frequency θ – Phase Relation

Since the three variables are the amplitude, frequency, and phase angle, the modulation can be done by varying any one of them. Thus there are three modulation types namely:

- **Amplitude Modulation (AM)**
- **Frequency Modulation (FM)**
- **Phase Modulation (PM)**

- We have introduced linear modulation. In particular,
  - ✓ DSB-SC, Double sideband suppressed carrier
  - ✓ DSB-LC, Double sideband large carrier (AM)
  - ✓ SSB, Single sideband
  - ✓ VSB, Vestigial sideband

CONTENT:

- **AMPLITUDE MODULATION**
- **AM TRANSMITTER**
- **SSB - SC**
- **VSB - SC**
- **DSB - SC**
- **HILBERT TRANSFORM**
- **SUPER HETERODYNE RECEIVER**
- **COMPARISION OF VARIOUS AM TECHNIQUES**

2.1 AMPLITUDE MODULATION:

"**Modulation is the process of superimposing a low frequency signal on a high frequency carrier signal.**"

OR

"**The process of modulation can be defined as varying the RF carrier wave in accordance with the intelligence or information in a low frequency signal**."

OR

"**Modulation is defined as the precess by which some characteristics, usually amplitude, frequency or phase, of a carrier is varied in accordance with instantaneous value of some other voltage, called the modulating voltage.**"

Need For Modulation

1. If two musical programs were played at the same time within distance, it would be difficult for anyone to listen to one source and not hear the second source. Since all musical sounds have approximately the same frequency range, form about 50 Hz to 10KHz. If a desired program is shifted up to a band of frequencies between 100KHz and 110KHz, and the second program shifted up to the band between 120KHz and 130KHz, Then both programs gave still 10KHz bandwidth and the listener can (by band selection) retrieve the program of his own choice. The

receiver would down shift only the selected band of frequencies to a suitable range of 50Hz to 10KHz.

2. A second more technical reason to shift the message signal to a higher frequency is related to antenna size. It is to be noted that the antenna size is inversely proportional to the frequency to be radiated. This is 75 meters at 1 MHz but at 15KHz it has increased to 5000 meters (or just over 16,000 feet) a vertical antenna of this size is impossible.
3. The third reason for modulating a high frequency carrier is that RF (radio frequency) energy will travel a great distance than the same amount of energy transmitted as sound power.

Types of Modulation

The carrier signal is a sine wave at the carrier frequency. Below equation shows that the sine wave has three characteristics that can be altered.

Instantaneous voltage (E) $= E_{c(max)} Sin(2\pi f_c t + \theta)$

The term that may be varied are the carrier voltage Ec, the carrier frequency fc, and the carrier phase angle θ. So three forms of modulations are possible.

1. **AmplitudeModulation**

   Amplitude modulation is an increase or decrease of the carrier voltage (Ec), will all other factors remaining constant.

2. **FrequencyModulation**

   Frequency modulation is a change in the carrier frequency (fc) with all other factors remaining constant.

3. **PhaseModulation**

   Phase modulation is a change in the carrier phase angle (θ). The phase angle cannot change without also affecting a change in frequency. Therefore, phase modulation is in reality a second form of frequency modulation.

✓ **EXPLAINATION OF AM:**

The method of varying amplitude of a high frequency carrier wave in accordance with the information to be transmitted, keeping the frequency and phase of the carrier wave unchanged is called Amplitude Modulation. The information is considered as the modulating signal and it is superimposed on the carrier wave by applying both of them to the modulator. The detailed diagram showing the amplitude modulation process is given below.

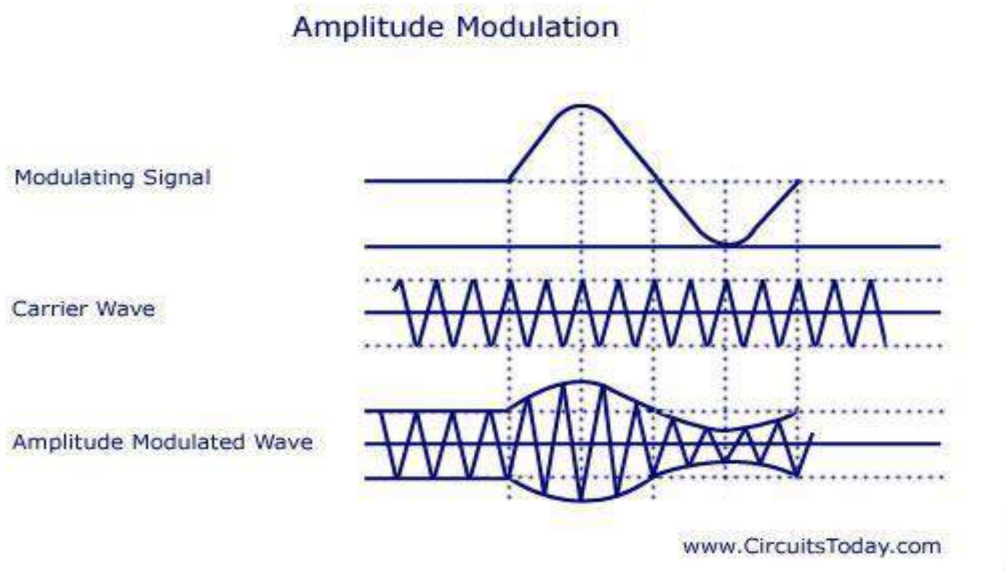


FIG2.1 Amplitude Modulation

As shown above, the carrier wave has positive and negative half cycles. Both these cycles are varied according to the information to be sent. The carrier then consists of sine waves whose amplitudes follow the amplitude variations of the modulating wave. The carrier is kept in an envelope formed by the modulating wave. From the figure, you can also see that the amplitude variation of the high frequency carrier is at the signal frequency and the frequency of the carrier wave is the same as the frequency of the resulting wave.

- ✓ **Analysis of Amplitude Modulation Carrier Wave:** Let $v_c = V_c \text{ Sin } w_c t v_m = V_m \text{ Sin } w_m t$

v_c – Instantaneous value of the carrier V_c – Peak value of the carrier W_c – Angular velocity of the carrier v_m – Instantaneous value of the modulating signal V_m – Maximum value of the modulating signal w_m – Angular velocity of the modulating signal f_m – Modulating signal frequency

It must be noted that the phase angle remains constant in this process. Thus it can be ignored. The amplitude of the carrier wave varies at f_m.The amplitude modulated wave is given by the equation $A = V_c + v_m = V_c + V_m \text{ Sin } w_m t = V_c [1+ (V_m/V_c \text{ Sin } w_m t)]$

$= V_c (1 + m\text{Sin}w_m t)$

m – Modulation Index. The ratio of V_m/V_c.

Instantaneous value of amplitude modulated wave is given by the equation $v = A \text{ Sin } w_c t = Vc (1 + m \text{ Sin } w_m t) \text{ Sin } wct = V_c \text{ Sin } w_c t + mVc (\text{Sin } w_m t \text{ Sin } w_c t)$

$v = V_c \text{ Sin } wct + [mV_c/2 \text{ Cos } (wc-wm)t – mVc/2 \text{ Cos } (wc + wm)t]$

The above equation represents the sum of three sine waves. One with amplitude of Vc and a frequency of $w_c/2$, the second one with an amplitude of $mV_c/2$ and frequency of

$(w_c – w_m)/2$ and the third one with an amplitude of $mV_c/2$ and a frequency of $(w_c + w_m)/2$.

In practice the angular velocity of the carrier is known to be greater than the angular velocity of the modulating signal ($w_c>>w_m$). Thus, the second and third cosine equations are more close to the carrier frequency. The equation is represented graphically as shown below.

Frequency Spectrum of AM Wave:

Lower side frequency – $(w_c – w_m)/2$

Upper side frequency – $(w_c +w_m)/2$

The frequency components present in the AM wave are represented by vertical lines approximately located along the frequency axis. The height of each vertical line is drawn in proportion to its amplitude. Since the angular velocity of the carrier is greater than the angular velocity of the modulating signal, the amplitude of side band frequencies can never exceed half of the carrier amplitude.

Thus there will not be any change in the original frequency, but the side band frequencies $(w_c – w_m)/2$ and $(w_c +w_m)/2$ will be changed. The former is called the upper side band (USB) frequency and the later is known as lower side band (LSB) frequency.

Since the signal frequency $w_m/2$ is present in the side bands, it is clear that the carrier voltage component does not transmit any information.

Two side banded frequencies will be produced when a carrier is amplitude modulated by a single frequency. That is, an AM wave has a band width from $(w_c – w_m)/2$ to $(w_c +w_m)/2$, that is, $2w_m/2$ or twice the signal frequency is produced. When a modulating signal has more than one frequency, two side band frequencies are produced by every frequency. Similarly for two frequencies of the modulating signal 2 LSB's and 2 USB's frequencies will be produced.

The side bands of frequencies present above the carrier frequency will be same as the ones present below. The side band frequencies present above the carrier frequency is known to be the upper side band and all those below the carrier frequency belong to the lower side band. The USB frequencies represent the some of the individual modulating frequencies and the LSB frequencies represent the difference between the modulating frequency and the carrier frequency. The total bandwidth is represented in terms of the higher modulating frequency and is equal to twice this frequency.

Modulation Index (m):

The ratio between the amplitude change of carrier wave to the amplitude of the normal carrier wave is called modulation index. It is represented by the letter _m'.

It can also be defined as the range in which the amplitude of the carrier wave is varied by the modulating signal. $m = V_m/V_c$.

Percentage modulation, $\%m = m*100 = V_m/V_c *$

100 The percentage modulation lies between 0 and 80%.

Another way of expressing the modulation index is in terms of the maximum and minimum values of the amplitude of the modulated carrier wave. This is shown in the figure below.

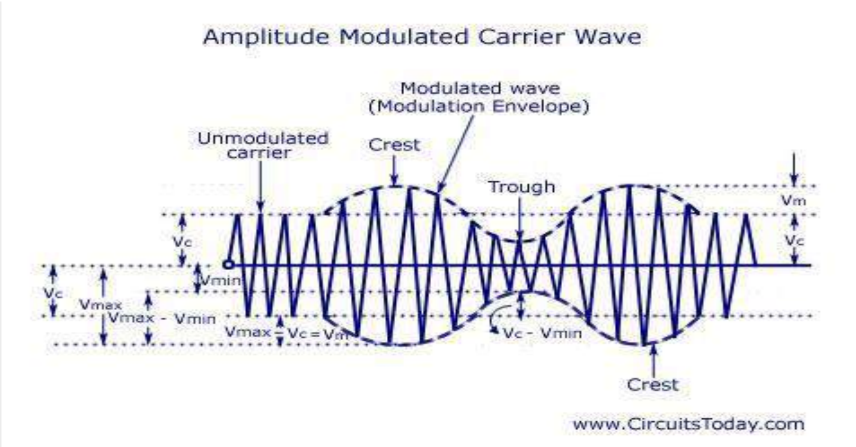


FIG 2.2 Amplitude Modulation Carrier Wave

$$2\ V_{in} = V_{max} - V_{min}$$

Vin = (Vmax – Vmin)/2

Vc = Vmax – Vin

= Vmax – (Vmax-Vmin)/2

=(Vmax + Vmin)/2

Substituting the values of Vm and Vc in the equation m = Vm/Vc , we get

M = Vmax – Vmin/Vmax + Vmin

As told earlier, the value of ‗m' lies between 0 and 0.8. The value of m determines the strength and the quality of the transmitted signal. In an AM wave, the signal is contained in the variations of the carrier amplitude. The audio signal transmitted will be weak if the carrier wave is only modulated to a very small degree. But if the value of m exceeds unity, the transmitter output produces erroneous distortion.

Power Relations in an AM wave:

A modulated wave has more power than had by the carrier wave before modulating. The total power components in amplitude modulation can be written as:

Ptotal = Pcarrier + PLSB + PUSB Considering additional resistance like antenna resistance R.

Pcarrier = [(Vc/√2)/R]2 = V2C/2R

Each side band has a value of $m/2\ V_c$ and r.m.s value of $mV_c/2\sqrt{2}$. Hence power in LSB and USB can be written as

$P_{LSB} = P_{USB} = (mV_c/2\sqrt{2})^2/R = m^2/4*V^2C/2R = m_2/4\ P_{carrier}$

$P_{total} = V^2{}_C/2R + [m^2/4*V^2C/2R] + [m^2/4*V^2C/2R] = V^2{}_C/2R\ (1 + m^2/2) = P_{carrier}(1 + m^2/2)$

In some applications, the carrier is simultaneously modulated by several sinusoidal modulating signals. In such a case, the total modulation index is given as

$Mt = \sqrt{(m1^2 + m2^2 + m3^2 + m4^2 + \ldots}$..

If Ic and It are the r.m.s values of unmodulated current and total modulated current and R is the resistance through which these current flow, then

Ptotal/Pcarrier = (It.R/Ic.R)2 = (It/Ic)2

Ptotal/Pcarrier = (1 + m2/2)

$It/Ic = 1 + m^2/2$

Limitations of Amplitude Modulation:

1. Low Efficiency- Since the useful power that lies in the small bands is quite small, so the efficiency of AM system is low.
2. Limited Operating Range – The range of operation is small due to low efficiency. Thus, transmission of signals is difficult.
3. Noise in Reception – As the radio receiver finds it difficult to distinguish between the amplitude variations that represent noise and those with the signals, heavy noise is prone to occur in its reception.
4. Poor Audio Quality – To obtain high fidelity reception, all audio frequencies till 15 KiloHertz must be reproduced and this necessitates the bandwidth of 10 KiloHertz to minimise the interference from the adjacent broadcasting stations. Therefore in AM broadcasting stations audio quality is known to be poor.

2.2 AM TRANSMITTERS:

Transmitters that transmit AM signals are known as AM transmitters. These transmitters are used in medium wave (MW) and short wave (SW) frequency bands for AM broadcast. The MW band has frequencies between 550 KHz and 1650 KHz, and the SW band has frequencies ranging from 3 MHz to 30 MHz. The two types of AM transmitters that are used based on their transmitting powers are:

- High Level
- Low Level

High level transmitters use high level modulation, and low level transmitters use low level modulation. The choice between the two modulation schemes depends on the transmitting power of the AM transmitter. In broadcast transmitters, where the transmitting power may be of the order of kilowatts, high level modulation is employed. In low power transmitters, where only a few watts of transmitting power are required , low level modulation is used.

High-Level and Low-Level Transmitters Below figure's show the block diagram of high-level and low-level transmitters. The basic difference between the two transmitters is the power amplification of the carrier and modulating signals

Figure (a) shows the block diagram of high-level AM transmitter.

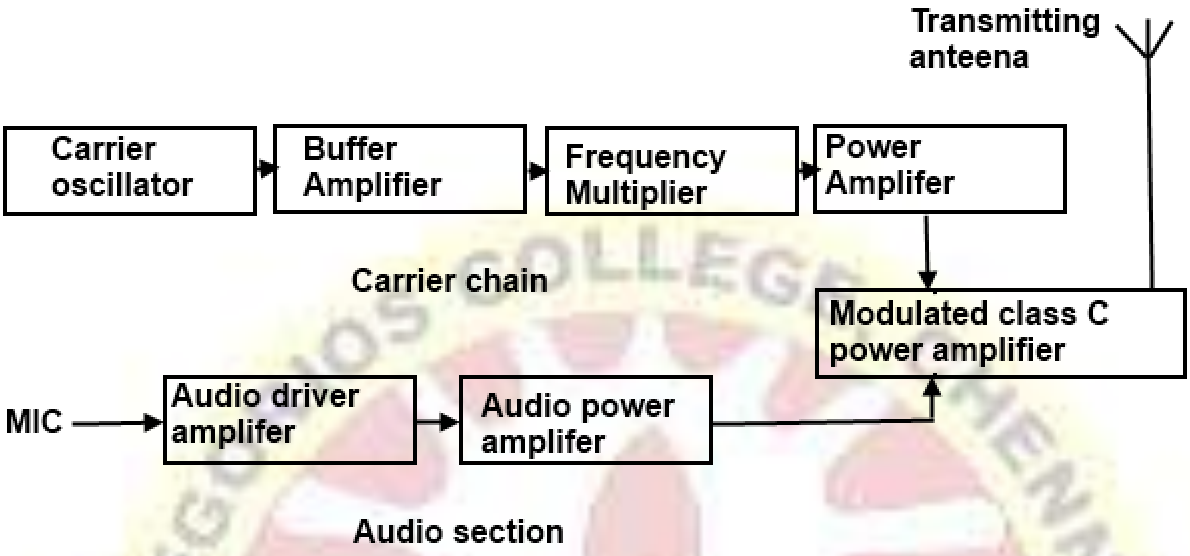


Figure (a) Block diagram of high level AM transmitter

Figure (a) is drawn for audio transmission. In high-level transmission, the powers of the carrier and modulating signals are amplified before applying them to the modulator stage, as shown in figure (a). In low-level modulation, the powers of the two input signals of the modulator stage are not amplified. The required transmitting power is obtained from the last stage of the transmitter, the class C power amplifier.

The various sections of the figure (a) are:

- Carrier oscillator
- Buffer amplifier
- Frequency multiplier
- Power amplifier
- Audio chain
- Modulated class C power amplifier

✓ **Carrier oscillator**

The carrier oscillator generates the carrier signal, which lies in the RF range. The frequency of the carrier is always very high. Because it is very difficult to generate high frequencies with good frequency stability, the carrier oscillator generates a sub multiple with the required carrier frequency. This sub multiple frequency is multiplied by the frequency multiplier stage to get the required carrier frequency. Further, a crystal oscillator can be used in this stage to generate a low frequency carrier with the best frequency stability. The frequency multiplier stage then increases the frequency of the carrier to its requirements.

- **Buffer Amplifier**

The purpose of the buffer amplifier is twofold. It first matches the output impedance of the carrier oscillator with the input impedance of the frequency multiplier, the next stage of the carrier oscillator. It then isolates the carrier oscillator and frequency multiplier.
This is required so that the multiplier does not draw a large current from the carrier oscillator. If this occurs, the frequency of the carrier oscillator will not remain stable.

- **Frequency Multiplier**

The sub-multiple frequency of the carrier signal, generated by the carrier oscillator , is now applied to the frequency multiplier through the buffer amplifier. This stage is also known as harmonic generator. The frequency multiplier generates higher harmonics of carrier oscillator frequency. The frequency multiplier is a tuned circuit that can be tuned to the requisite carrier frequency that is to be transmitted.

- **Power Amplifier**

The power of the carrier signal is then amplified in the power amplifier stage. This is the basic requirement of a high-level transmitter. A class C power amplifier gives high power current pulses of the carrier signal at its output.

- **Audio Chain**

The audio signal to be transmitted is obtained from the microphone, as shown in figure (a). The audio driver amplifier amplifies the voltage of this signal. This amplification is necessary to drive the audio power amplifier. Next, a class A or a class B power amplifier amplifies the power of the audio signal.

- **Modulated Class C Amplifier**

This is the output stage of the transmitter. The modulating audio signal and the carrier signal, after power amplification, are applied to this modulating stage. The modulation takes place at this stage. The class C amplifier also amplifies the power of the AM signal to the reacquired transmitting power. This signal is finally passed to the antenna., which radiates the signal into space of transmission.
Figure (b) shows the block diagram of a low-level AM transmitter.

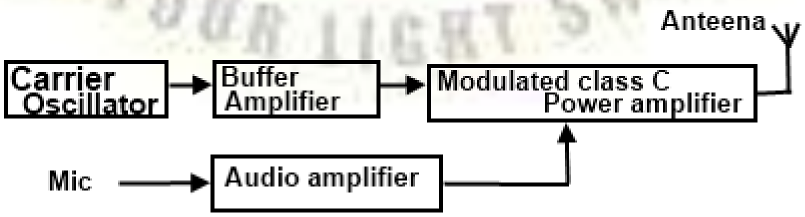


Figure (b) Block diagram of Low-level AM transmitter

The low-level AM transmitter shown in the figure (b) is similar to a high-level transmitter, except that the powers of the carrier and audio signals are not amplified. These two signals are directly applied to the modulated class C power amplifier.

Modulation takes place at the stage, and the power of the modulated signal is amplified to the required transmitting power level. The transmitting antenna then transmits the signal.

Coupling of Output Stage and Antenna

The output stage of the modulated class C power amplifier feeds the signal to the transmitting antenna. To transfer maximum power from the output stage to the antenna it is necessary that the impedance of the two sections match. For this , a matching network is required. The matching between the two should be perfect at all transmitting frequencies. As the matching is required at different frequencies, inductors and capacitors offering different impedance at different frequencies are used in the matching networks. The matching network must be constructed using these passive components. This is shown in figure ©

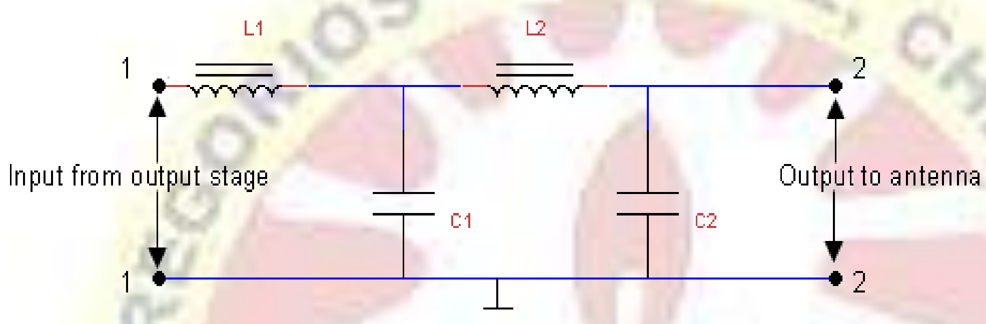


Figure (c) Double Pi Matching network

The matching network used for coupling the output stage of the transmitter and the antenna is called double π-network. This network is shown in figure (c). It consists of two inductors , L_1 and L_2 and two capacitors, C_1 and C_2. The values of these components are chosen such that the input impedance of the network between 1 and 1'. Shown in figure (c) is matched with the output impedance of the output stage of the transmitter. Further, the output impedance of the network is matched with the impedance of the antenna.

The double π matching network also filters unwanted frequency components appearing at the output of the last stage of the transmitter. The output of the modulated class C power amplifier may contain higher harmonics, such as second and third harmonics, that are highly undesirable. The frequency response of the matching network is set such that these unwanted higher harmonics are totally suppressed, and only the desired signal is coupled to the antenna.

✓ **Comparision of Am and Fm Signals**

Both AM and FM system are used in commercial and non-commercial applications. Such as radio broadcasting and television transmission. Each system has its own merits and demerits. In a Particular application, an AM system can be more suitable than an FM system. Thus the two are equally important from the application point of view.

✓ **Advantage of FM systems over AM Systems** The advantages of FM over AM systems are:

- The amplitude of an FM wave remains constant. This provides the system designers an opportunity to remove the noise from the received signal. This is done in FM receivers by employing an amplitude limiter circuit so that the noise above the limiting amplitude is suppressed. Thus, the FM system is considered a noise immune system. This is not possible in AM systems because the baseband signal is carried by the amplitude variations it self and the envelope of the AM signal cannot be altered.

- Most of the power in an FM signal is carried by the side bands. For higher values of the modulation index, mc, the major portion of the total power is contained is side bands, and the carrier signal contains less power. In contrast, in an AM system, only one third of the total power is carried by the side bands and two thirds of the total power is lost in the form of carrier power.
- In FM systems, the power of the transmitted signal depends on the amplitude of the unmodulated carrier signal, and hence it is constant. In contrast, in AM systems, the power depends on the modulation index ma. The maximum allowable power in AM systems is 100 percent when ma is unity. Such restriction is not applicable int case of FM systems. This is because the total power in an FM system is independent of the modulation index, mf and frequency deviation fd. Therefore, the power usage is optimum in an FM system.
- In an AM system, the only method of reducing noise is to increase the transmitted power of the signal. This operation increases the cost of the AM system. In an FM system, you can increase the frequency deviation in the carrier signal to reduce the noise. if the frequency deviation is high, then the corresponding variation in amplitude of the baseband signal can be easily retrieved. if the frequency deviation is small, noise 'can overshadow this variation and the frequency deviation cannot be translated into its corresponding amplitude variation. Thus, by increasing frequency deviations in the FM signal, the noise effect can he reduced. There is no provision in AM system to reduce the noise effect by any method, other than increasing itss transmitted power.
- In an FM signal, the adjacent FM channels are separated by guard bands. In an FM system there is no signal transmission through the spectrum space or the guard band. Therefore, there is hardly any interference of adjacent FM channels. However, in an AM system, there is no guard band provided between the two adjacent channels. Therefore, there is always interference of AM radio stations unless the received signalis strong enough to suppress the signal of the adjacent channel.

The disadvantages of FM systems over AM systems

- There are an infinite number of side bands in an FM signal and therefore the theoretical bandwidth of an FM system is infinite. The bandwidth of an FM system is limited by Carson's rule, but is still much higher, especially in WBFM. In AM systems, the bandwidth is only twice the modulation frequency, which is much less than that of WBFN. This makes FM systems costlier than AM systems.
- The equipment of FM system is more complex than AM systems because of the complex circuitry of FM systems; this is another reason that FM systems are costlier AM systems.
- The receiving area of an FM system is smaller than an AM system consequently FM channels are restricted to metropolitan areas while AM radio stations can be

received anywhere in the world. An FM system transmits signals through line of sight propagation, in which the distance between the transmitting and receiving antenna should not be much. in an AM system signals of short wave band stations are transmitted through atmospheric layers that reflect the radio waves over a wider area.

2.3 SSB TRANSMISSION:

There are two methods used for SSB Transmission.

1. Filter Method
2. Phase Shift Method 3.Block diagram of SSB

Filter Method:

This is the filter method of SSB suppression for the transmission. Fig 2.3

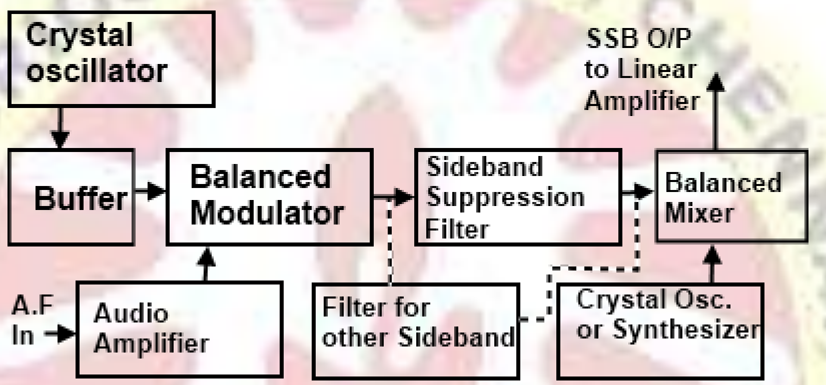


FIG 2.3 Filter Method

1. A crystal controlled master oscillator produces a stable carrier frequency fc (say 100 KHz)
2. This carrier frequency is then fed to the balanced modulator through a buffer amplifier which isolates these two satges.
3. The audio signal from the modulating amplifier modulates the carrier in the balanced modulator. Audio frequency range is 300 to 2800 Hz. The carrier is also suppressed in this stage but allows only to pass the both side bands. (USB & LSB).
4. A band pass filter (BPF) allows only a single band either USB or LSB to pass through it. It depends on our requirements.
5. This side band is then heterodyned in the balanced mixer stage with 12 MHz frequency produced by crystal oscillator or synthesizer depends upon the requirements of our transmission. So in mixer stage, the frequency of the crystal oscillator or synthersizer is added to SSB signal. The output frequency thus being raised to the value desired for transmission.
6. Then this band is amplified in driver and power amplifier stages and then fed to the aerial for the transmission.

✓ **Phase Shift Method:**

The phaseing method of SSB generation uses a phase shift technique that causes one of the side bands to be conceled out. A block diagram of a phasing type SSB generator is shown in fig2.4.

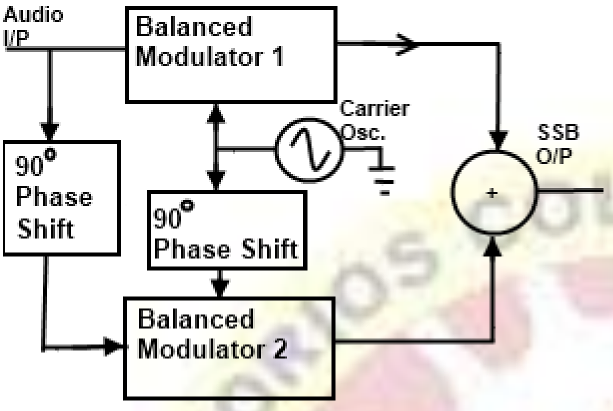


FIG 2.4 Phase Shift Method

It uses two balanced modulators instead of one. The balanced modulators effectively eliminate the carrier. The carrier oscillator is applied directly to the upper balanced modulator along with the audio modulating signal. Then both the carrier and modulating signal are shifted in phase by 90° and applied to the second, lower, balanced modulator. The two balanced modulator output are then added together algebraically. The phase shifting action causes one side band to be canceled out when the two balanced modulator outputs are combined.

✓ **Block diagram of SSB:**

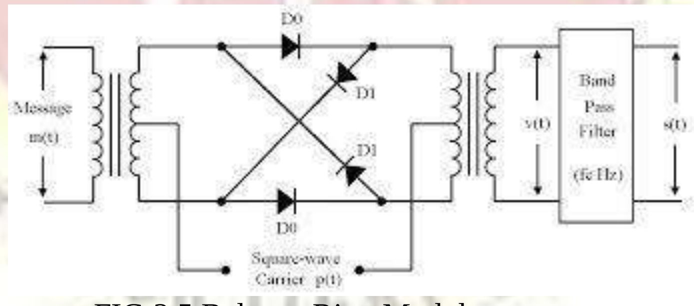


FIG 2.5 Balance Ring Modulator

Operation of Balance Ring Modulator:

✓ Ring modulation is a signal-processing function in electronics, an implementation of amplitude modulationorfrequency mixing,performed by multiplying two signals, where one is typically a sine-waveor another simple waveform. It is referred to as "ring" modulation because the analog circuitofdiodesoriginally used to implement this technique took the shape of a ring. This circuit is similar to a bridge rectifier, except that instead of the diodes facing "left" or "right", they go "clockwise" or "anti-clockwise". A ring modulator is an effects unitworking on this principle.

- ✓ The carrier, which is AC, at a given time, makes one pair of diodes conduct, and reversebiases the other pair. The conducting pair carries the signal from the left transformersecondary to the primary of the transformer at the right. If the left carrier terminal is positive, the top and bottom diodes conduct. If that terminal is negative, then the "side" diodes conduct, but create a polarity inversion between the transformers. This action is much like that of a DPDT switch wired for reversing connections.
- ✓ Ring modulators frequency mixorheterodynetwowaveforms, and output the sum and differenceof the frequencies present in each waveform. This process of ring modulation produces a signal rich in partials. As well, neither the carrier nor the incoming signal is prominent in the outputs, and ideally, not at all.
- ✓ Two oscillators, whose frequencies were harmonically related and ring modulated against each other, produce sounds that still adhere to the harmonic partials of the notes, but contain a very different spectral make up. When the oscillators' frequencies are not harmonically related, ring modulation creates inharmonic, often producing bell-like or otherwise metallic sounds.
- ✓ If the same signal is sent to both inputs of a ring modulator, the resultant harmonic spectrum is the original frequency domain doubled (if $f_1 = f_2 = f$, then $f_2 - f_1 = 0$ and $f_2 + f_1 = 2f$). Regarded as multiplication, this operation amounts to squaring. However, some distortion occurs due to the forward voltage drop of the diodes.
- ✓ Some modern ring modulators are implemented using digital signal processingtechniques by simply multiplying the time domainsignals, producing a nearly-perfect signal output. Before digital music synthesizers became common, at least some analog synthesizers (such as the ARP 2600) used analog multipliers for this purpose; they were closely related to those used in electronic analog computers. (The "ring modulator" in the ARP 2600 could multiply control voltages; it could work at DC.)
- ✓ Multiplication in the time domainis the same as convolutionin the frequency domain, so the output waveform contains the sum and difference of the input frequencies. Thus, in the basic case where two sine waves of frequencies f_1 and f_2 ($f_1 < f_2$) are multiplied, two new sine waves are created, with one at $f_1 + f_2$ and the other at $f_2 - f_1$. The two new waves are unlikely to be harmonically related and (in a well designed ring modulator) the original signals are not present. It is this that gives the ring modulator its unique tones.
- ✓ Intermodulationproducts can be generated by carefully selecting and changing the frequencyof the two input waveforms. If the signals are processed digitally, the frequencydomain convolution becomes circular convolution. If the signals are wideband, this will cause aliasingdistortion, so it is common to oversamplethe operation or low-pass filter the signals prior to ring modulation.
- ✓ One application is spectral inversion, typically of speech; a carrier frequency is chosen to be above the highest speech frequencies (which are low-pass filtered at, say, 3 kHz, for a carrier of perhaps 3.3 kHz), and the sum frequencies from the

modulator are removed by more low-pass filtering. The remaining difference frequencies have an inverted spectrum High frequencies become low, and vice versa.

- **Advantages:**
- It allows better management of the frequency spectrum. More transmission can fit into a given frequency range than would be possible with double side band DSB signals. All of the transmitted power is message power none is dissipate as carrier power.
- **Disadvantages:**

1. The cost of a single side band SSB receiver is higher than the double side band DSB counterpart be a ratio of about 3:1.
2. The average radio user wants only to flip a power switch and dial a station. Single side band SSB receivers require several precise frequency control settings to minimize distortion and may require continual readjustment during the use of the system.

2.4 VESTIGIAL SIDE BAND (VSB) MODULATION:

• The following are the drawbacks of SSB signal generation:

1. Generation of an SSB signal is difficult.

2. Selective filtering is to be done to get the original signal back.

3. Phase shifter should be exactly tuned to 90_o.

• To overcome these drawbacks, VSB modulation is used. It can view as a compromise between SSB and DSB-SC. Figure3.5 shows all the three modulation schemes.

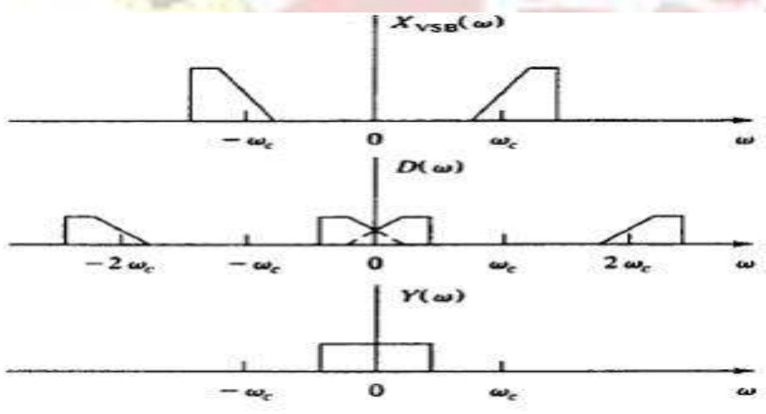


Fig. 3-13 Synchronous demodulation of VSB signals

Spectrum of VSB Signals:

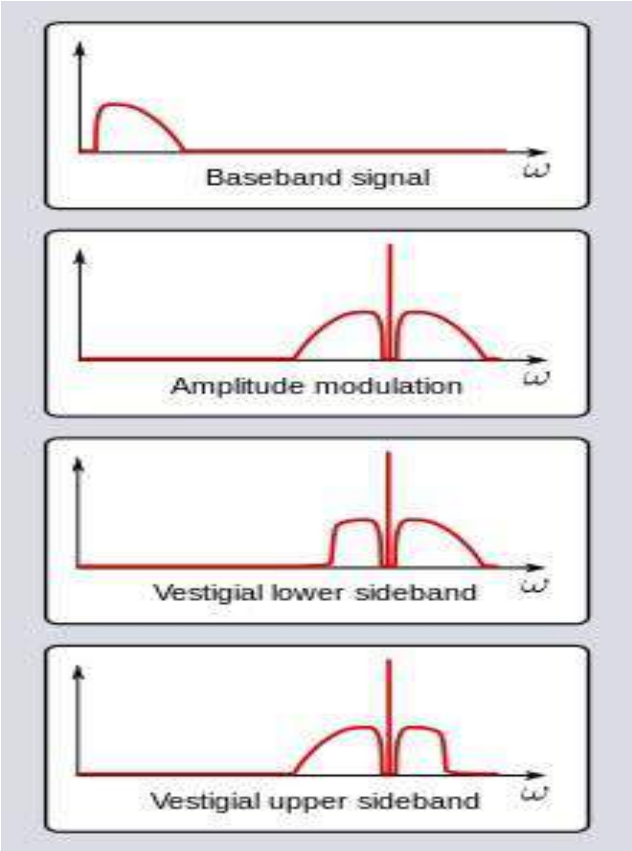


FIG2.6 Spectrum of VSB Signals

Vestigial sideband (VSB) transmission is a compromise between DSB and SSB

- In VSB modulation, one passband is passed almost completely whereas only a residual portion of the other sideband is retained in such a way that the demodulation process can still reproduce the original signal.
- VSB signals are easier to generate because some roll-off in filter edges is allowed. This results in system simplification. And their bandwidth is only slightly greater than that of SSB signals (-25 %).
- The filtering operation can be represented by a filter H(f) that passes some of the lower (or upper) sideband and most of the upper (or lower) sideband.

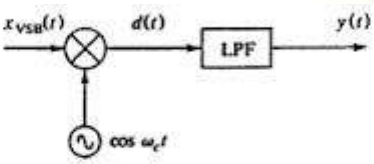


Fig. 3-11 VSB demodulator

- Heterodyning means the translating or shifting in frequency.
- By heterodyning the incoming signal at ωRF with the local oscillator frequency ωLO, the message is translated to an intermediate frequency ωIF, which is equal to either the sum or the difference of ωRF and ωIF.

- If ωIF = 0, the bandpass filter becomes a low-pass filter and the original baseband signal is presented at the output. This is called homodyning

Heterodyning: Image Response:

Methods to solve the image response in heterodyne receiver

1. Careful selection of intermediate frequency ωIF for a given frequency band.

2. Attenuate the image signal before heterodyning.

Advantages:

- VSB is a form of amplitude modulation intended to save bandwidth over regular AM. Portions of one of the redundant sidebands are removed to form a vestigialsidebandsignal.
- The actual information is transmitted in the sidebands, rather than the carrier; both sidebands carry the same information. Because LSB and USB are essentially mirror images of each other, one can be discarded or used for a second channel or for diagnostic purposes.

Disadvantages:

VSB transmission is similar to (SSB) transmission, in which one of the sidebands is completely removed. In VSB transmission, however, the second sideband is not completely removed, but is filtered to remove all but the desired range offrequencies.

2.5 DSB-SC:

Double-sideband suppressed-carrier transmission (DSB-SC) istransmissionin which frequencies produced byamplitude modulation(AM) are symmetrically spaced above and below thecarrierfrequencyandthecarrierlevelis reduced to the lowest practical level, ideally being completely suppressed.

✓ **Spectrum:**

DSB-SC is basically anamplitudemodulationwave without the carrier, therefore reducing power waste, giving it a 50% efficiency. This is an increase compared to normal AM transmission (DSB), which has a maximum efficiency of 33.333%, since 2/3 of the power is in the carrier which carries no intelligence, and each sideband carries the same information. Single Side Band (SSB) Suppressed Carrier is 100% efficient.

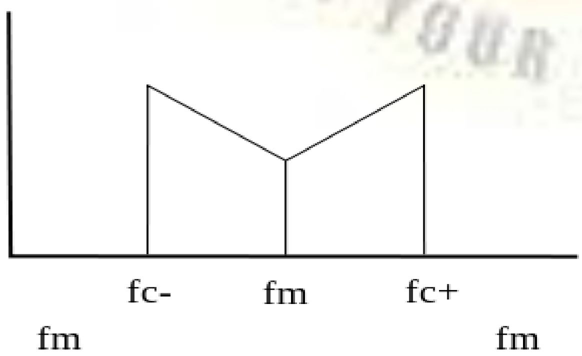


FIG 2.7 Spectrum plot of an DSB-SC signal

✓ **Generation:**

DSB-SC is generated by a mixer. This consists of a message signal multiplied by a carrier signal. The mathematical representation of this process is shown below, where the product-to-sum trigonometric identityis used.

$$\underbrace{V_m \cos(\omega_m t)}_{\text{Message}} \times \underbrace{V_c \cos(\omega_c t)}_{\text{Carrier}} = \underbrace{\frac{V_m V_c}{2}\left[\cos\left(\left(\omega_m + \omega_c\right)t\right) + \cos\left(\left(\omega_m - \omega_c\right)t\right)\right]}_{\text{Modulated Signal}}$$

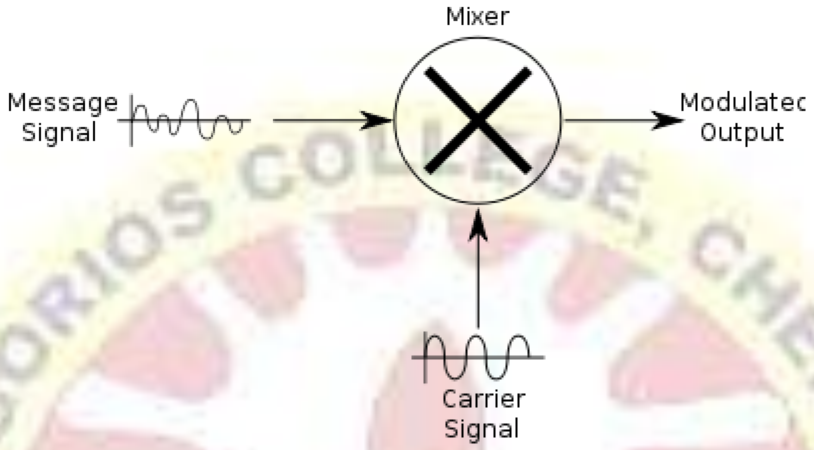


FIG 2.8 Generation of DSB-SC signal

- **Demodulation:**

Demodulation is done by multiplying the DSB-SC signal with the carrier signal just like the modulation process. This resultant signal is then passed through a low pass filter to produce a scaled version of original message signal. DSB-SC can be demodulated ifmodulationindexis less than unity.

$$\overbrace{\frac{V_m V_c}{2}\left[\cos\left(\left(\omega_m + \omega_c\right)t\right) + \cos\left(\left(\omega_m - \omega_c\right)t\right)\right]}^{\text{Modulated Signal}} \times \overbrace{V_c' \cos(\omega_c t)}^{\text{Carrier}}$$

$$= \left(\frac{1}{2}V_c V_c'\right)\underbrace{V_m \cos(\omega_m t)}_{\text{original message}} + \frac{1}{2}V_c V_c' V_m\left[\cos((\omega_m + 2\omega_c)t) + \cos((\omega_m - 2\omega_c)t)\right]$$

The equation above shows that by multiplying the modulated signal by the carrier signal, the result is a scaled version of the original message signal plus a second term. Since $\omega_c \gg \omega_m$, this second term is much higher in frequency than the original message.

Once this signal passes through a low pass filter, the higher frequency component is removed, leaving just the original message.

- **Distortion and Attentuation:**

For demodulation, the demodulation oscillator's frequency and phase must be exactly the same as modulation oscillator's, otherwise, distortion and/or attenuation will occur.

To see this effect, take the following conditions:

- Message signal to be transmitted: $f(t)$
- Modulation (carrier) signal: $V_c \cos(\omega_c)$
- Demodulation signal (with small frequency and phase deviations from the

$$f(t) \times V_c \cos(\omega_c) \times V_c' \cos\left[(\omega_c + \Delta\omega)t + \theta\right]$$
$$= \frac{1}{2} V_c V_c' f(t) \cos\left(\Delta\omega \cdot t + \theta\right) + \frac{1}{2} V_c V_c' f(t) \cos\left[(2\omega_c + \Delta\omega)t + \theta\right]$$
$$\xrightarrow{\text{After low pass filter}} \frac{1}{2} V_c V_c' f(t) \cos\left(\Delta\omega \cdot t + \theta\right)$$

The $\cos\left(\Delta\omega \cdot t + \theta\right)$

modulation

signal): $V_c' \cos\left[(\omega_c + \Delta\omega)t + \theta\right]$

The resultant signal can then be given by

terms results in distortion and attenuation of the original message signal. In particular, $\Delta\omega \cdot t$ contributes to distortion while θ adds to the attenuation.

2.7 SUPERHETERODYNE RECEIVER:

A superheterodyne receiver(often shortened to superhet) uses frequency mixing to convert a received signal to a fixed intermediate frequency (IF) which can be more conveniently processed than the original radio carrier frequency.

Basic Superheterodyne Block Diagram and Functionality:

The basic block diagram of a basic superhet receiver is shown below. This details the most basic form of the receiver and serves to illustrate the basic blocks and their function.

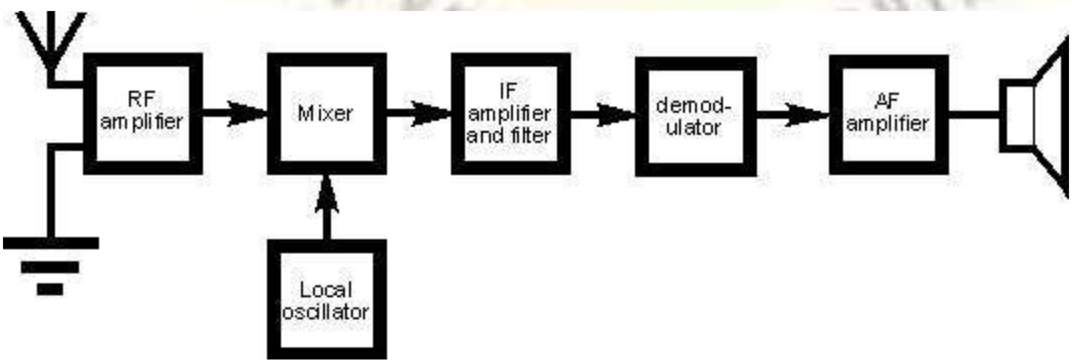


FIG2.10 Block Diagram of a Basic Superheterodyne Radio Receiver

The way in which the receiver works can be seen by following the signal as is passes through the receiver.

- Front end amplifier and tuning block: Signals enter the front end circuitry from the antenna. This circuit block performs two main functions:
  - **Tuning:** Broadband tuning is applied to the RF stage. The purpose of this is to reject the signals on the image frequency and accept those on the wanted frequency. It must also be able to track the local oscillator so that as the receiver is tuned, so the RF tuning remains on the required frequency. Typically the selectivity provided at this stage is not high. Its main purpose is to reject signals on the image frequency which is at a frequency equal to twice that of the IF away from the wanted frequency. As the tuning within this block provides all the rejection for the image response, it must be at a sufficiently sharp to reduce the image to an acceptable level. However the RF tuning may also help in preventing strong offchannel signals from entering the receiver and overloading elements of the receiver, in particular the mixer or possibly even the RF amplifier.
  - **Amplification:** In terms of amplification, the level is carefully chosen so that it does not overload the mixer when strong signals are present, but enables the signals
    to be amplified sufficiently to ensure a good signal to noise ratio is achieved. The amplifier must also be a low noise design. Any noise introduced in this block will be amplified later in the receiver.
- **Mixer / frequency translator block:** The tuned and amplified signal then enters one port of the mixer. The local oscillator signal enters the other port. The performance of the mixer is crucial to many elements of the overall receiver performance. It should eb as linear as possible. If not, then spurious signals will be generated and these may appear as 'phantom' received signals.
- **Local oscillator:** The local oscillator may consist of a variable frequency oscillator that can be tuned by altering the setting on a variable capacitor. Alternatively it may be a frequency synthesizer that will enable greater levels of stability and setting accuracy.
- **Intermediate frequency amplifier, IF block :** Once the signals leave the mixer they enter the IF stages. These stages contain most of the amplification in the receiver as well as the filtering that enables signals on one frequency to be separated from those on the next. Filters may consist simply of LC tuned transformers providing inter-stage coupling, or they may be much higher performance ceramic or even crystal filters, dependent upon what is required.
- **Detector / demodulator stage:** Once the signals have passed through the IF stages of the superheterodyne receiver, they need to be demodulated. Different demodulators are required for different types of transmission, and as a result some receivers may have a variety of demodulators that can be switched in to accommodate the different types of transmission that are to be encountered. Different demodulators used may include:

- **AM diode detector:** This is the most basic form of detector and this circuit block would simple consist of a diode and possibly a small capacitor to remove any remaining RF. The detector is cheap and its performance is adequate, requiring a sufficient voltage to overcome the diode forward drop. It is also not particularly linear, and finally it is subject to the effects of selective fading that can be apparent, especially on the HF bands.
- **Synchronous AM detector:** This form of AM detector block is used in where improved performance is needed. It mixes the incoming AM signal with another on the same frequency as the carrier. This second signal can be developed by passing the whole signal through a squaring amplifier. The advantages of the synchronous AM detector are that it provides a far more linear demodulation performance and it is far less subject to the problems of selective fading.
- **SSB product detector:** The SSB product detector block consists of a mixer and a local oscillator, often termed a beat frequency oscillator, BFO or carrier insertion oscillator, CIO. This form of detector is used for Morse code transmissions where the BFO is used to create an audible tone in line with the on-off keying of the transmitted carrier. Without this the carrier without modulation is difficult to detect. For SSB, the CIO re-inserts the carrier to make the modulation comprehensible.
- **Basic FM detector:** As an FM signal carries no amplitude variations a demodulator block that senses frequency variations is required. It should also be insensitive to amplitude variations as these could add extra noise. Simple FM detectors such as the Foster Seeley or ratio detectors can be made from discrete components although they do require the use of transformers.
- **PLL FM detector:**A phase locked loop can be used to make a very good FM demodulator. The incoming FM signal can be fed into the reference input, and the VCO drive voltage used to provide the detected audio output.
- Quadrature FM detector: This form of FM detector block is widely used within ICs. IT is simple to implement and provides a good linear output.

- **Audio amplifier:** The output from the demodulator is the recovered audio. This is passed into the audio stages where they are amplified and presented to the headphones or loudspeaker.

COMPARISION OF VARIOUS AM:

PARAMETER	**VSB - SC**	**SSB - SC**	**DSB-SC**
Definition	A vestigial sideband (in radio communication) is a sideband that has been only partly cut off or suppressed.	Single-sideband modulation (SSB) is a refinement of amplitude modulation that more efficiently uses electrical power and bandwidth.	Inradiocommunications, asidebandis abandoffrequencieshig her than or lower thanthecarrier frequency, containing power as a result of themodulationprocess.
Application	Tv broadcastings & Radio broadcastings	Tv broadcastings &ShortwaveRadio broadcastings	Tv broadcastings & Radio broadcastings Garage door opens
			keyless remotes
Uses	Transmits TV signals	Short wave radio communications	Two way radio communications.

APPLICATION & ITS USES:

- Radio broadcastings
- Tv broadcastings
- Garage door opens keyless remotes
- Transmits TV signals
- Short wave radio communications Two way radio communication.

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2. Simon Haykins - ―Communication Systems‖ John Wilsey 2005.
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GLOSSARY TERMS:

1. **Amplitude modulation:** The modulation of a wave by varying its amplitude, used especially as a means of broadcasting an audio signal by combining it with a radio carrier wave.
2. **The modulation index:** (modulation depth) of a modulation scheme describes by how much the modulated variable of the carrier signal varies around its unmodulated level.
3. **NarrowbandFM:** If the modulation index of *FM* is kept under 1, then the *FM* produced is regarded as narrow band *FM*.

4. **Frequency modulation (FM):** the encoding ofinformationinacarrierwaveby varying theinstantaneousfrequencyof the wave.
5. **Amplication:** The level is carefully chosen so that it does not overload the mixer when strong signals are present, but enables the signals to be amplified sufficiently to ensure a good signal to noise ratio is achieved.
6. **Modulation:** The process by which some of the characteristics of carrier wave is varied in accordance with the message signal.

TUTORIAL PROBLEMS:

1. A 400 watts carrier is modulated to a depth of 75% calculate the total power in a double side band full carrier AM wave.

   Solution:

   Carrier power Pc = 400 watts, m= 0.75

   Total power in a DSB-FC AM Wave = pt = $\mathbf{pc}(\mathbf{1}+\frac{m^2}{2})$

   $= 400(1+\frac{(0.75)^2}{2})$

   = 512.5 watts.

2. For the maximum envelope voltage Vmax = 20V and a minimum positive envelope voltage of Vmin = 6V Determine Modulation Index.

   Solution:

   Vmax = 20V ;Vmin = 6V

   (a) Modulation index, m $= \frac{vmax-vmin}{vmax+vmin}$

   = 14/26

   = 0.538.

   (b) Carrier Wave Vc:

   Vmax = Vc+ Vm

   20 = Vc+ Vm

   Vmin = Vc - Vm

   6 = Vc - VmVc = 13V.

WORKED OUT PROBLEMS:

1. Calculate the % power saving when the carrier and one of the sidebands are suppressed in an am wave modulated to depth of 60%.

(a)Total transmitted power $\boldsymbol{pt = pc = (1 + \frac{m^2}{2})}$

(b) $\mathbf{PSB} = \mathbf{pc}(\frac{m^2}{4})$

(c) % **power savings** $= \left(\frac{\text{pt}-\text{PSB}}{\text{PT}}\right) * \mathbf{100}$ Ans : 92.37%.

2. For an AM DSBFC envelope with Vmax = 40V and Vmin = 10V , determine the
   (a) Unmodulated carrier wave ; Vmax = Vc +Vm ; Vmin = Vc –Vm Ans : Vc = 25V.
   (b) % Modulation index = $\frac{(Vmax-Vmin)}{(Vmax+Vmin)}$ *100 .

UNIT 3

3.0 ANGLE MODULATION

PREREQUISTING ABOUT ANGLE MODULATION:

Angle modulation is a class ofanalogmodulation. These techniques are based on altering theangle(orphase) of asinusoidalcarrier wavetotransmitdata, as opposed to varying theamplitude, such as inAMtransmission.

Angle Modulation is modulation in which the angle of a sine-wave carrier is varied by a modulating wave. Frequency Modulation (FM) and Phase Modulation (PM) are two types of angle modulation. In frequency modulation the modulating signal causes the carrier frequency to vary. These variations are controlled by both the frequency and the amplitude of the modulating wave. In phase modulation the phase of the carrier is controlled by the modulating waveform.

The two main types of angle modulation are:

- Frequency modulation(FM), with its digital correspondencefrequency-shiftkeying(FSK).
- Phase modulation(PM), with its digital correspondencephase-shift keying(PSK).

CONTENT:

- **FREQUENCY & PHASE MODULATION**
- **NARROW BAND FM**

- **WIDE BAND FM**
- **GENERATION OF WIDE BAND FM**
- **TRANSMISSION BANDWIDTH**
- **FM TRANSMITTER**

3.1 FREQUENCY & PHASE MODULATION:

Besides using the amplitude of carrier to carrier information, one can also use the angle of a carrier to carrier information. This approach is called angle modulation, and includes frequency modulation (FM) and phase modulation (PM). The amplitude of the carrier is maintained constant. The major advantage of this approach is that it allows the trade-off between bandwidth and noise performance.

An angle modulated signal can be written as

$s(t) = A\cos\theta(t)$

where $\theta(t)$ is usually of the form $\theta(t) = 2\pi fct + \emptyset(t)$ and fc is the carrier frequency. The signal $\emptyset(t)$ is derived from the message signal m(t) . If $\emptyset(t) = kpm(t)$ for some constant kp ,the resulting modulation is called phase modulation. The parameter kp is called the phase sensitivity.Intelecommunication andsignal processing, frequency modulation (FM) is the encoding ofinformationin acarrier waveby varying theinstantaneous frequencyof the wave. (Compare withamplitude modulation, in which theamplitudeof the carrier wave varies, while the frequency remains constant.) Frequency modulation is known asphase modulationwhen the carrier phase modulation is the timeintegralof the FM signal.

If the information to be transmitted (i.e., thebaseband signal) is $x_m(t)$ and thesinusoidalcarrier

is $x_c(t) = A_c \cos(2\pi f_c t)$, where $f_c$is the carrier's base frequency, and A_c is the carrier's amplitude, the modulator combines the carrier with the baseband data signal to get the transmitted signal:

$$y(t) = A_c \cos\left(2\pi \int_0^t f(\tau)d\tau\right)$$
$$= A_c \cos\left(2\pi \int_0^t \left[f_c + f_\Delta x_m(\tau)\right] d\tau\right)$$
$$= A_c \cos\left(2\pi f_c t + 2\pi f_\Delta \int_0^t x_m(\tau)d\tau\right)$$

In this equation, $f(\tau)$ f_Δ is theinstantaneous frequencyof the oscillator and is thefrequency deviation, which represents the maximum shift away from f_c in one direction, assuming $x_m(t)$ is limited to the range ±1.

While most of the energy of the signal is contained within $f_c \pm f_\Delta$, it can be shown byFourier analysisthat a wider range of frequencies is required to precisely represent an FM signal.

Thefrequency spectrumof an actual FM signal has components extending infinitely, although their amplitude decreases and higher-order components are often neglected in practical design problems.

Sinusoidal baseband signal:

Mathematically, a baseband modulated signal may be approximated by asinusoidalcontinuous wavesignal with a frequency f_m.

The integral of such a signal is:

$$\int_0^t x_m(\tau)d\tau = \frac{A_m \cos(2\pi f_m t)}{2\pi f_m}$$

In this case, the expression for y(t) above simplifies to:

$$y(t) = A_c \cos\left(2\pi f_c t + \frac{f_\Delta}{f_m}\cos\left(2\pi f_m t\right)\right)$$

where the amplitude A_m of the modulatingsinusoidis represented by the peak deviation f_Δ Theharmonicdistribution of asine wavecarrier modulated by such asinusoidalsignal can be represented withBessel functions; this provides the basis for a mathematical understanding of frequency modulation in the frequency domain.

Modulation index:

As in other modulation systems, the value of the modulation index indicates by how much the modulated variable varies around its unmodulated level. It relates to variations in the carrier frequency:

$$h = \frac{\Delta f}{f_m} = \frac{f_\Delta |x_m(t)|}{f_m}$$

where f_m is the highest frequency component present in the modulating signal $x_m(t)$, and Δf is the peak frequency-deviation—i.e. the maximum deviation of theinstantaneous frequencyfrom the carrier frequency. For a sine wave modulation, the modulation index is seen to be the ratio of the amplitude of the modulating sine wave to the amplitude of the carrier wave (here unity).

If $h \ll 1$, the modulation is called narrowband FM, and its bandwidth is approximately $2f_m$. For digital modulation systems, for example Binary Frequency Shift Keying (BFSK), where a binary signal modulates the carrier, the modulation index is given by:

$$h = \frac{\Delta f}{f_m} = \frac{\Delta f}{\frac{1}{2T_s}} = 2\Delta f T_s$$

$$f_m = \frac{1}{2T_s}$$

where T_s is the symbol period, and is used as the highest frequency of the modulating binary waveform by convention, even though it would be more accurate to say it is the highest fundamental of the modulating binary waveform. In the case of digital modulation, the carrier f_c is never transmitted. Rather, one of two frequencies is transmitted, either $f_c + \Delta f$ or $f_c - \Delta f$, depending on the binary state 0 or 1 of the modulation signal.

If $h \gg 1$, the modulation is called wideband FM and its bandwidth is approximately $2f_\Delta$. While wideband FM uses more bandwidth, it can improve thesignal-to-noise ratiosignificantly; for example, doubling the value of Δf, while keeping f_m constant, results in an eight-fold improvement in the signal-to-noise ratio. (Compare this withChirp spread spectrum, which uses extremely wide frequency deviations to achieve processing gains comparable to traditional, betterknown spread-spectrum modes).

With a tone-modulated FM wave, if the modulation frequency is held constant and the modulation index is increased, the (non-negligible) bandwidth of the FM signal increases but the spacing between spectra remains the same; some spectral components decrease in strength as others increase. If the frequency deviation is held constant and the modulation frequency increased, the spacing between spectra increases.

Frequency modulation can be classified as narrowband if the change in the carrier frequency is about the same as the signal frequency, or as wideband if the change in the carrier frequency is much higher (modulation index >1) than the signal frequency.[6]For example, narrowband FM is used fortwo way radiosystems such asFamily Radio Service, in which the carrier is allowed to deviate only 2.5 kHz above and below the center frequency with speech signals of no more than 3.5 kHz bandwidth. Wideband FM is used forFM broadcasting, in which music and speech are transmitted with up to 75 kHz deviation from the center frequency and carry audio with up to a 20-kHz bandwidth.

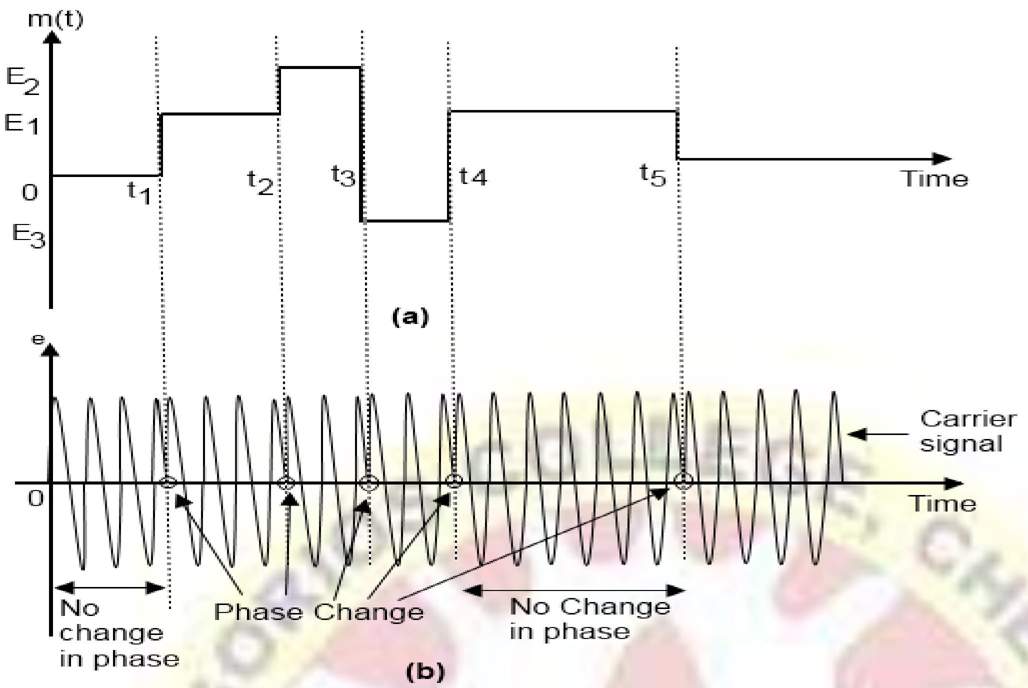
Carson's rule:

$$\mathrm{BT} = 2(\Delta \mathrm{f} + \mathrm{fm}).$$

3.2 PHASE MODULATION:

Phase Modulation (PM) is another form of angle modulation. PM and FM are closely related to each other. In both the cases, the total phase angle θ of the modulated signal varies. In an FM wave, the total phase changes due to the change in the frequency of the carrier corresponding to the changes in the modulating amplitude.

In PM, the total phase of the modulated carrier changes due to the changes in the instantaneous phase of the carrier keeping the frequency of the carrier signal constant. These two types of modulation schemes come under the category of angle modulation. However, PM is not as extensively used as FM.



Phase Modulation Waveforms

(a) Non-Sinusoidal Modulating Signal,m(t). (b) Phase-Modulated Carrier Signal e

At time t1, the amplitude of m(t) increases from zero to E1. Therefore, at t1, the phase modulated carrier also changes corresponding to E1, as shown in Figure (a). This phase remains to this attained value until time t2, as between t1 and t2, the amplitude of m(t) remains constant at El. At t2, the amplitude of m(t) shoots up to E2, and therefore the phase of the carrier again increases corresponding to the increase in m(t). This new value of the phase attained at time t2remains constant up to time t3. At time t3, m(t) goes negative and its amplitude becomes E3. Consequently, the phase of the carrier also changes and it decreases from the previous value attained at t2. The decrease in phase corresponds to the decrease in amplitude of m(t). The phase of the carrier remains constant during the time interval between t3 and t4. At t4, m(t) goes positive to reach the amplitude El resulting in a corresponding increase in the phase of modulated carrier at time t4. Between t4 and t5, the phase remains constant. At t5 it decreases to the phase of the unmodulated carrier, as the amplitude of m(t) is zero beyond t5.

Equation of a PM Wave:

To derive the equation of a PM wave, it is convenient to consider the modulating signal as a pure sinusoidal wave. The carrier signal is always a high frequency sinusoidal wave. Consider the modulating signal, em and the carrier signalec, as given by, equation 1 and 2, respectively.

$e_m = E_m \cos \omega_m t$ ------------(1) $e_c = E_c \sin \omega_c t$-----(2)

The initial phases of the modulating signal and the carrier signal are ignored in Equations (1) and (2) because they do not contribute to the modulation process due to their constant values. After PM, the phase of the carrier will not remain constant. It will vary according to the modulating signal em maintaining the amplitude and frequency as constants. Suppose, after PM, the equation of the carrier is represented as:

$e = E_c \sin \theta$ ------------------(3)

Where θ, is the instantaneous phase of the modulated carrier, and sinusoid ally varies in proportion to the modulating signal. Therefore, after PM, the instantaneous phase of the modulated carrier can be written as:

$\theta = \omega_c t + K_p e_m$ -------------------(4)

Where, kp is the constant of proportionality for phase modulation.

Substituting Equation (1) in Equation (4), yon get:

$\theta = \omega_c t + K_p E_m \cos \omega_m t$ ---------------------(5)

In Equation (5), the factor, kpEm is defined as the modulation index, and is given as: $m_p = K_p E_m$ ------------------------(6)

where, the subscript p signifies; that mp is the modulation index of the PM wave. Therefore, equation (5) becomes

$\theta = \omega_c t + m_p \cos \omega_m t$ ---------------------(7)

Substituting Equation (7) and (3), you get:

$e = E_c \sin (\omega_c t + m_p \cos \omega_m t)$ --------------------(8)

3.3 NARROW BAND FM MODULATION:

The case where $|\theta m(t)| \ll 1$ for all t is called narrow band FM. Using the approximations $\cos x \simeq 1$ and $\sin x \simeq x$ for $|x| \ll 1$, the FM signal can be approximated as: $s(t) = Ac \cos[\omega ct + \theta m(t)]$

$= Ac \cos \omega ct \cos \theta m(t) - Ac \sin \omega ct \sin\theta m(t) \simeq Ac \cos \omega ct - Ac\theta m(t) \sin \omega ct$

or in complex notation

$s(t) = ACRE\{e^{jwct} (1 + j\theta m(t)\}$

This is similar to the AM signal except that the discrete carrier component Ac co$s wc(t)$ is 90∘ out of phase with the sinusoid Ac sin$wc(t)$multiplying the phase angle θm(t). The spectrum of narrow band FM is similar to that of AM.

- ✓ **The Bandwidth of an FM Signal:**

The following formula, known as Carson‘s rule is often used as an estimate of the FM signalbandwidth: $BT = 2(\Delta f + fm)$ Hz where Δf is the peak frequency deviation and fm is the maximum baseband message frequency component.

- ✓ **FM Demodulation by a Frequency Discriminator:**

A frequency discriminator is a device that converts a received FM signal into a voltage that is proportional to the instantaneous frequency of its input without using a local oscillator and, consequently, in a non coherent manner.

- When the instantaneous frequency changes slowly relative to the time-constants of the filter, a quasi-static analysis can be used.
- In quasi-static operation the filter output has the same instantaneous frequency as the input but with an envelope that varies according to the amplitude response of the filter at the instantaneous frequency.
- The amplitude variations are then detected with an envelope detector like the ones used for AM demodulation.
-

An FM Discriminator Using the Pre-Envelope:

When θm(t) is small and band-limited so that cos θm(t) and sinθm(t) are essentially band-limited signals with cut off frequencies less than fc, the pre-envelope of the FM signal is

$s+(t) = s(t) + j\hat{}s(t) = Ac e^{j(\omega ct+\theta m(t))}$

The angle of the pre-envelope is $\varphi'(t) = \arctan[\hat{}s(t)/s(t)] = \omega ct + \theta m(t)$

The derivative of the phase is $=\omega ct + k\theta m(t)$

$$\frac{d\varphi(t)}{dt} = \frac{s(t)d}{dt^{s(t)}} - \frac{s^{t\frac{d}{dt}s(t)}}{s2(t)+s\hat{}2(t)} = \omega ct + k\omega m\ (t)$$

which is exactly the instantaneous frequency. This can be approximated in discrete-time by using FIR filters to form the derivatives and Hilbert transform. Notice that the denominator is the squared envelope of the FM signal. This formula can also be derived by observing,

$$\frac{d}{dt}s(t) = \frac{d}{dt}ACcos[\omega ct + \theta m(t)] = -AC[\omega ct + k\omega m(t)]\sin[\omega ct + \theta m(t)]$$

$$\frac{d}{dt}s\hat{}(t) = \frac{d}{dt}ACsin[\omega ct + \theta m(t)] = AC[\omega ct + k\omega m(t)]\cos[\omega ct + \theta m(t)]$$

So,

$$\frac{s(t)d}{dts\hat{}(t)} - \frac{s\hat{}(t)d}{dts(t)} = AC^2[\omega ct + k\omega m(t)] * \cos2[wct + \theta m(t) + \sin2[wct + \theta m(t)$$

The bandwidth of an FM discriminator must be at least as great as that of the received FM signal which is usually much greater than that of the baseband message. This limits the degree of noise reduction that can be achieved by preceding the discriminator by a bandpass receive filter.

Using a Phase-Locked Loop for FM Demodulation:

A device called a phase-locked loop (PLL) can be used to demodulate an FM signal with better performance in a noisy environment than a frequency discriminator. The block diagram of a discrete-time version of a PLL as shown in figure,

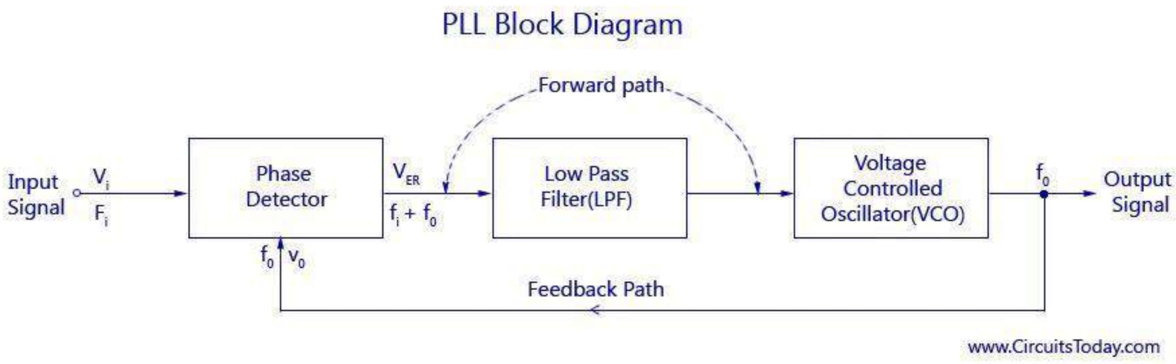


FIG 3.2 PLL Block diagram

The block diagram of a basic PLL is shown in the figure below. It is basically a flip flop consisting of a phase detector, a low pass filter (LPF),and a Voltage Controlled Oscillator (VCO) The input signal Vi with an input frequency fi is passed through a phase detector. A phase detector

basically a comparator which compares the input frequency fiwith the feedback frequency fo .The phase detector provides an output error voltage Ver (=fi+fo),which is a DC voltage. This DC voltage is then passed on to an LPF. The LPF removes the high frequency noise and produces a steady DC level, Vf (=Fi-Fo). Vf also represents the dynamic characteristics of the PLL.

The DC level is then passed on to a VCO. The output frequency of the VCO (fo) is directly proportional to the input signal. Both the input frequency and output frequency are compared and adjusted through feedback loops until the output frequency equals the input frequency. Thus the PLL works in these stages – free-running, capture and phase lock.

As the name suggests, the free running stage refer to the stage when there is no input voltage applied. As soon as the input frequency is applied the VCO starts to change and begin producing an output frequency for comparison this stage is called the capture stage. The frequency comparison stops as soon as the output frequency is adjusted to become equal to the input frequency. This stage is called the phase locked state.

Comments on PLL Performance:

• The frequency response of the linearized loop characteristics of a band-limited differentiator. • The loop parameters must be chosen to provide a loop bandwidth that passes the desired baseband message signal but is as small as possible to suppress out-of-band noise.

• The PLL performs better than a frequency discriminator when the FM signal is corrupted by additive noise. The reason is that the bandwidth of the frequency discriminator must be large enough to pass the modulated FM signal while the PLL bandwidth only has to be large enough to pass the baseband message. With wideband FM, the bandwidth of the modulated signal can be significantly larger than that of the baseband message.

•

Bandwidth of FM PLL vs. Costas Loop:

The PLL described in this experiment is very similar to the Costas loop presented in coherent demodulation of DSBSC-AM. However, the bandwidth of the PLL used for FM demodulation must be large enough to pass the baseband message signal, while the Costas loop is used to generate a stable carrier reference signal so its bandwidth should be very small and just wide enough to track carrier drift and allow a reasonable acquisition time.

3.4 WIDE-BAND FM:

$s(t) = AC\cos(2\pi fct + \varphi(t)$

Finding its FT is not easy: $\phi(t)$ is inside the cosine.

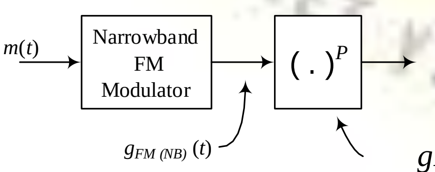
To analyze the spectrum, we use complex envelope.

s(t) can be written as: Consider single tone FM: $s(t) = AC\cos(2\pi fct + \beta\sin 2\pi fm(t))$

Wideband FM is defined as the situation where the modulation index is above 0.5. Under these circumstances the sidebands beyond the first two terms are not insignificant. Broadcast FM stations use wideband FM, and using this mode they are able to take advantage of the wide bandwidth available to transmit high quality audio as well as other services like a stereo channel, and possibly other services as well on a single carrier.
The bandwidth of the FM transmission is a means of categorising the basic attributes for the signal, and as a result these terms are often seen in the technical literature associated with frequency modulation, and products using FM. This is one area where the figure for modulation index is used.

GENERATION OF WIDEBAND FM SIGNALS:

Indirect Method for Wideband FM Generation:
Consider the following block diagram



gFM (WB) (t)

Assume a BPF is included in this block to pass the signal with the highest carrier freuqnecy and *reject all others* FIG 3.3 Block diagram of FM generation

A narrowband FM signal can be generated easily using the block diagram of the narrowband FM modulator that was described in a previous lecture. The narrowband FM modulator generates a narrowband FM signal using simple components such as an integrator (an OpAmp), oscillators, multipliers, and adders. The generated narrowband FM signal can be converted to a wideband FM signal by simply passing it through a non–linear device with power P. Both the carrier frequency and the frequency deviation f of the narrowband signal are increased by a factor P. Sometimes, the desired increase in the carrier frequency and the desired increase in f are different. In this case, we increase f to the desired value and use a frequency shifter (multiplication by a sinusoid followed by a BPF) to change the carrier frequency to the desired value.

- ✓ **System 1:**

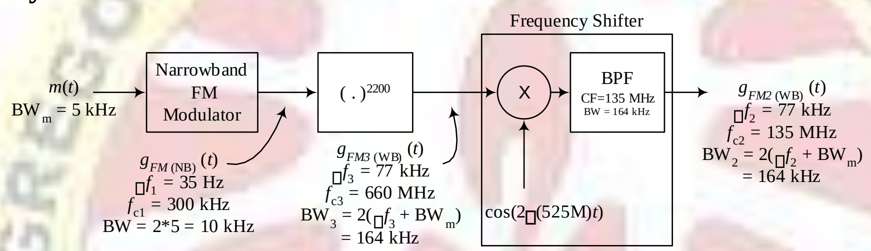


FIG 3.4 Block diagram of FM generation

In this system, we are using a single non–linear device with an order of 2200 or multiple devices with a combined order of 2200. It is clear that the output of the non–linear device has the correct f but an incorrect carrier frequency which is corrected using a the frequency shifter with an oscillator that has a frequency equal to the difference between the frequency of its input signal and the desired carrier frequency. We could also have used an oscillator with a frequency that is the sum of the frequencies of the input signal and the desired carrier frequency. This system is characterized by having a frequency shifter with an oscillator frequency that is relatively large.

- ✓ **System 2:**

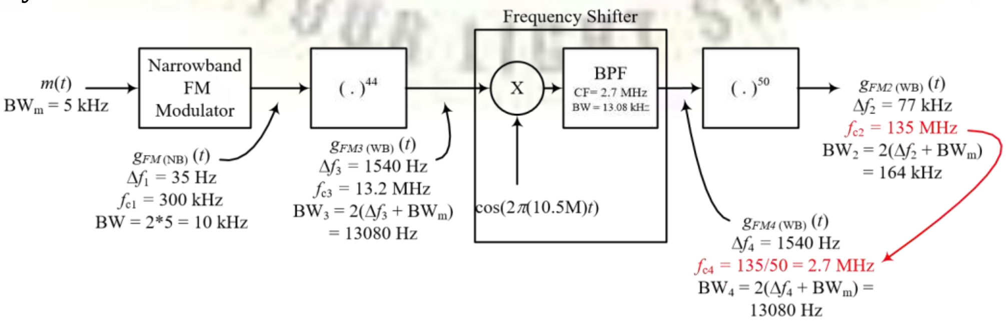


FIG 3.5 Block diagram of FM generation

In this system, we are using two non–linear devices (or two sets of non–linear devices) with orders 44 and 50 (44*50 = 2200). There are other possibilities for the factorizing 2200 such as 2*1100,4*550,8*275,10*220.. Depending on the available components, one of these factorizations may be better than the others. In fact, in this case, we could have used the same factorization but put 50 first followed by 44. We want the output signal of the overall system to be as shown in the block diagram above, so we have to insure that the input to the non–linear device with order 50 has the correct carrier frequency such that its output has a carrier frequency of 135 MHz. This is done by dividing the desired output carrier frequency by the non–linearity order of 50, which gives 2.7 Mhz. This allows us to figure out the frequency of the require oscillator which will be in this case either 13.2–2.7 = 10.5 MHz or 13.2+2.7 = 15.9 MHz. We are generally free to choose which ever we like unless the available components dictate the use of one of them and not the other. Comparing this system with System 1 shows that the frequency of the oscillator that is required here is significantly lower (10.5 MHz compared to 525 MHz), which is generally an advantage.

.TRANSMISSION BANDWIDTH:

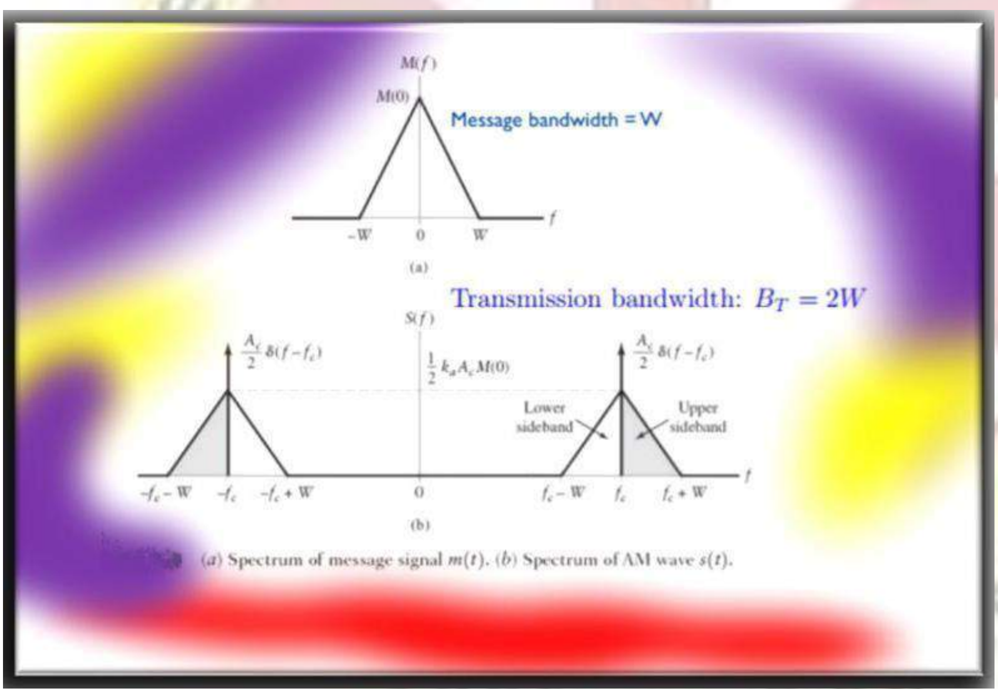


FIG 2.6 Spectrum of FM Bandwidth

FM TRANSMITTER

- **Indirect method (phase shift) of modulation**

The part of the Armstrong FM transmitter (Armstrong phase modulator) which is expressed in dotted lines describes the principle of operation of an Armstrong phase modulator. It should be noted, first that the output signal from the carrier oscillator is supplied to circuits that perform the task of modulating the carrier signal. The oscillator does not change frequency, as is the case of direct FM. These points out the major advantage of phase modulation (PM), or indirect FM, over direct FM. That is the phase modulator is crystal controlled for frequency.

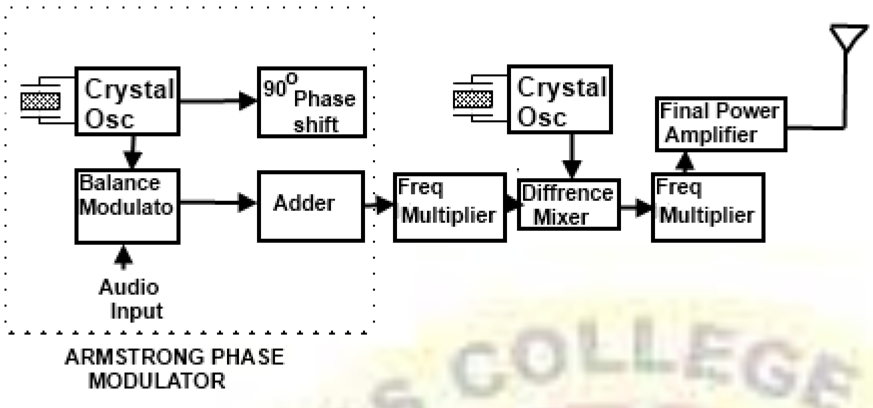


FIG 3.7 Armstrong Modulator

The crystal-controlled carrier oscillator signal is directed to two circuits in parallel. This signal

(usually a sine wave) is established as the reference past carrier signal and is assigned a value 0°.The balanced modulator is an amplitude modulator used to form an envelope of double sidebands and to suppress the carrier signal (DSSC). This requires two input signals, the carrier signal and the modulating message signal. The output of the modulator is connected to the adder circuit; here the 90° phase-delayed carriers signal will be added back to replace the suppressed carrier. The act of delaying the carrier phase by 90° does not change the carrier frequency or its waveshape. This signal identified as the 90° carrier signal.

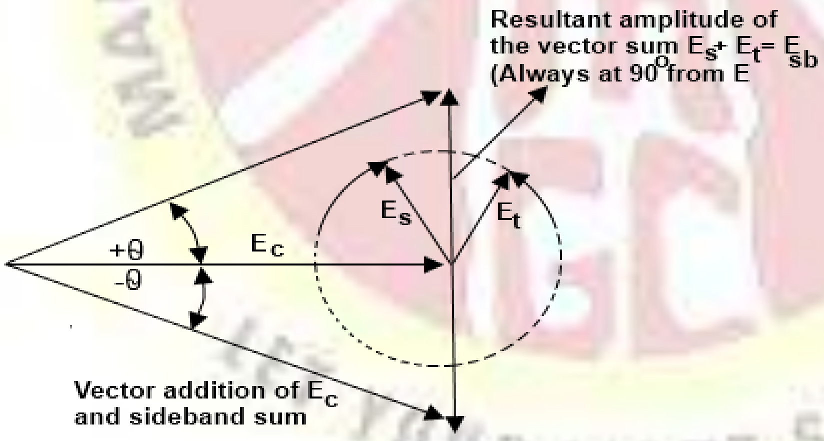


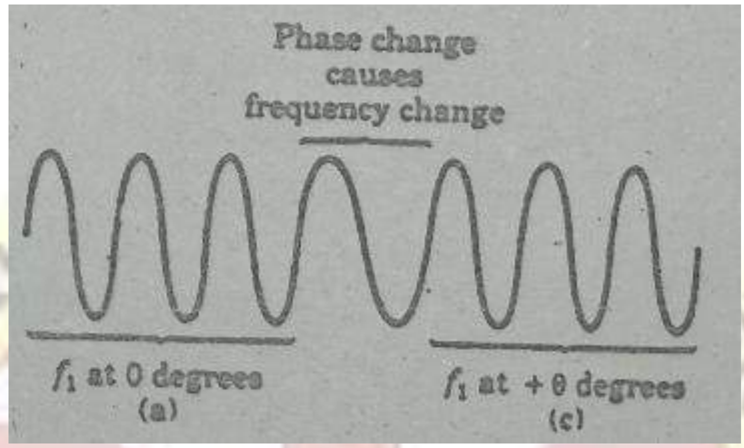
FIG 2.8 Phasor diagram of Armstrong Modulator

$$\% \text{ of modulation} = \frac{E_{max} - E_{min}}{E_{max} + E_{min}} \times 100$$

The carrier frequency change at the adder output is a function of the output phase shift and is found by. fc = $\Delta\theta f_s$ (in hertz)

When θ is the phase change in radians and f_s is the lowest audio modulating frequency. In most FM radio bands, the lowest audio frequency is 50Hz. Therefore, the carrier frequency change at the adder output is 0.6125 x 50Hz = ± 30Hz since 10% AM represents the upper limit of carrier voltage change, then ± 30Hz is the maximum deviation from the modulator for PM.

The 90° phase shift network does not change the signal frequency because the components and resulting phase change are constant with time. However, the phase of the adder output voltage is in a continual state of change brought about by the cyclical variations of the message signal, and during the time of a phase change, there will also be a frequency change.



In figure. (c). during time (a), the signal has a frequency f_1, and is at the zero reference phase. During time (c), the signal has a frequency f_1 but has changed phase to θ. During time (b) when the phase is in the process of changing, from 0 to θ. the frequency is less than f_1.

Using Reactance modulator direct method

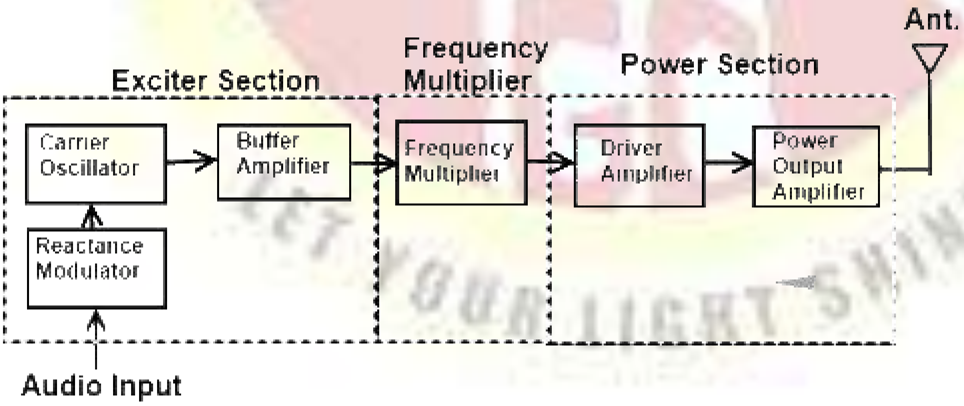


FIG 3.9 Reactance Modulator The FM transmitter has three basic sections.

1. The exciter section contains the carrier oscillator, reactance modulator and the buffer amplifier.
2. The frequency multiplier section, which features several frequency multipliers.

3. The poweroutputection, which includes a low- level power amplifier, the final power amplifier, and the impedance matching network to properly load the power section with the antenna impedance.

The essential function of each circuit in the FM transmitter may be described as follows.

- ✓ **The Exciter**
  1. The function of the carrier oscillator is to generate a stable sine wave signal at the rest frequency, when no modulation is applied. It must be able to linearly change frequency when fully modulated, with no measurable change in amplitude.
  2. The buffer amplifier acts as a constant high-impedance load on the oscillator to help stabilize the oscillator frequency. The buffer amplifier may have a small gain.
  3. The modulator acts to change the carrier oscillator frequency by application of the message signal. The positive peak of the message signal generally lowers the oscillator's frequency to a point below the rest frequency, and the negative message peak raises the oscillator frequency to a value above the rest frequency. The greater the peak-to-peak message signal, the larger the oscillator deviation.
- ✓ Frequency multipliers are tuned-input, tuned-output RF amplifiers in which the output resonant circuit is tuned to a multiple of the input frequency. Common frequency multipliers are 2x, 3x and 4x multiplication. A 5x Frequency multiplier is sometimes seen, but its extreme low efficiency forbids widespread usage. Note that multiplication is by whole numbers only. There can not a 1.5x multiplier, for instance.
- ✓ The final power section develops the carrier power, to be transmitted and often has a low-power amplifier driven the final power amplifier. The impedance matching network is the same as for the AM transmitter and matches the antenna impedance to the correct load on the final over amplifier.
- ✓ **Frequency Multiplier**

A special form of class C amplifier is the frequency. multiplier. Any class C amplifier is capable of performing frequency multiplidàtion if the tuned circuit in the collector resonates at some integer multiple of the input frequency.

For example a frequency doubler can be constructed by simply connecting a parallel tuned circuit in the collector of a class C amplifier that resonates at twice the input frequency. When the collector current pulse occurs, it excites or rings the tuned circuit at twice the input frequency. A current pulse flows for every other cycle of the input.

A Tripler circuit is constructed in the same way except that the tuned circuit resonates at 3 times the input - frequency. In this way, the tuned circuit receives one input pulse for every three cycles of oscillation it produces Multipliers can be constructed to increase the input frequency by any integer factor up to approximately 10. As' the multiplication factor gets higher, the power output of the multiplier decreases. For most practical applications, the best result is obtained with multipliers of 2 and 3.

Another way to look the operation of class C multipliers is .to .remember that the non-sinusoidal current pulse is rich in harmonics. Each time the pulse occurs, the second, third, fourth, fifth, and higher harmonics are generated. The purpose of the tuned circuit in the collector is to act as a filter to select the desired harmonics.

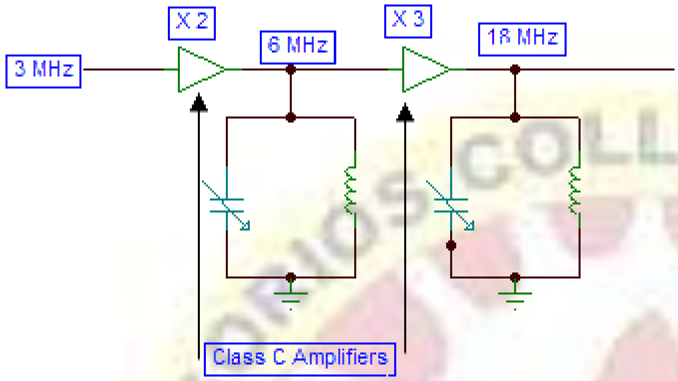


FIG 3.10 Block Diagram of Frequency Multiplier - 1

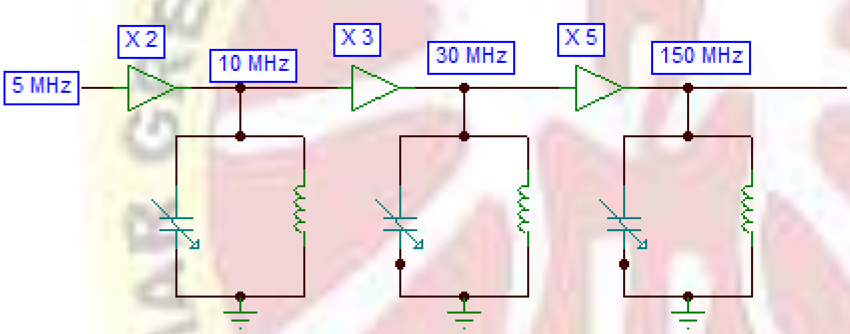


FIG 3.10 Block Diagram of Frequency Multiplier - 2

In many applications a multiplication factor greater than that achievable with a single multiplier stage is required. In such cases two or more multipliers are cascaded to produce an overall multiplication of 6. In the second example, three multipliers provide an overall multiplication of

30. The total multiplication factor is simply the product of individual stage multiplication factors. □**Reactance Modulator**

The reactance modulator takes its name from the fact that the impedance of the circuit acts as a reactance (capacitive or inductive) that is connected in parallel with the resonant circuit of the Oscillator. The varicap can only appear as a capacitance that becomes part of the frequency determining branch of the oscillator circuit. However, other discrete devices can appear as a capacitor or as an inductor to the oscillator, depending on how the circuit is arranged. A colpitts oscillator uses a capacitive voltage divider as the phase-reversing feedback path and would most likely tapped coil as the phase-reversing element in the feedback loop and most commonly uses a modulator that appears inductive

.

COMPARISION OF VARIOUS MODULATIONS:

- ✓ **Comparisons of Various Modulations:**

Amplitude modulation	**Frequency modulation**	**Phase modulation**
1. Amplitude of the carrier wave is varied in accordance with the message signal.	1. Frequency of the carrier wave is varied in accordance with the message signal.	1. Phase of the carrier wave is varied in accordance with the message signal.
2.Much affected by noise.	2.More immune to the noise.	2. Noise voltage is constant.
3.System fidelity is poor.	3.Improved system fidelity.	Improved system fidelity.
4.Linear modulation	4.Non Linear modulation	4.Non Linear modulation

- ✓ **Comparisons of Narrowband and Wideband FM:**
- ✓

Narrowband FM	**Wideband FM**
1. Modulation index > 1.	1. Modulation index < 1.
2.Bandwidth B = $2\Delta f Hz$.	2.Bandwidth B = $2fmHz$.
3. Occupies more bandwidth.	3. Occupies less bandwidth.
4.Used in entertainment broadcastings	4.Used in FM Mobile communication services.

APPLICATION & ITS USES:

- Magnetic Tape Storage.
- Sound
- Noise Fm Reduction
- Frequency Modulation (FM) stereo decoders, FM Demodulation networks for FM operation.
- Frequency synthesis that provides multiple of a reference signal frequency.
- Used in motor speed controls, tracking filters.

TUTORIAL PROBLEMS:

1. If the modulating frequency is 1 kHZ and the maximum deviation is 10 KHZ, what is the required for an FM signal?

Solution:

$\mathrm{fm} = 1\mathrm{khz}, \Delta \mathrm{f} = 10\mathrm{khz}$

Bandwidth = $2(mf + 1)$fm

$mf = \frac{\Delta f}{fm} = 10$

B = 22KHZ.

2. Cosider an angle modulated wave (pm) $v = 10\sin(\omega ct + 5\sin wmt)$, Let fm = 2khz calculate the modulation index and find the bandwidth.

Solution:

The equation is of the form, $v = 10\sin(\omega ct + 5\sin wmt)$, A = 10v, fm = 2 kHz, m = 5

Bandwidth = $2(m+1)fm = 24\ khz$.

WORKED OUT PROBLEMS:

1. Find the deviation ratio if the maximum frequency deviation is 60 kHz and the fm = 10khz.

$\textbf{Deviation ratio} = \left(\frac{\Delta f}{fm}\right);\ \textbf{Ans : 6}$

2. Angle modulated signal is given by $xn(t) = 5\cos[2\pi 10^6 t + 0.2\cos 200\,\pi t]$ Find whether xa(t) is

PM or FM?

Ans :xa(t) can be either FM or PM.

UNIT IV

ANALOG PULSE CODE MODULATION

INTRODUCTION

The word communication arises from the Latin word “commūnicāre”, which means “to share”. Communication is the basic step for the exchange of information.

For example, a baby in a cradle, communicates with a cry that she needs her mother. A cow moos loudly when it is in danger. A person communicates with the help of a language. Communication is the bridge to share.

Communication can be defined as the process of exchange of information through means such as words, actions, signs, etc., between two or more individuals.

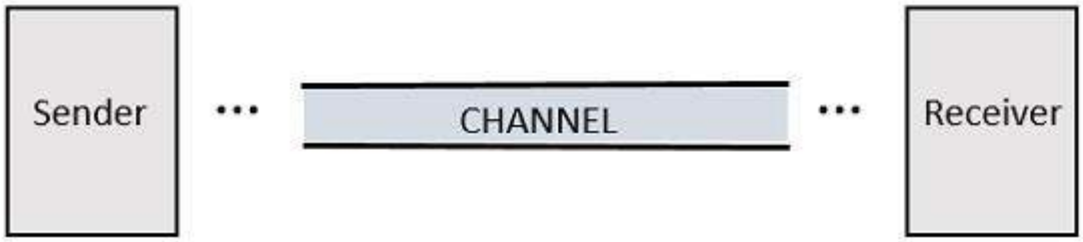
Need for Communication

For any living being, while co-existing, there occurs the necessity of exchange of some information. Whenever a need for exchange of information arises, some means of communication should exist. While the means of communication, can be anything such as gestures, signs, symbols, or a language, the need for communication is inevitable.

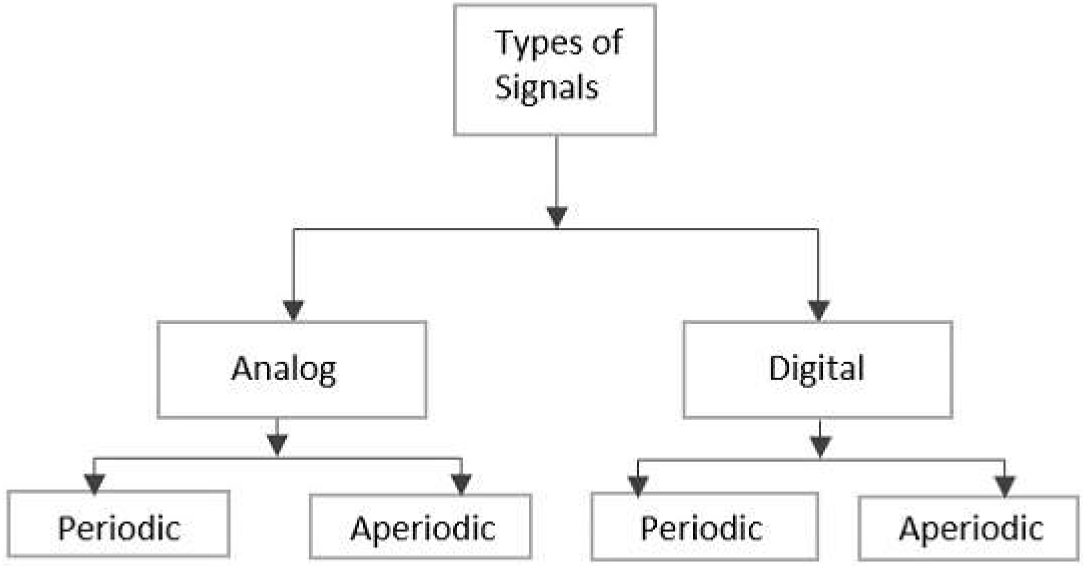
Language and gestures play an important role in human communication, while sounds and actions are important for animal communication. However, when some message has to be conveyed, a communication has to be established.

Parts of Communication System

Any system which provides communication, consists of the three important and basic parts as shown in the following figure.



- The **Sender** is the person who sends a message. It could be a transmitting station from where the signal is transmitted.
- The **Channel** is the medium through which the message signals travel to reach the destination.
- The **Receiver** is the person who receives the message. It could be a receiving station where the signal transmitted is received.



What is Signal Modulation?

A message carrying signal has to get transmitted over a distance and for it to establish a reliable communication, it needs to take the help of a high frequency signal which should not affect the original characteristics of the message signal.

The characteristics of the message signal, if changed, the message contained in it also alters. Hence it is a must to take care of the message signal. 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.

Modulation is the process of changing the parameters of the carrier signal, in accordance with the instantaneous values of the modulating signal.

Need for Modulation

The baseband signals are incompatible for direct transmission. For such a signal, to travel longer distances, its strength has to be increased by modulating with a high frequency carrier wave, which doesn't affect the parameters of the modulating signal.

Advantages of Modulation

The antenna used for transmission, had to be very large, if modulation was not introduced. The range of communication gets limited as the wave cannot travel to a distance without getting distorted.

Following are some of the advantages for implementing modulation in the communication systems.

- Antenna size gets reduced.
- No signal mixing occurs.
- Communication range increases.
- Multiplexing of signals occur.
- Adjustments in the bandwidth is allowed.
- Reception quality improves.

Signals in the Modulation Process

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

Message or Modulating Signal

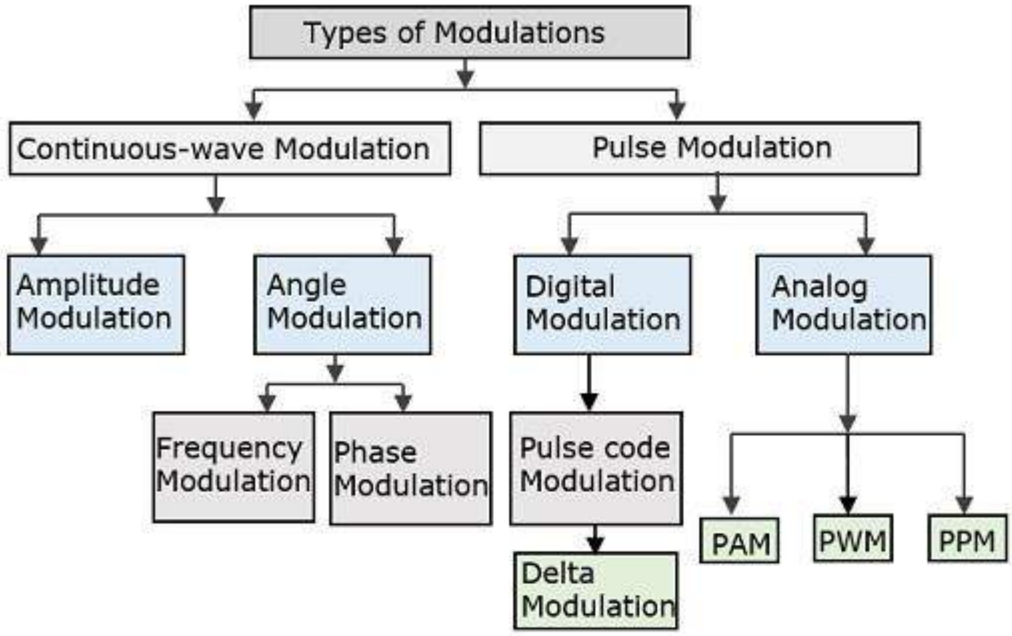
The signal which contains a message to be transmitted, is called as a **message signal**. It is a baseband signal, which has to undergo the process of modulation, to get transmitted. Hence, it is also called as the **modulating signal**.

Carrier Signal

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

Modulated Signal

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



The types of modulations are broadly classified into continuous-wave modulation and pulse modulation.

Continuous-wave Modulation

In the 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**.

  The angle modulation is further divided into frequency 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, 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)**, which will be discussed in subsequent chapters. Hence, PCM is a technique where the analog signals are converted into a digital form.

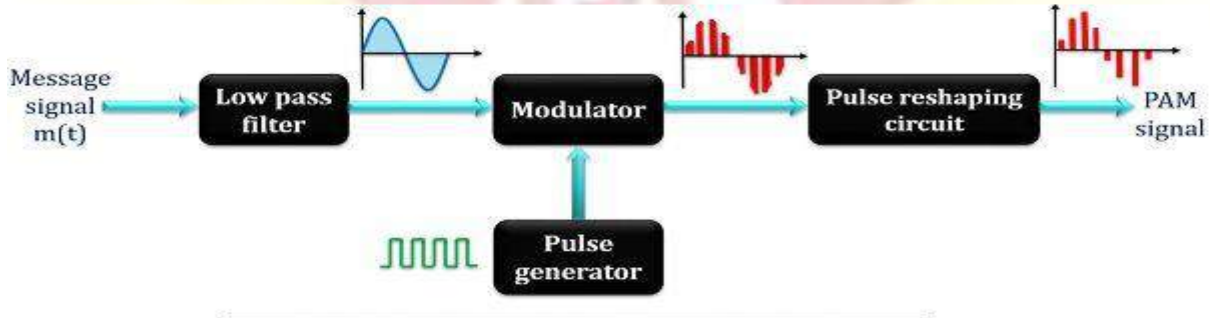
Pulse Amplitude Modulation

Pulse Amplitude Modulation (PAM) is an analog modulating scheme in which the amplitude of the pulse carrier varies proportional to the instantaneous amplitude of the message signal.

The pulse amplitude modulated signal, will follow the amplitude of the original signal, as the signal traces out the path of the whole wave. In natural PAM, a signal sampled at the Nyquist rate is reconstructed, by passing it through an efficient **Low Pass Frequency (LPF)** with exact cutoff frequency

Block diagram of PAM generator

The figure below shows the block diagram of a PAM generato



Block diagram of PAM signal generator

Here the modulating signal is given to the low pass filter in order to band limit the message signal.

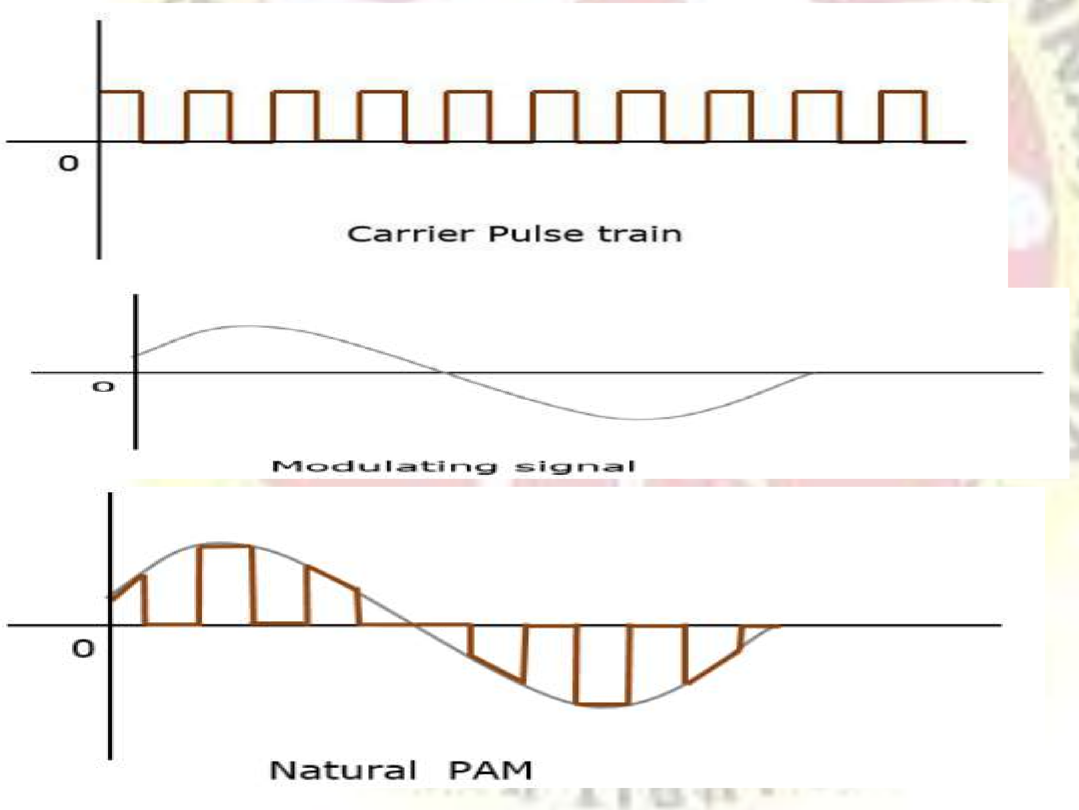
The LPF at the beginning is placed in order to avoid aliasing of the samples. The LPF passes only the low-frequency component of the signal and eliminates the high-frequency signal component. The output of LPF is then provided to a modulator, where it gets mixed with the rectangular pulse train.

Basically, the pulsed carrier gets modulated by the message signal here. The rectangular carrier pulse is generated by the pulse generator circuit.

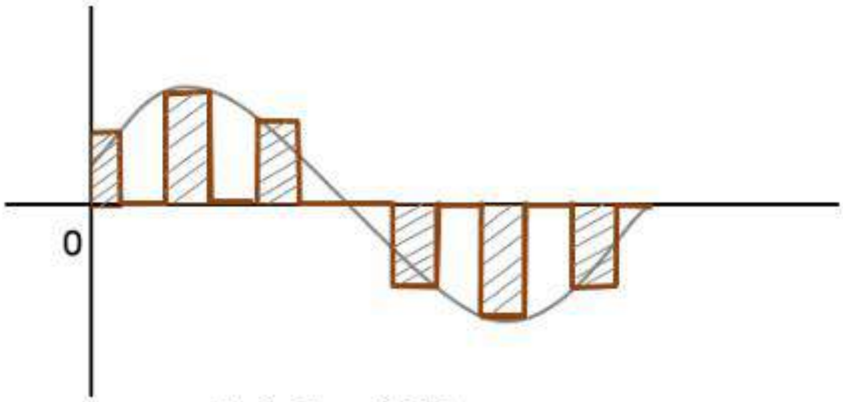
The modulator generates a **pulse amplitude modulated signal**. The sampled pulses can be achieved either by natural or flat top sampling. The output of the modulator is provided to the pulse reshaping circuit. This basically shapes the pulses so that it can be easily detected at the receiver.

As we have already discussed why flat top sampling is preferred over natural sampling. Regeneration of flat-top pulses by the repeater is somewhat easier in case of long distance signal transmission

The following figures explain the Pulse Amplitude Modulation.



Though the PAM signal is passed through an LPF, it cannot recover the signal without distortion. Hence to avoid this noise, flat-top sampling is done as shown in the following figure.



Flat-Top PAM

Flat-top sampling is the process in which sampled signal can be represented in pulses for which the amplitude of the signal cannot be changed with respect to the analog signal, to be sampled. The tops of amplitude remain flat. This process simplifies the circuit design.

Advantages of Pulse Amplitude Modulation

- PAM is the simplest form of pulse modulation.
- Its implementation is quite easy.

Disadvantages of Pulse Amplitude Modulation

- The transmission bandwidth required is very large.
- Due to the variation in amplitude, the power required by the generating unit also varies.
- Less immune to noise due to amplitude variation.

Applications of Pulse Amplitude Modulation

It is used in LED lighting, in microcontrollers in order to produce control signals and in the Ethernet communication system.

Pulse Width Modulation (PWM)

Definition: A modulation technique where the width of the pulses of the pulsed carrier wave is changed according to the modulating signal is known as **Pulse Width Modulation (PWM)**. It is also known as **Pulse duration modulation (PDM)**.

Basics of Pulse Width Modulation

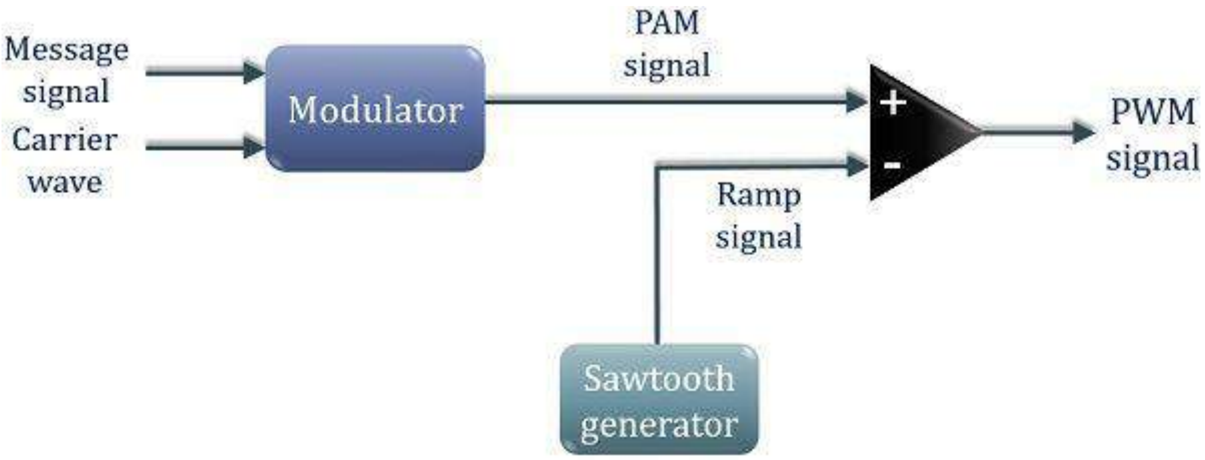
It is a type of **Pulse Time Modulation (PTM) technique** where the timing of the carrier pulse is varied according to the modulating signal.

In pulse duration modulation (PDM), the amplitude of the pulse is kept constant and only the variation in width is noticed. As the information component is present in width of the pulses. Thus, during signal transmission, the signal undergoes pulse width modulation. Due to constant amplitude property, it gets less affected by noise. However, during transmission channel noise introduces some variation in amplitude as it is additive in nature. But that is totally easy removable at the receiver by making use of limiter circuit.

As the width of the pulses contains information. Thus the noise factor does not cause much signal distortion. Hence the immunity to the noise of a PWM system is better than the PAM system.

Generation of PWM signal Waveform representation

The figure below shows the process of pulse width modulation. It is commonly known as an indirect method of PWM generation.



Generation of PWM signal

The message signal and the carrier waveform is fed to a modulator which generates PAM signal. This pulse amplitude modulated signal is fed to the non-inverting terminal of the comparator.

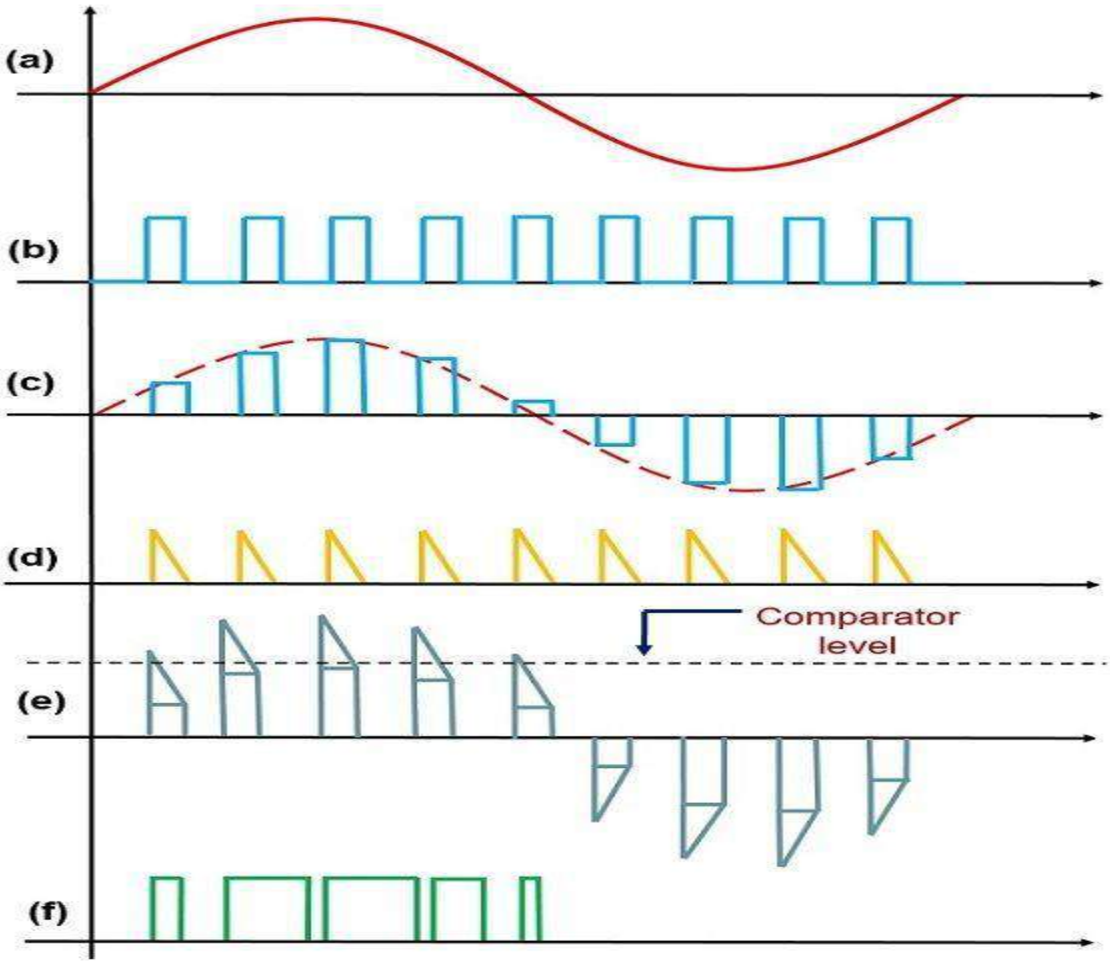
A ramp signal generated by the sawtooth generator is fed to the inverting terminal of the comparator.

These two signals are added and compared with the reference voltage of the comparator circuit. The level of the comparator is so adjusted to have the intersection of the reference with the slope of the waveform.

The PWM pulse begins with the leading edge of the ramp signal and the width of the pulse is determined by the comparator circuit.

The width of the PWM signal is proportional to the omitted portion of the ramp signal by the comparator level.

The figure below will help you to understand in a better way how PWM signal is generated by the comparator:



Here, the first image i.e., (a) shows the waveform of the sinusoidal modulating signal and the second one (b) shows the pulsed carrier. After modulation, a PAM signal is generated that is shown in (c). This PAM signal, when added with ramp signal shown in (d), is compared with the reference voltage of the comparator shown in figure (e).

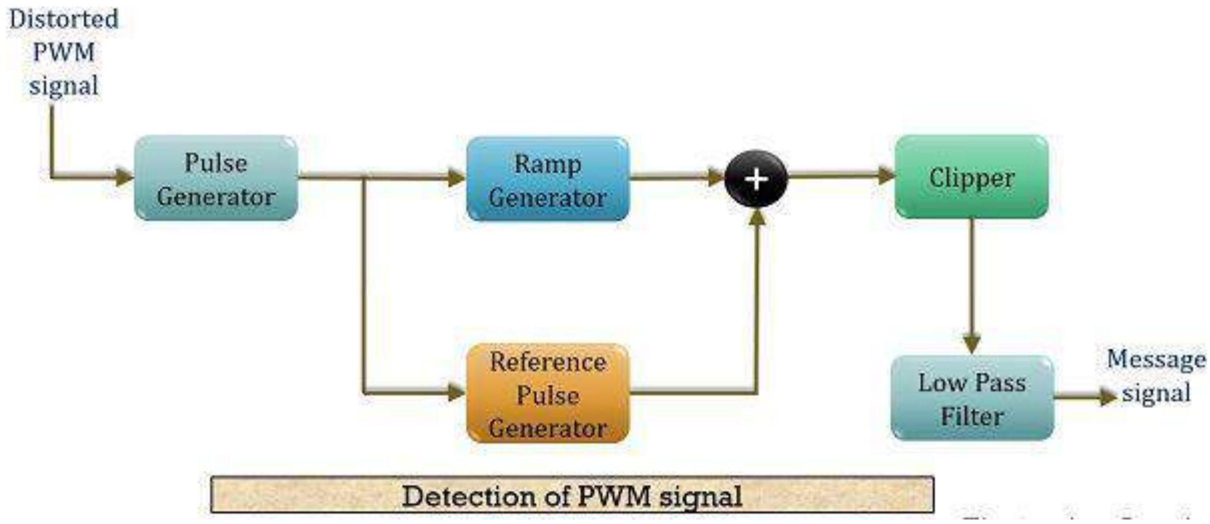
Lastly, figure (f) shows the PWM signal.

We have already mentioned that the width of the pulse is directly dependent on the portion of the waveform that lies above the comparator level.

This is how a pulse width modulated signal is generated.

Detection of PWM signal

The figure below shows the PWM detection circuit, that provides the original message signal from the modulated one.



As we know during signal transmission, some noise gets added to the PWM signal. So firstly to remove the noise introduced in the transmitted signal, the incoming signal is fed to a pulse generator. This regenerates the PWM signal.

This regenerated PWM pulse is then given to a reference pulse generator that generates pulses of constant amplitude along with constant width.

The regenerated pulses are also given to the ramp signal generator, that generates a ramp signal of constant slope, whose duration is similar to the pulse duration. Thus we have ramp signal height proportional to the PWM pulse width.

The constant amplitude pulses are then provided to a summation unit in order to get added with the ramp signal. The added output is then fed to a clipper, this clips off the signal up to its threshold value thereby generating a PAM signal at its output.

This PAM signal is then given to an LPF in order to generate the original message signal from the modulated one.

Advantages of Pulse Width Modulation

1. It is more immune to channel induced noise than PAM.
2. As noise adds to the amplitude thus the reconstruction of PWM signal from distorted PWM signal is somewhat easy.
3. The transmission and reception do not need to be synchronized.

Disadvantages of Pulse Width Modulation

1. Due to changing width of the pulses, variation in transmission power is also noticed.
2. Bandwidth requirement in case of PWM is somewhat larger than PAM.

Applications of Pulse Width Modulation

It is used in telecommunications, brightness controlling of light or speed controlling of fans etc.

Pulse Position Modulation

Pulse Position Modulation (PPM) is an analog modulating scheme in which 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 varies according to the instantaneous sampled value of the message signal.

Basics of Pulse Position Modulation

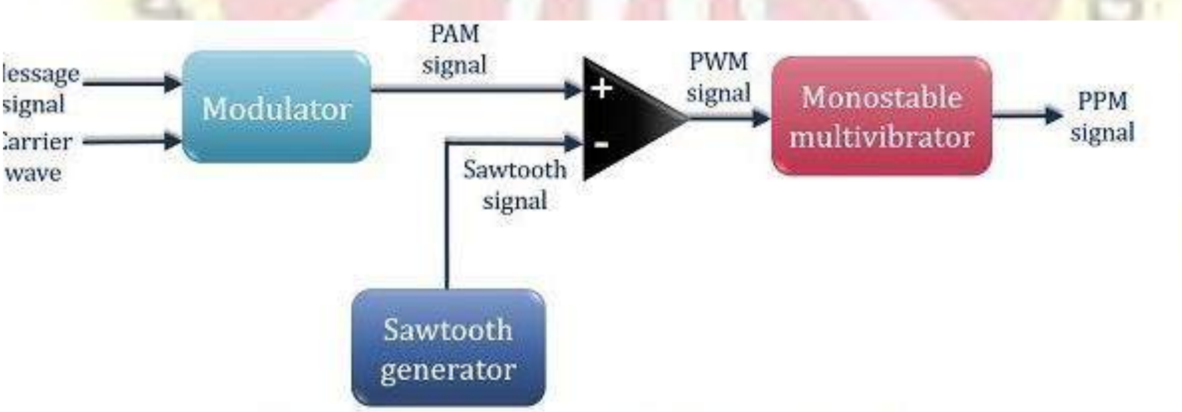
The information is transmitted with the varying position of the pulses in pulse position modulation.

The basic idea about the generation of a PPM waveform is that here, as the amplitude of the message signal increases, the pulse shifts according to the reference.

Block diagram for generation of PPM signal

As we have already discussed that a PPM signal can be easily generated by making use of a PWM signal. Thus, here we have assumed that a PWM signal is already generated at the output of the comparator and now we have to generate a PPM signal.

The figure below shows the block diagram for generating a PPM signal:



First, a PAM signal is produced with is further processed at the comparator in order to generate a PWM signal.

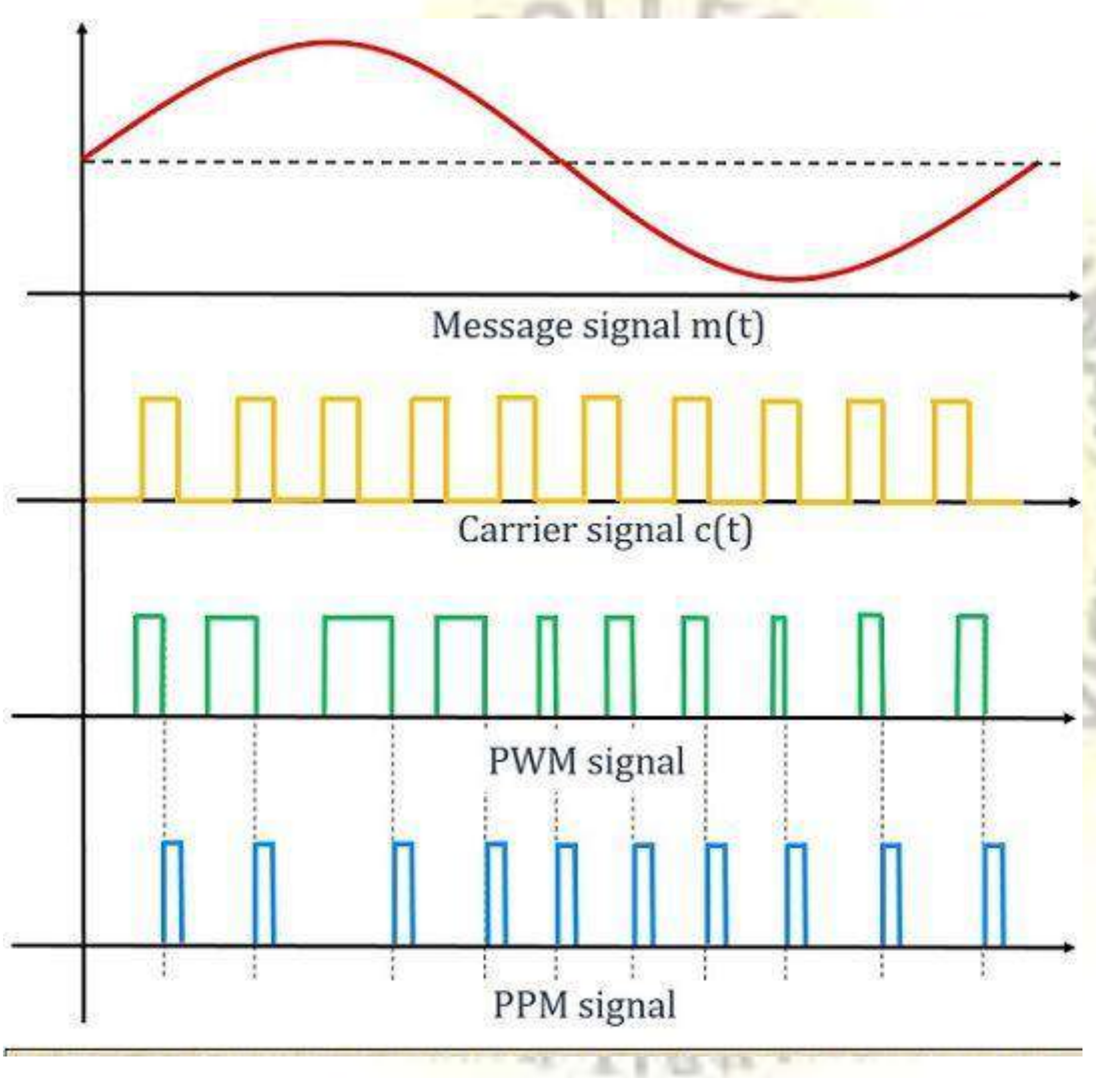
The output of the comparator is fed to a monostable multivibrator. It is negative edge triggered. Hence, with the trailing edge of the PWM signal, the output of the monostable goes high.

This is why a **pulse of PPM signal begins with the trailing edge of the PWM signal**.

It is to be noted in case of PPM that the duration for which the output will be high depends on the RC components of the multivibrator. This is the reason why a constant width pulse is obtained in case of the PPM signal.

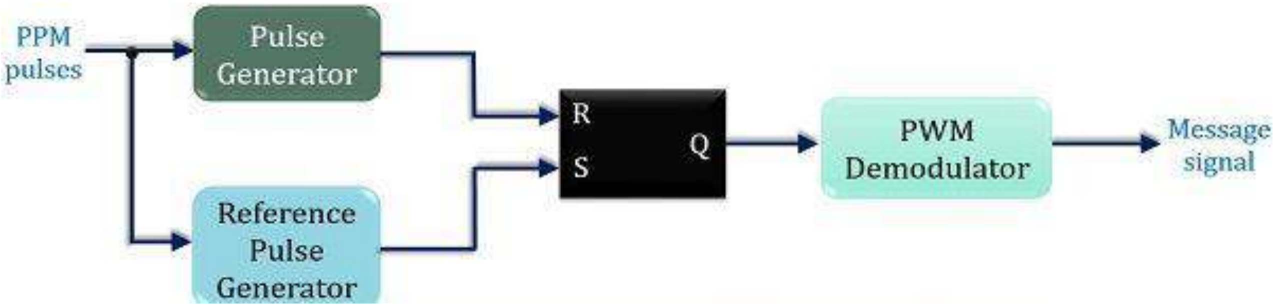
With the modulating signal, the trailing edge of PWM signal shifts, thus with that shift, the PPM pulses shows shifts in its position.

The figure below shows the **waveform representation of the PPM signal**:



. Detection (Demodulation) of PPM signal

The figure below shows the block diagram for the detection of a PPM signal at the receiver:



As we can see in the above figure that the demodulation circuit consists of a pulse generator, **SR flip-flop**, reference pulse generator and a PWM demodulator.

The PPM signal transmitted from the modulation circuit gets distorted by the noise during transmission. This distorted PPM signal reaches the demodulator circuit. The pulse generator employed in the circuit generates a pulsed waveform. This waveform is of fixed duration which is fed to the reset pin (R) of the SR flip-flop.

The reference pulse generator generates, reference pulse of a fixed period when transmitted PPM signal is applied to it. This reference pulse is used to set the flip-flop.

These **set and reset signals generate a PWM signal** at the output of the flip-flop. This PWM signal is then further processed in order to provide the original message signal.

Advantage

As the amplitude and width are constant, the power handled is also constant.

Disadvantage

The synchronization between transmitter and receiver is a must.

Comparison between PAM, PWM, and PPM

The comparison between the above modulation processes is presented in a single table.

PAM	**PWM**	**PPM**
Amplitude is varied	Width is varied	Position is varied
Bandwidth depends on the width of the pulse	Bandwidth depends on the rise time of the pulse	Bandwidth depends on the rise time of the pulse

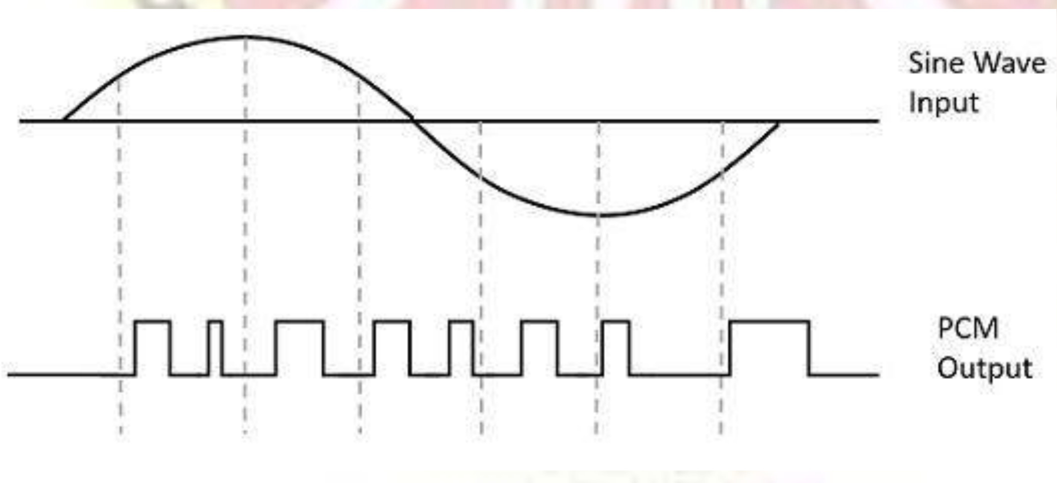
Instantaneous transmitter power varies with the amplitude of the pulses	Instantaneous transmitter power varies with the amplitude and width of the pulses	Instantaneous transmitter power remains constant with the width of the pulses
System complexity is high	System complexity is low	System complexity is low
Noise interference is high	Noise interference is low	Noise interference is low
It is similar to amplitude modulation	It is similar to frequency modulation	It is similar to phase modulati

Digital Modulation

Digital Modulation, which falls under the classification of pulse modulation. Digital modulation has Pulse Code Modulation (PCM) as the main classification. It further gets processed to delta modulation and ADM.

Pulse Code Modulation

A signal is Pulse Code modulated to convert its analog information into a binary sequence, i.e., 1s and 0s. The output of a **Pulse Code Modulation (PCM)** will resemble a binary sequence. The following figure shows an example of PCM output with respect to instantaneous values of a given sine wave.



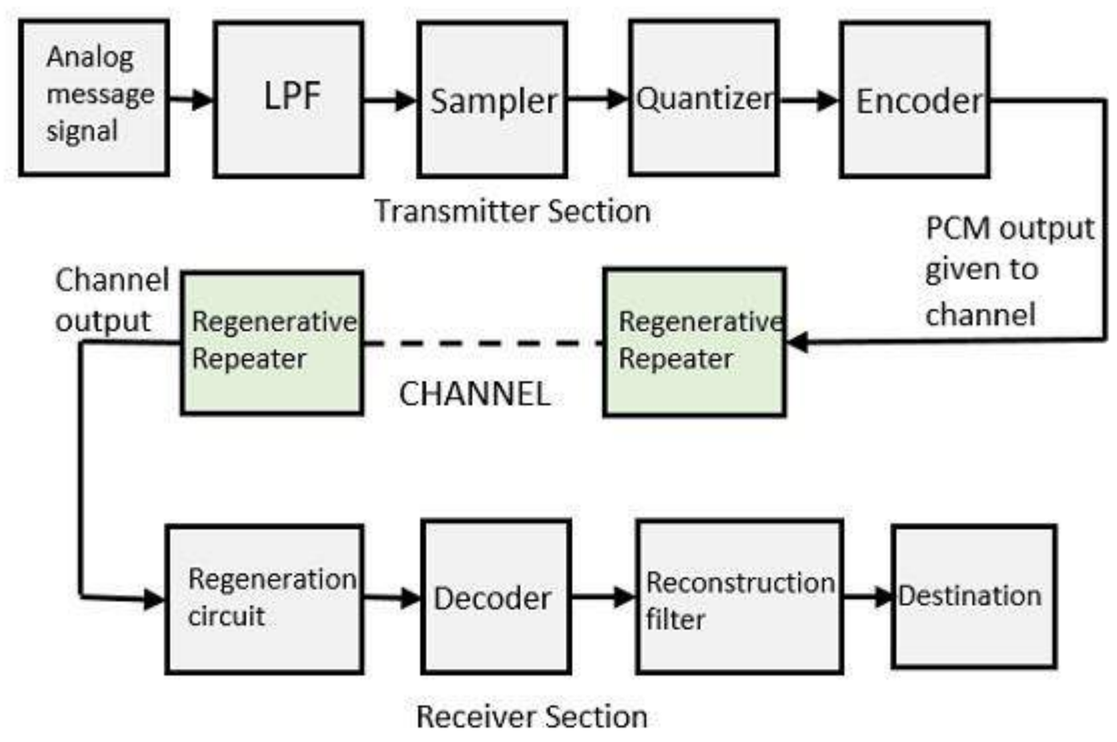
Instead of a pulse train, PCM produces a series of numbers or digits, and hence this process is called as digital. Each one of these digits, though in binary code, represent the approximate amplitude of the signal sample at that instant.

In Pulse Code Modulation, the message signal is represented by a sequence of coded pulses. This message signal is achieved by representing the signal in discrete form in both time and amplitude.

Basic Elements of PCM

The transmitter section of a Pulse Code Modulator circuit consists of **Sampling, Quantizing** and **Encoding**, which are performed in the **analog-to-digital converter** section. The low pass filter prior to sampling prevents aliasing of the message signal.

The basic operations in the receiver section are **regeneration of impaired signals, decoding,** and **reconstruction** of the quantized pulse train. The following figure is the block diagram of PCM which represents the basic elements of both the transmitter and the receiver sections.



Low Pass Filter (LPF)

This filter eliminates the high frequency components present in the input analog signal which is greater than the highest frequency of the message signal, to avoid aliasing of the message signal.

Sampler

This is the circuit which uses the technique that helps to collect the sample data at instantaneous values of the message signal, so as to reconstruct the original signal. The sampling rate must be greater than twice the highest frequency component **W** of the message signal, in accordance with the sampling theorem. $\mathbf{f_s \geq 2f_m}$

Quantizer

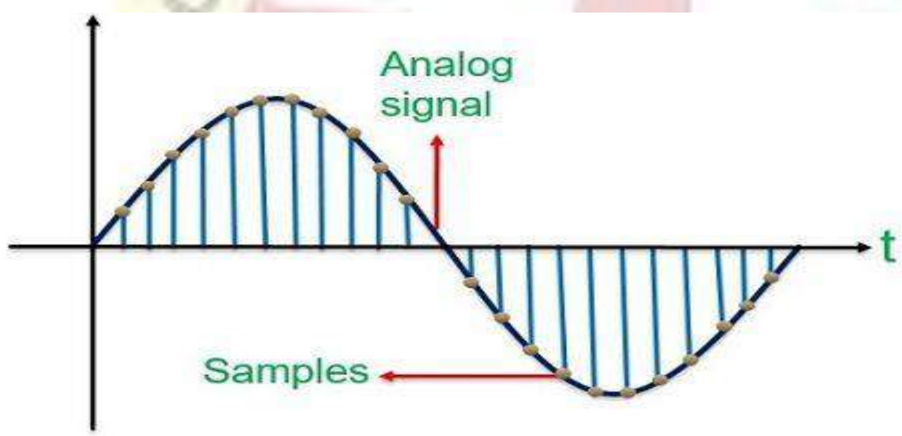
- **Quantizer**: A quantizer is a unit that rounds off each sample to the nearest discrete level. The sampler provides a continuous range signal and hence still an analog one. The quantizer performs the approximation of each sample thus assigning it a particular discrete level.

As it basically rounds off the value to a certain level this shows some variation by the actual amount. Thus we can say, quantizing a signal introduces some distortion or noise into it. This is known as **quantization error**.

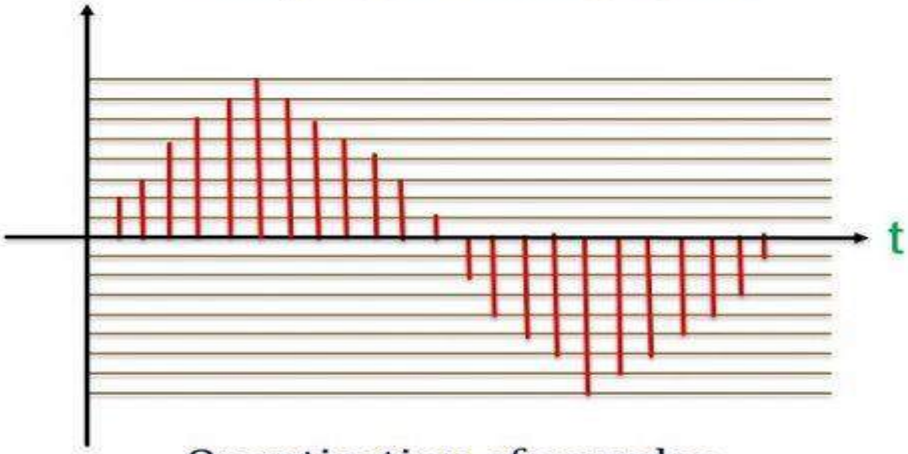
This noise factor is somewhat better than the channel noise as it is controllable.

For a **low signal level, the quantization error is high** i.e., bad SNR. But, for a **high signal level, the quantization error is low** providing good SNR.

The figure below shows the sampling of analog signal and further quantization of the samples



Sampling of analog signal



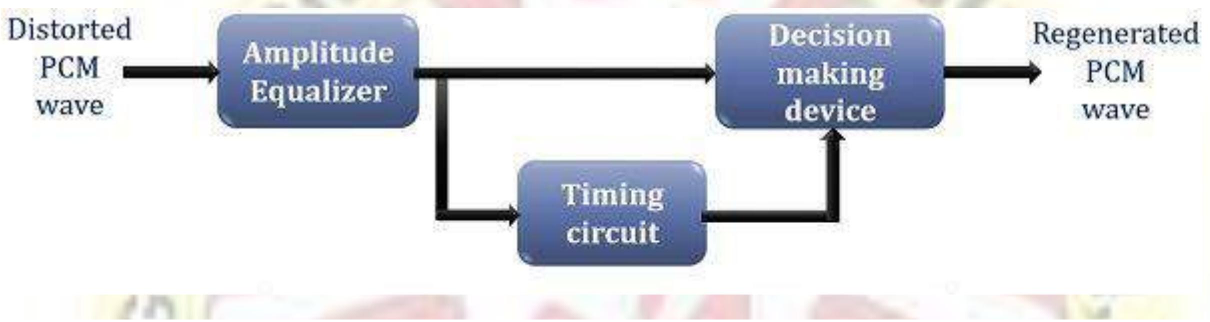
Quantization of samples

Encoder

The digitization of analog signal is done by the encoder. It designates each quantized level by a binary code. The sampling done here is the sample-and-hold process. These three sections will act as an analog to the digital converter. Encoding minimizes the bandwidth used.

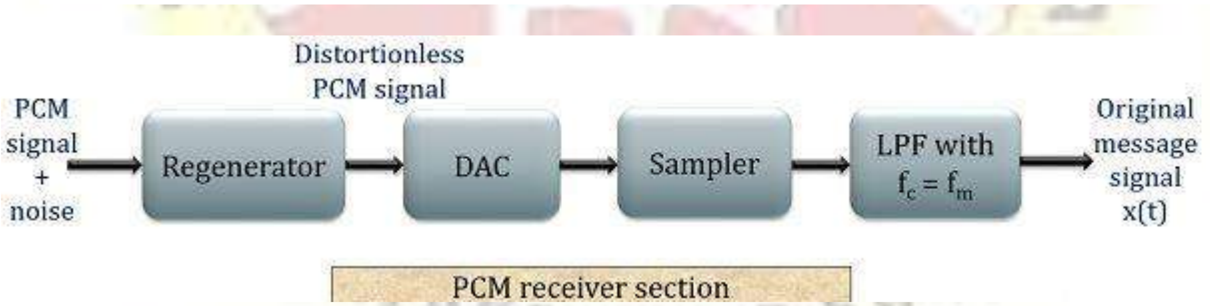
Regenerative Repeater

The output of the channel has one regenerative repeater circuit to compensate the signal loss and reconstruct the signal. It also increases the strength of the signal. The PCM signal when provided to the regenerative repeater, the equalizer circuit at the beginning performs the reshaping of the distorted signal. At the same time, the timing circuit generates a pulse train that is a derivative of input PCM pulses.



This pulse train is then utilized by the decision-making device in order to sample the PCM pulses. This sampling is done at the instant where maximum SNR can be achieved. In this way, the decision-making device generates the distortionless PCM wave.

PCM Receiver



Regenerator: A regenerative repeater is placed at the receiving end also so as to have an exact PCM transmitted signal. Here, also the regenerator works in a similar manner as that when employed in the transmission path. It eliminates the channel induced noise and reshapes the pulse.

DAC and Sampler: Digital to analog converter performs the conversion of digital signal again into its analog form by making use of the sampler. As the actual message signal was analog thus at the receiver end there is a necessity to again convert it into its original form.

LPF: The sampler generates analog signal but that is not the original message signal. Thus, the output of the sampler is fed to the LPF having cutoff frequency fm. This is sometimes termed as the reconstruction filter that produces the original message signal.

Hence, the Pulse Code Modulator circuit digitizes the analog signal given, codes it, and samples it. It then transmits in an analog form. This whole process is repeated in a reverse pattern to obtain the original signal.

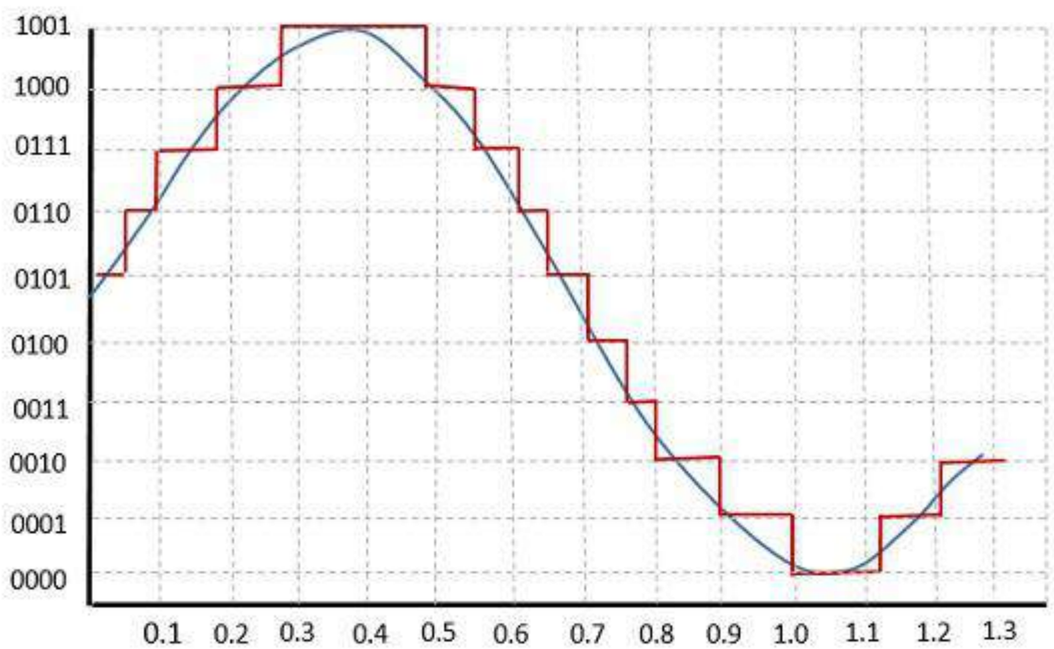
here are few modulation techniques which are followed to construct a PCM signal. These techniques like **sampling, quantization,** and **companding** help to create an effective PCM signal, which can exactly reproduce the original signal.

Quantization

The digitization of analog signals involves the rounding off of the values which are approximately equal to the analog values. The method of sampling chooses few points on the analog signal and then these points are joined to round off the value to a near stabilized value. Such a process is called as **Quantization**.

The quantizing of an analog signal is done by discretizing the signal with a number of quantization levels. Quantization is representing the sampled values of the amplitude by a finite set of levels, which means converting a **continuous-amplitude sample** into a **discrete-time signal**.

The following figure shows how an analog signal gets quantized. The blue line represents analog signal while the red one represents the quantized signal.



Both sampling and quantization results in the loss of information. The quality of a Quantizer output depends upon the number of quantization levels used. The discrete amplitudes of the quantized output are called as **representation levels** or **reconstruction levels**. The spacing between two adjacent representation levels is called a **quantum** or **step-size**.

Companding in PCM

The word **Companding** is a combination of **Com**pressing and Ex**panding**, which means that it does both. This is a non-linear technique used in PCM which compresses the data at the transmitter and expands the same data at the receiver. The effects of noise and crosstalk are reduced by using this technique.

There are two types of Companding techniques.

A-law Companding Technique

- Uniform quantization is achieved at **A = 1**, where the characteristic curve is linear and there is no compression.
- A-law has mid-rise at the origin. Hence, it contains a non-zero value.
- A-law companding is used for PCM telephone systems.
- A-law is used in many parts of the world.

μ-law Companding Technique

- Uniform quantization is achieved at **μ = 0**, where the characteristic curve is linear and there is no compression.
- μ-law has mid-tread at the origin. Hence, it contains a zero value.
- μ-law companding is used for speech and music signals.
- μ-law is used in North America and Japan.
- ***Differential PCM***
- The samples that are highly correlated, when encoded by PCM technique, leave redundant information behind. To process this redundant information and to have a better output, it is a wise decision to take predicted sampled values, assumed from its previous outputs and summarize them with the quantized values.
- Such a process is named as **Differential PCM** technique.

The sampling rate of a signal should be higher than the Nyquist rate, to achieve better sampling. If this sampling interval in a Differential PCM (DPCM) is reduced considerably, the sample-to-sample amplitude difference is very small, as if the difference is **1-bit quantization**, then the step-size is very small i.e., **Δ** (delta).

Advantages of PCM

1. Immune to channel induced noise and distortion.
2. Repeaters can be employed along the transmitting channel.
3. Encoders allow secured data transmission.

4. It ensures uniform transmission quality.

Disadvantages of PCM

1. Pulse code modulation increases the transmission bandwidth.
2. A PCM system is somewhat more complex than another system

What is Delta Modulation?

A modulation technique that converts or encodes message signal into a binary bit stream is known as Delta Modulation. Here only 1 bit is used to encode 1 voltage level thus, the technique allows transmission of only 1 bit per sample.

As PCM has the property of converting message signal directly into a sequence of a binary coded pulse, this resultantly increases the bandwidth requirement of the system. So, in order to remove the drawbacks of PCM, delta modulation is used.

Features of Delta Modulation

- An over-sampled input is taken to make full use of a signal correlation.
- The quantization design is simple.
- The input sequence is much higher than Nyquist rate.
- The quality is moderate.
- The design of the modulator and the demodulator is simple.
- The stair-case approximation of output waveform.
- The step-size is very small, i.e., **Δ** (delta).
- The bit rate can be decided by the user.
- It requires simpler implementation.

Delta Modulation is a simplified form of DPCM technique, also viewed as 1-bit DPCM scheme. As the sampling interval is reduced, the signal correlation will be higher.

Operating principle of Delta Modulation

The operating principle of DM is such that, a comparison between present and previously sampled value is performed, the difference of which decides the increment or decrement in the transmitted values.

Simply put, when the two sample values are compared, either we get difference having a positive polarity or negative polarity.

If the difference polarity is positive, then the step of the signal denoted by Δ is increased by 1. As against in case when difference polarity is negative then step of the signal is decreased i.e., reduction in Δ.

When **+Δ is noticed i.e., increase in step size, then 1 is transmitted**. However, in the case of **–Δ i.e., decrease in step size, 0 is transmitted**.

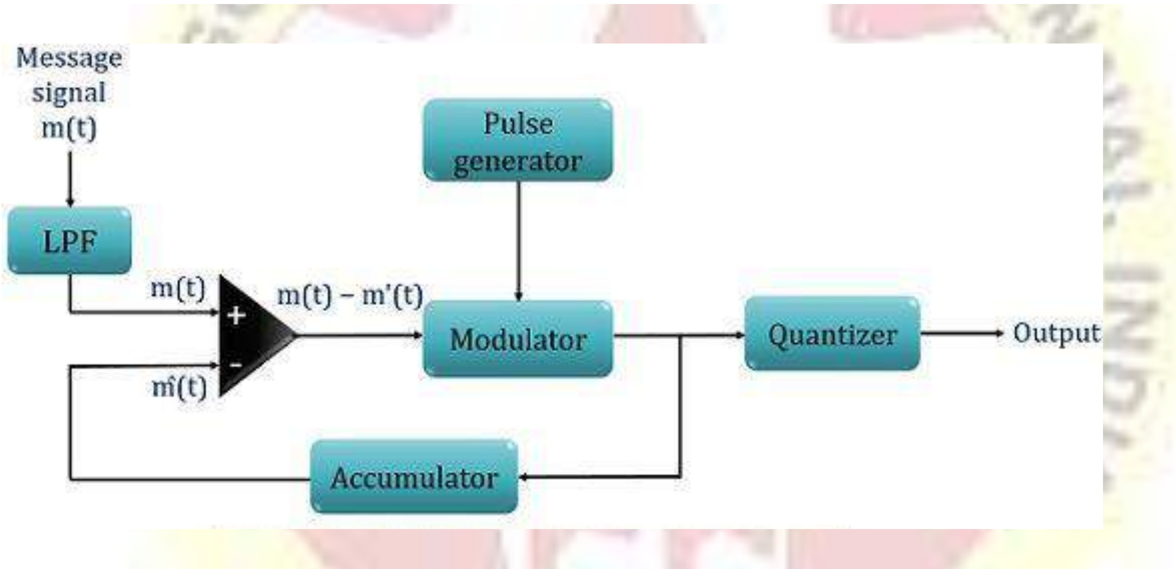
Hence, **allowing only a single binary bit to get transmitted for each sample**.

Block diagram for Delta Modulation

Let us first understand the generation of delta modulated signal.

Generation of delta modulated signal

The block diagram for the generation delta modulated signal is shown below:



As we can see the above figure consists of an LPF, a comparator, a product modulator along with pulse generator and quantizer. Here, a feedback path is also provided to the circuit, where the output of modulator acts as input to the comparator.

The message signal that is to be transmitted is fed to a low pass filter that passes the low-frequency component and eliminates the high-frequency component. It is also referred to as **aliasing filter**.

The output of LPF is then given to a comparator unit, which compares the message signal m(t) with an arbitrary signal m'(t) for the first time. The comparator after comparing 2 signals generates the difference between the tw

The difference can be of either positive polarity or negative polarity. This depends on message and arbitrary signals that are getting subtracted.

This difference signal now acts as input to the product modulator. Another input to the modulator is a pulse signal generated by the pulse generator. These two signals are multiplied in the modulator.

The output of the modulator is a pulsed signal whose pulses will be of equal magnitude having polarity either positive and negative.

The polarity totally depends on the output of the comparator. The output of the modulator is given to quantizer. The quantizer generates the output in the form of steps.

If positive magnitude pulse is provided to the quantizer as its input then quantizer performs increment by 1 step size, Δ. It is very easy to understand that positive pulse at the output of the modulator shows that message signal is greater than the arbitrary signal. Thus quantizer increases Δ by 1.

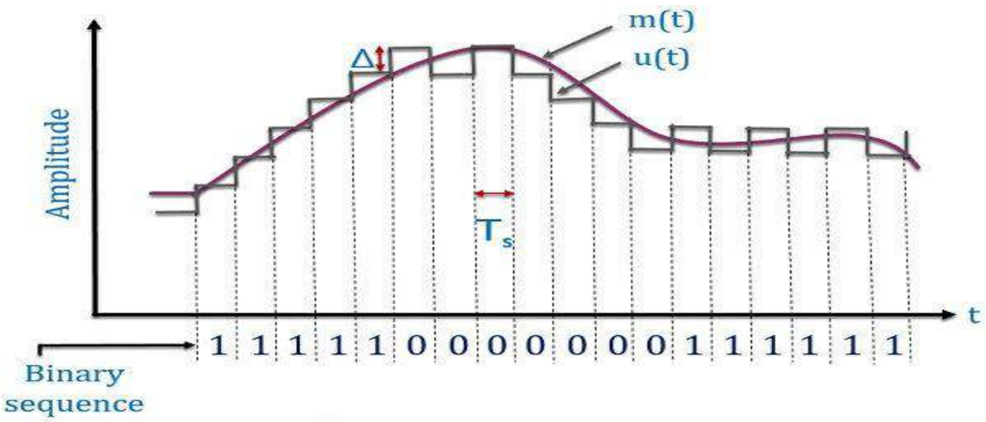
Similarly, in the case of negative magnitude pulse, the step size gets decreased by 1. This is so because m'(t) exceeds m(t), thereby generating a pulse of negative polarity. Thus, quantizer decreases Δ by 1.

The output of the modulator at the same time, through a feedback path, is provided to the accumulator.

An accumulator is nothing but a device that stores the signal for further operation. The output of the accumulator now behaves like the second input of the comparator. Thus, we say that the **present sample value is compared with the previous one** for further operation. Hence the process repeats in such a manner.In the end, depending on the staircase signal if the step size is +Δ then binary 1 is transmitted and if it is –Δ then binary 0 is transmitted.

Waveform Representation of Delta Modulation

The figure below shows the delta modulation waveform:

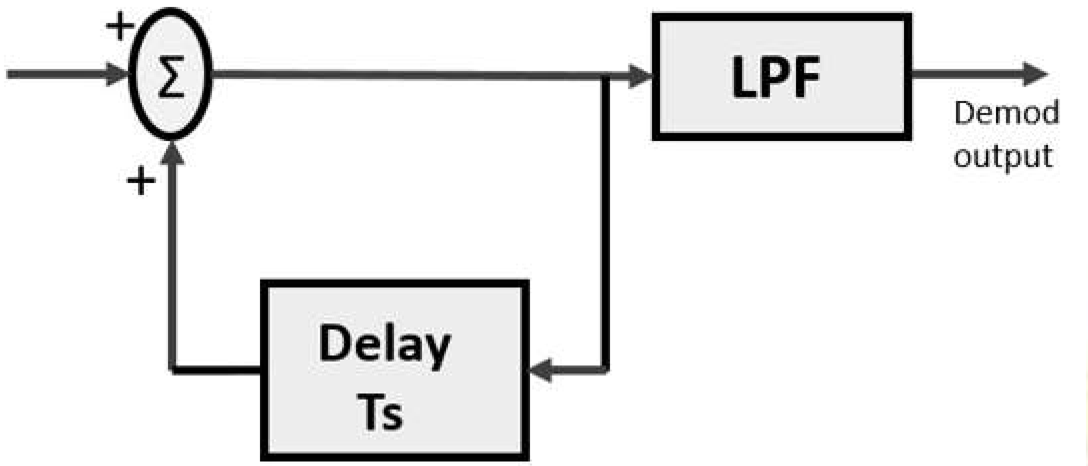


Here, the analog input signal is m(t) and the quantized signal is denoted by u(t). The binary sequence according to the step size that is actually transmitted is shown at the bottom of the figure shown above.

Delta Demodulator

The delta demodulator comprises of a low pass filter, a summer, and a delay circuit. The predictor circuit is eliminated here and hence no assumed input is given to the demodulator.

Following is the block diagram for delta demodulator.



Low pass filter is used for many reasons, but the prominent one is noise elimination for out-of-band signals. The step-size error that may occur at the transmitter is called **granular noise**, which is eliminated here. If there is no noise present, then the modulator output equals the demodulator input.

Advantages of DM over DPCM

- 1-bit quantizer
- Very easy design of modulator & demodulator

However, there exists some **noise in DM** and following are the types of noise.

- Slope Over load distortion (when Δ is small)
- Granular noise (when Δ is large)

Adaptive Delta Modulation

In digital modulation, we

come across certain problems in determining the step-size, which influences the quality of the output wave.

The larger step-size is needed in the steep slope of modulating signal and a smaller stepsize is needed where the message has a small slope. As a result, the minute details get missed. Hence, it would be better if we can control the adjustment of step-size, according to our requirement in order to obtain the sampling in a desired fashion. This is the concept of **Adaptive Delta Modulation (ADM)**.

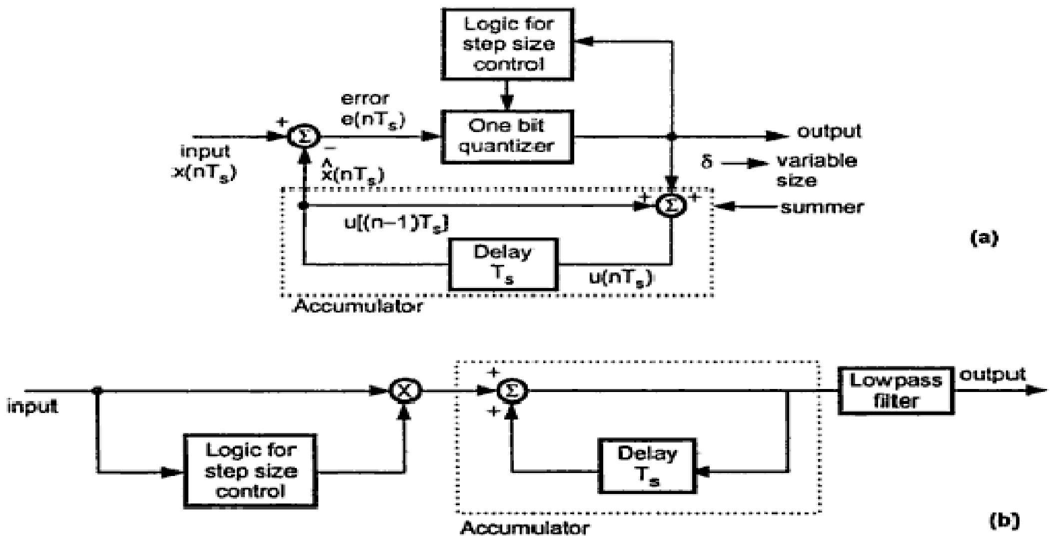


Fig. Adaptive delta modulator (a) Transmitter (b) Receiver

For example if one bit quantizer output is high (1), then step size may be doubled for next sample. If one bit quantizer output is low, then step size may be reduced by one step. Fig. shows the waveforms of adaptive delta modulator and sequence of bits transmitted.

In the receiver of adaptive delta modulator shown in Fig. (b) the first part generates the step size from each incoming bit. Exactly the same process is followed as that in transmitter. The previous input and present input decides the step size. It is then given to an accumulator which builds up staircase waveform. The low-pass filter then smoothens out the staircase waveform to reconstruct the smooth signal.

Digital Modulation provides more information capacity, high data security, quicker system availability with great quality communication. Hence, digital modulation techniques have a greater demand, for their capacity to convey larger amounts of data than analog ones.

There are many types of digital modulation techniques and we can even use a combination of these techniques as well. In this chapter, we will be discussing the most prominent digital modulation techniques.

UNIT V

5.0DIGITAL COMMUNICATION

Digital communication is defined as the method to transfer information from one place to another using digital signals ie. 0 and 1.

In digital communication system, the signal used to transfer information should be discrete in time and discrete in amplitude. They are also called a digital signal. For most of the time keyboard in the source of digital signal such as o and 1. But if the source is analog, first, we have to convert it into a digital signal using analog to digital converter.

ElementsofDigitalCommunicationSystems:

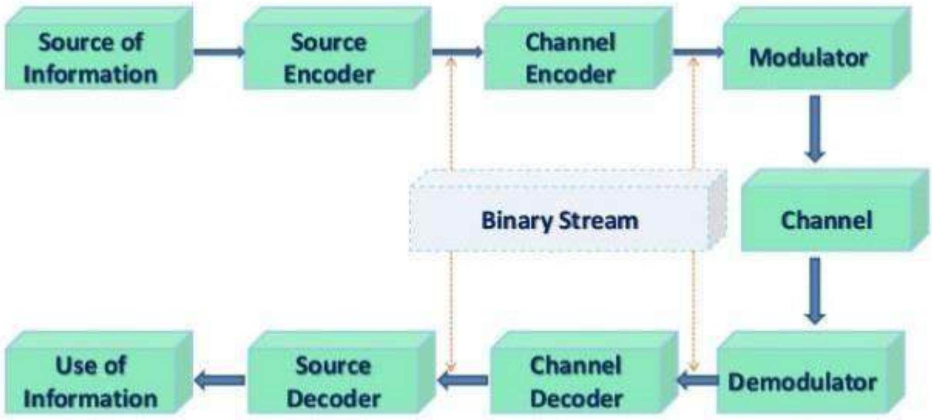


Fig.1ElementsofDigitalCommunicationSystems

1. InformationSourceandInputTransducer:

The source of information can be analog or digital, e.g. analog: audio or videosignal, digital: like teletype signal. In digital communication the signal produced bythis source is converted into digital signal which consists of 1′s and 0′s. For this weneedasourceencoder.

2. SourceEncoder:

In digital communication we convert the signal from source into digital signalas mentioned above. The point to remember is we should like to use as few binarydigitsaspossibletorepresentthesignal.Insuchawaythisefficientrepresentationof the source output results in little or no redundancy. This sequence of binary digitsiscalled***informationsequence.***

Source Encoding or Data Compression: the process of efficiently convertingthe output of whether analog or digital source into a sequence of binary digits isknownas sourceencoding.

3. ChannelEncoder:

Theinformationsequenceispassedthroughthechannelencoder.Thepurposeofthec hannelencoderistointroduce,incontrolledmanner,someredundancy in the binary information sequence that can be used at the receiver toovercome the effects of noise and interference encountered in the transmission onthesignalthroughthechannel.

For example take k bits of the information sequence and map that k bits tounique n bit sequence called code word. The amount of redundancy introduced ismeasured by the ratio *n/k* and the reciprocal of this ratio (k/n) is known as *rate ofcodeorcoderate.*

4. DigitalModulator:

The binary sequence is passed to digital modulator which in turns convert thesequence into electric signals so that we can transmit them on channel (we will seechannel later). The digital modulator maps the binary sequences into signal waveforms , for example if we represent 1 by sin x and 0 by cos x then we will transmit sinxfor1andcos xfor 0. (acasesimilartoBPSK)

5. Channel:

Thecommunicationchannelisthephysicalmediumthatisusedfortransmitting signals from transmitter to receiver. In wireless system, this channelconsists of atmosphere , for traditional telephony, this channel is wired , there areopticalchannels,under water acousticchannels etc.We further discriminate thischannelsonthebasisoftheirpropertyandcharacteristics,likeAWGNchanneletc.

6. DigitalDemodulator:

Thedigitaldemodulatorprocessesthechannelcorruptedtransmittedwaveform and reduces the waveform to the sequence of numbers that representsestimates ofthetransmitteddatasymbols.

7. ChannelDecoder:

This sequence of numbers then passed through the channel decoder whichattempts to reconstruct the original information sequence from the knowledgeofthe code used by the channel encoder and the redundancy contained in the receiveddata

8. SourceDecoder:

At the end, if an analog signal is desired then source decoder tries to decodethe sequence from the knowledge of the encoding algorithm. And which results intheapproximatereplicaoftheinputatthetransmitterend.

9 .outputTransducer:

Finallywegetthedesiredsignalindesired formatanalogordigital.

Advantagesofdigitalcommunication:

- Can withstand channel noise and distortion much better as long as the noise and the distortion are within limits.
- Regenerative repeaters prevent accumulation of noise along the path.
- Digital hardware implementation is flexible.
- Digital signals can be coded to yield extremely low error rates, high fidelity and well as privacy.
- Digital communication is inherently more efficient than analog in realizing the exchange of SNR for bandwidth.
- It is easier and more efficient to multiplex several digital signals.
- Digital signal storage is relatively easy and inexpensive.
- Reproduction with digital messages is extremely reliable without deterioration.
- The cost of digital hardware continues to halve every two or three years, while performance or capacity doubles over the same time period.

Disadvantages

- TDM digital transmission is not compatible with the FDM
- A Digital system requires large bandwidth.

5.1 MULTIPLEXING

It has been observed that most of the individual data-communicating devices typicallyrequiremodestdatarate.But,communicationmediausuallyhavemuchhigh erbandwidth. As a consequence, two communicating stations do not utilize the full capacityofadatalink.Moreover,whenmanynodescompetetoaccessthenetwork,so meefficient techniques for utilizing the data link are very essential. When the bandwidth of amediumisgreaterthanindividualsignalstobetransmittedthroughthechannel,amed ium can be shared by more than one channel of signals.The process of making themosteffectiveuseoftheavailablechannelcapacityiscalled**Multiplexing**.Foreffi ciency,thechannelcapacitycanbesharedamonganumberofcommunicatingstations just like a large water pipe can carry water to several separate houses at once.Most common use of multiplexing is in long-haul communication using coaxial cable,microwaveand optical fibre.

Figure5.1depictsthefunctioningofmultiplexingfunctionsingeneral.The**multiplexer** is connected to the **demultiplexer** by a single data link. The multiplexercombines (multiplexes) data from these 'n' input lines and transmits them through thehigh capacity data link, which is being demultiplexed at the other end and is delivered tothe appropriate output lines. Thus, **Multiplexing** can also be defined as a technique thatallowssimultaneous transmissionofmultiplesignalsacrossasingledatalink.

Figure 5.1

Basic concept of multiplexingMultiplexingtechniquescanbecategorizedintothefollowingthreetypes:

•

Frequency-division multiplexing (FDM)*:** It is most popular and is used extensivelyin radio and TV transmission. Here the frequency spectrum is divided into severallogicalchannels,givingeachuserexclusivepossessionofaparticularfrequencyband.- ***division Multiplexing (TDM): It is also called synchronous TDM, which iscommonlyusedformultiplexingdigitizedvoicestream.Theuserstaketurnsusingtheentirecha nnelforshortburstoftime.

- ***Statistical TDM:*** This is also called asynchronous TDM, which simply improves onthe efficiency of synchronous TDM.

5.2Frequency-DivisionMultiplexing(FDM)

In frequency division multiplexing, the available bandwidth of a single physical mediumis subdivided into several independent frequency channels. Independent message signalsare translated into different frequency bands using modulation techniques, which arecombined by a linear summing circuit in the multiplexer, to a composite signal. Theresulting signal is then transmitted along the single channel by electromagnetic means asshown in Fig. 5.2 Basic approach is to divide the available bandwidth of a singlephysicalmediumintoanumberofsmaller,independentfrequencychannels.Usingmodulation, independent message signals are translated into different frequency bands.All the modulated signals are combined in a linear summing circuit to form a compositesignal for transmission. The carriers used to modulate the individual message signals arecalled*sub-carriers*,shownasf_1, f_2, …,f_n in Fig5.2 (a).

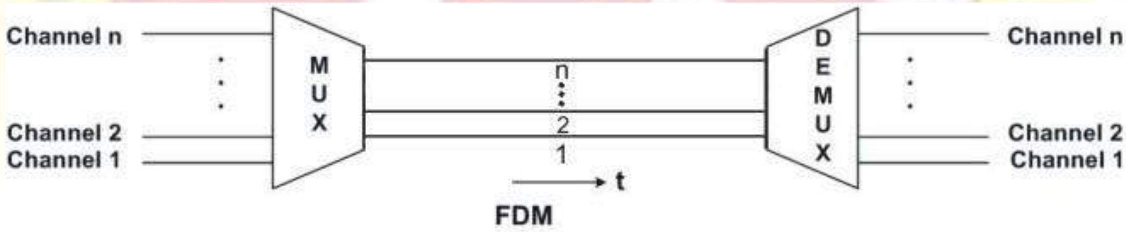


Figure5.2 Basic concept of FDM

At the receiving end the signal is applied to a bank of band-pass filters, which separatesindividual frequency channels. The band pass filter outputs are then demodulated anddistributedto differentoutput channelsasshownin Fig.5.2(b).

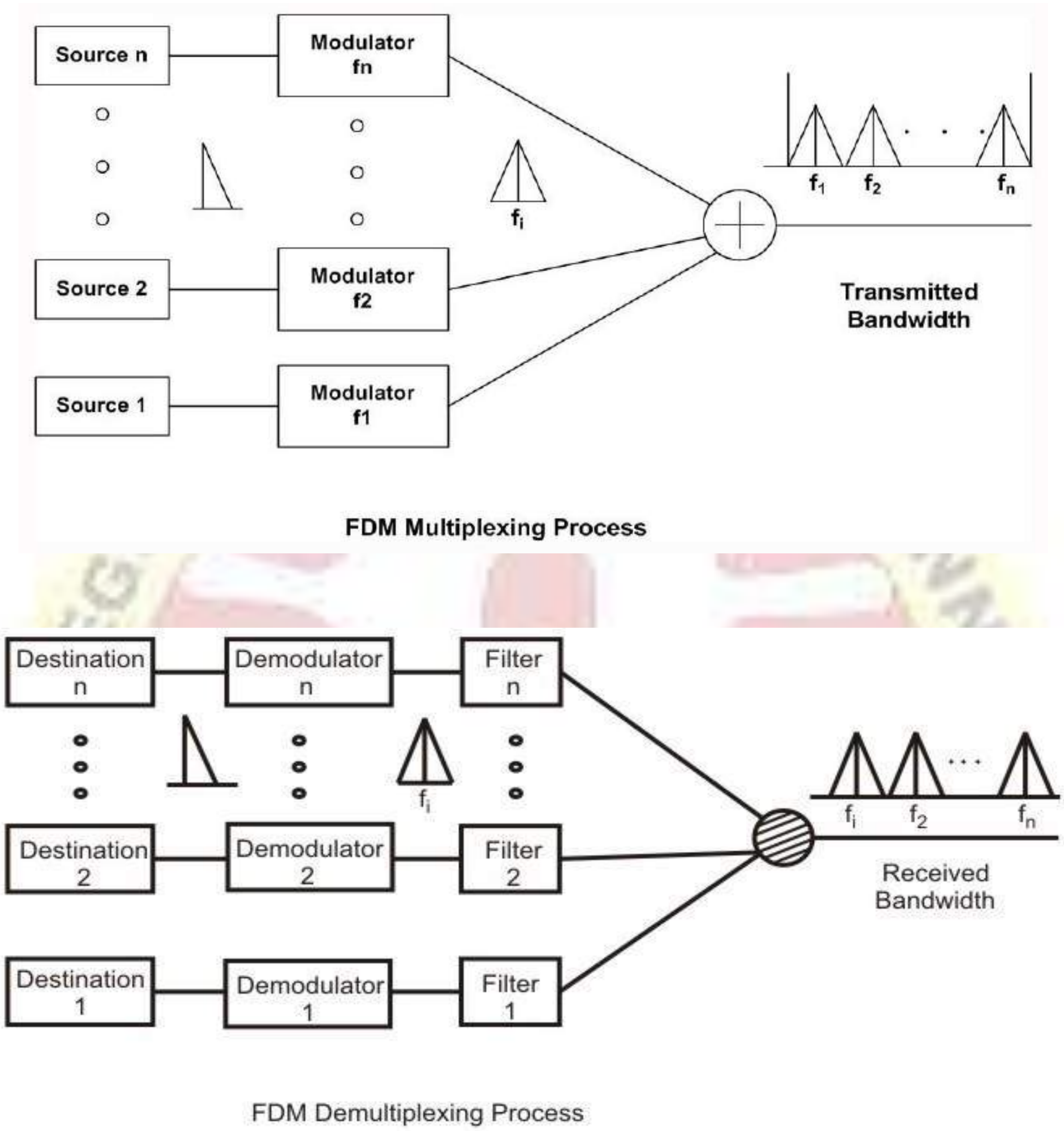


Figure5.2(a)FDMmultiplexingprocess,(b)FDMdemultiplexingprocess

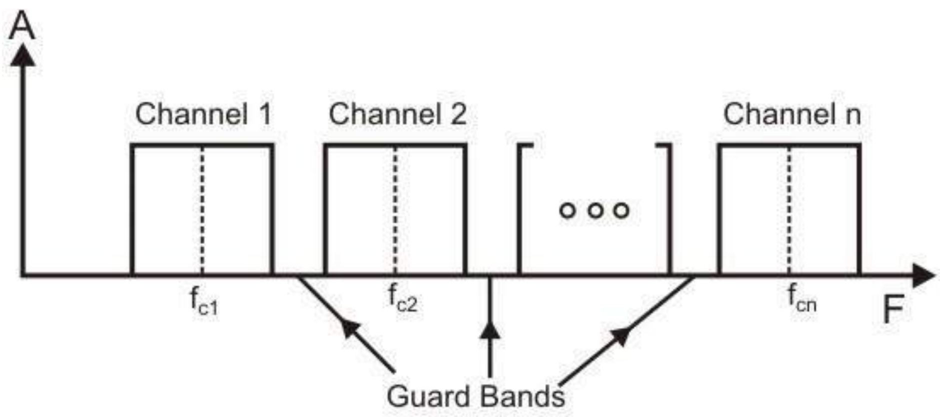


Figure5.3 Useof guard bandsin FDM

If the channels are very close to one other, it leads to inter-channel cross talk.Channelsmustbeseparatedbystripsofunusedbandwidthtopreventinter-channelcrosstalk.These unused channels between each successive channel are known as **guard band**s asshown in Fig. 5.3

FDMarecommonlyusedinradiobroadcastsandTVnetworks.Since,thefrequencyband used for voice transmission in a telephone network is 4000 Hz, for a particular cableof 48 KHz bandwidth, in the 70 to 108 KHz range, twelve separate 4 KHz sub channelscould be used for transmitting twelve different messages simultaneously. Each radioand TV station, in a certain broadcast area, is allotted a specific broadcast frequency, sothat independent channels can be sent simultaneously in different broadcast area. Forexample, the AM radio uses 540 to 1600 KHz frequency bands while the FM radio uses88 to 108 MHz frequency bands.

5.3Time-DivisionMultiplexing(TDM)

In frequency division multiplexing, all signals operate at the same time with differentfrequencies, but in Time-division multiplexing all signals operate with same frequency atdifferent times. This is a base band transmission system, where an electronic commutatorsequentially samples all data source and combines them to form a composite base bandsignal, which travels through the media and is being demultiplexed into appropriateindependent message signals by the corresponding commutator at the receiving end. Theincoming data from each source are briefly buffered. Each buffer is typically one bit orone character in length. The buffers are scanned sequentially to form a composite datastream. The scan operation is sufficiently rapid so that each buffer is emptied before moredata can arrive. Composite data rate must be at least equal to the sum of the individualdata rates. The composite signal can be transmitted directly or through a modem. Themultiplexingoperation isshown in Figure below

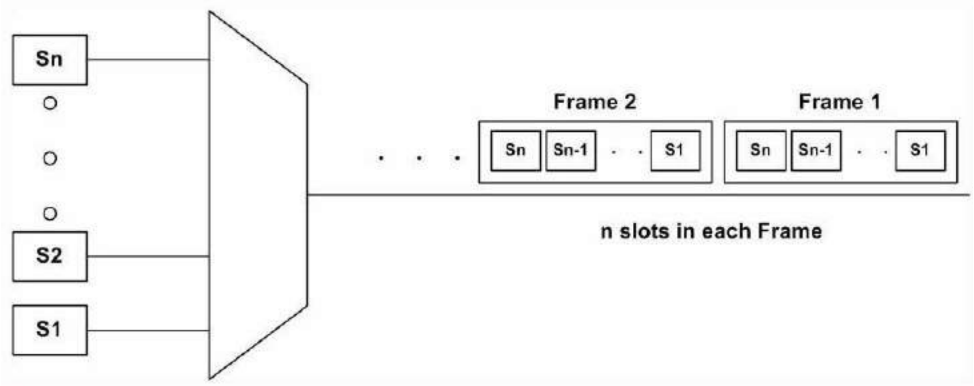
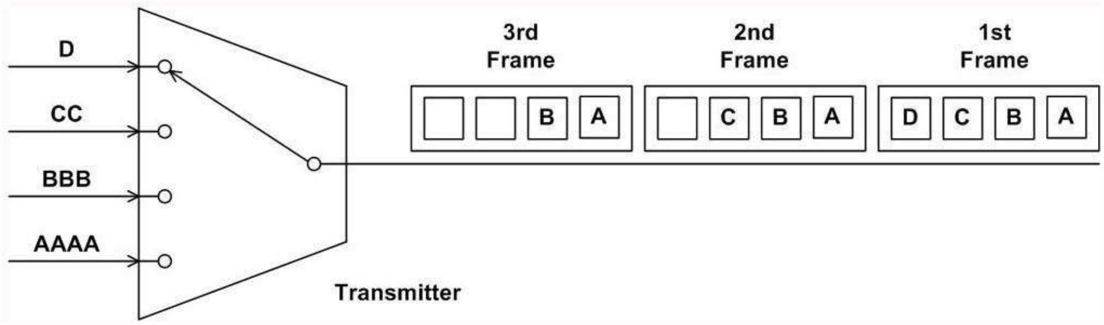


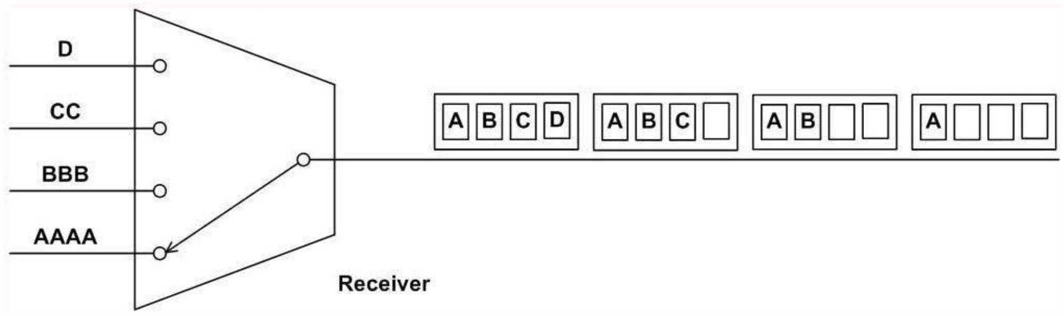
Figure5.5Timedivisionmultiplexingoperation

AsshownintheFig5.5thecompositesignalhassome*deadspace*betweenthesuccessive sampled pulses, which is essential to prevent interchannel cross talks. Alongwith the sampled pulses, one synchronizing pulse is sent in each cycle. These data pulsesalong with the control information form a *frame*.Each of these frames contain a cycle oftime slots and in each frame, one or more slots are dedicated to each data source. Themaximum bandwidth (data rate) of a TDM system should be at least equal to the samedata rate of the sources.

Synchronous TDM is called synchronous mainly because each time slot is pre assigned toafixedsource.Thetimeslotsaretransmittedirrespectiveofwhetherthesourceshaveany data to send or not. Hence, for the sake of simplicity of implementation, channelcapacity is wasted. Although fixed assignment is used TDM, devices can handle sourcesof different data rates. This is done by assigning fewer slots per cycle to the slower input devices than the faster devices Both multiplexing and demultiplexing operation for synchronous.

TDM are shown in Figure below





MultiplexinganddemultiplexinginsynchronousTDM

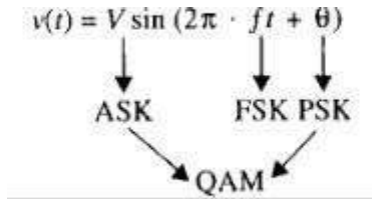
5.4 DIGITALMODULATIONTECHNIQUES

Digital Modulation provides more information capacity, high data security, quicker systemavailability with great quality communication. Hence, digital modulation techniques have a greaterdemand,fortheircapacitytoconveylargeramountsofdatathananalogones.

There are many types of digital modulation techniques and we can even use a combination of thesetechniques as well. In this chapter, we will be discussing the most prominent digital modulationtechniques.

if the information signal is digital and the amplitude (lV of the carrier is varied proportional totheinformationsignal,adigitallymodulatedsignalcalledamplitudeshiftkeying(ASK)isproduced.
If the frequency (f) is varied proportional to the information signal, frequency shift keying (FSK) isproduced,andifthe phase ofthe carrier(0)isvariedproportionalto the informationsignal,
phase shift keying (PSK) is produced. If both the amplitude and the phase are varied proportional tothe information signal, quadrature amplitude modulation (QAM) results. ASK, FSK,PSK,

andQAMareallforms ofdigitalmodulation:



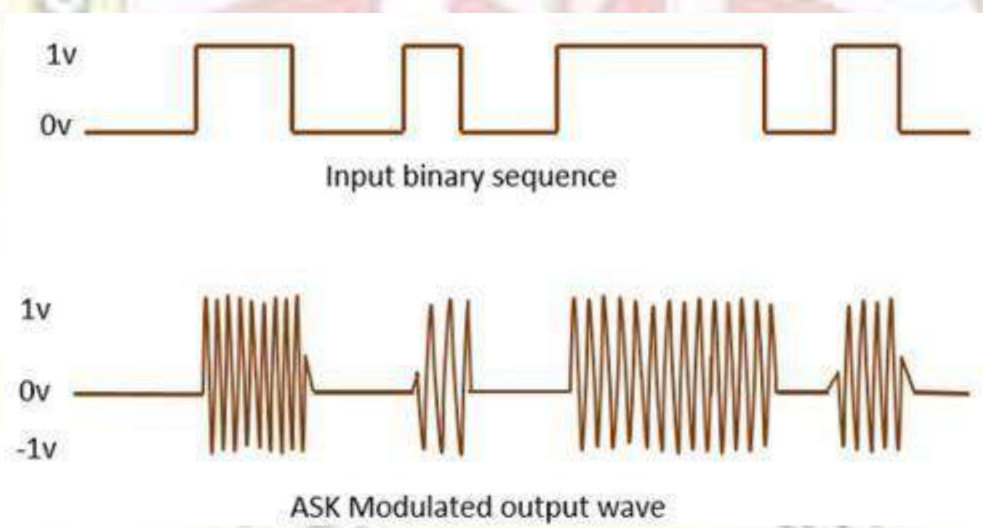
asimplifiedblockdiagramforadigitalmodulationsystem.

5.5AmplitudeShiftKeying

Theamplitudeoftheresultantoutputdependsupontheinputdatawhetheritshouldbeazerolevelora variationofpositive andnegative,dependinguponthe carrierfrequency.

AmplitudeShiftKeying(ASK)isatypeofAmplitudeModulationwhichrepresentsthebinarydata intheformofvariationsinthe amplitudeofasignal.

FollowingisthediagramforASKmodulatedwaveformalongwithits input.



Anymodulatedsignalhasahighfrequencycarrier.ThebinarysignalwhenASKismodulated,givesa zerovalueforLOW inputandgivesthe carrieroutputforHIGHinput.
Mathematically,amplitude-shiftkeyingis

$$v_{(ask)}(t) = [1 + v_m(t)]\left[\frac{A}{2}\cos(\omega_c t)\right]$$

where vask(t)=amplitude-shiftkeyingwave

vm(t)=digitalinformation(modulating)signal(volts)A/2 = unmodulated carrieramplitude (volts)
ωc=analogcarrierradianfrequency(radianspersecond, 2πfct)

InaboveEquation,themodulating signal[vm(t)]isanormalizedbinarywaveform,where+1V=logic1 and-1V=logic0.Therefore,foralogic1 input,vm(t)=+1V,Equation2.12 reducesto

$$v_{(ask)}(t) = [1 + 1]\left[\frac{A}{2}\cos(\omega_c t)\right]$$

$$= A\cos(\omega_c t)$$

Mathematically, amplitude-shift keying is (2.12) where vask(t) = amplitude-shift keying wavevm(t)=digitalinformation(modulating)signal(volts)A/2=unmodulatedcarrieramplitude(volts)

ωc= analog carrier radian frequency (radians per second, 2πfct) In Equation 2.12, the modulatingsignal [vm(t)] is a normalized binary waveform, where + 1 V = logic 1 and-1 V = logic 0.Therefore, foralogic1input, vm(t) =+1V,Equation2.12reducestoandforalogic0input, vm(t) =-1 V,Equationreducesto

$$v_{(ask)}(t) = [1 - 1]\left[\frac{A}{2}\cos(\omega_c t)\right]$$

Thus,themodulatedwavevask(t),iseitherAcos(ωct)or0.Hence,thecarrieriseither"on“or"off,"which iswhyamplitude-shiftkeyingissometimesreferredtoason-offkeying(OOK).
itcanbeseen thatforevery changeintheinputbinary datastream,thereisonechangeintheASKwaveform,and thetimeofonebit(tb)equalsthetime ofoneanalog signalingelement(t,).
B=fb/1=fb baud=fb/1=fb

Example:

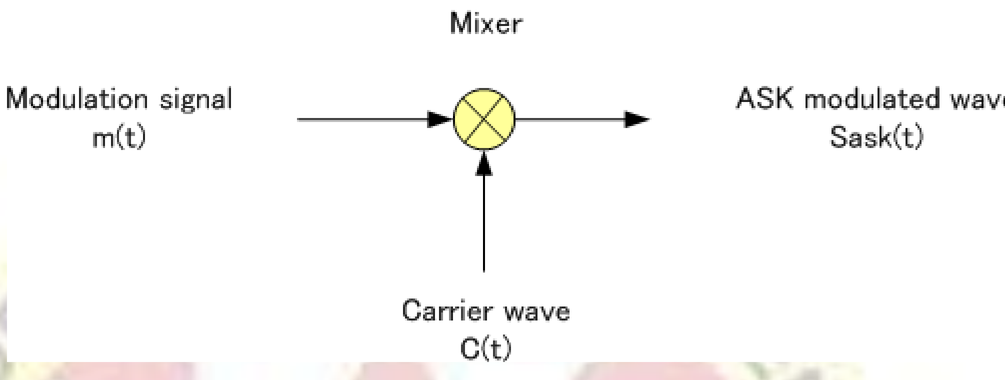
Determine the baud and minimum bandwidth necessary to pass a 10 kbpsbinary signal usingamplitude shift keying. 10Solution For ASK, N = 1, and the baud and minimum bandwidth aredeterminedfromEquations 2.11and2.10,respectively:

B =10,000/1=10,000

baud = 10,000 /1= 10,000

The use of amplitude-modulated analog carriers to transport digital information is a relatively low-quality, low-cost type of digital modulation and, therefore, is seldom used except for very low-speedtelemetrycircuits.

ASKTRANSMITTER:



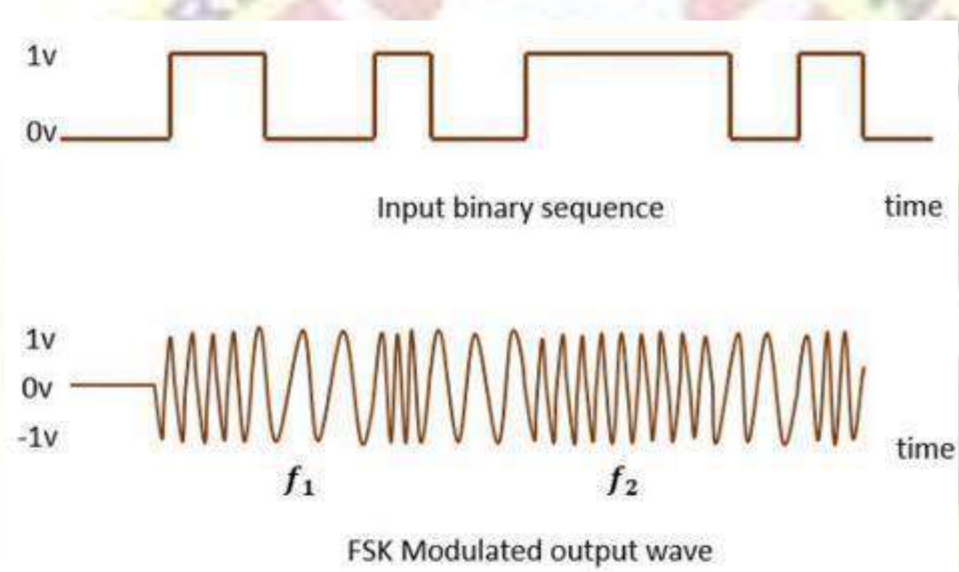
The input binary sequence is applied to the product modulator. The product modulator amplitudemodulates the sinusoidal carrier .it passes the carrier when input bit is '1' .it blocks the carrier wheninputbitis'0.'

5.6 FREQUENCYSHIFTKEYING

Thefrequency of the output signal will be either high or low, depending upon theinputdataapplied.

Frequency Shift Keying (FSK) is the digital modulation technique in which the frequency of thecarriersignal varies accordingtothediscretedigital changes.FSKis aschemeof frequencymodulation.

Followingisthediagramfor FSKmodulatedwaveformalongwithitsinput.



The output of a FSK modulated wave is high in frequency for a binary HIGH input and is low infrequencyforabinaryLOWinput.Thebinary1sand 0sarecalled**Mark**and**Space frequencies**.

FSK is a form of constant-amplitude angle modulation similar to standard frequency modulation(FM) except the modulating signal is a binary signal that varies between two discrete voltage levelsrather than a continuously changing analog waveform.Consequently,FSK issometimes called*binary FSK*(BFSK).ThegeneralexpressionforFSKis

$$v_{fsk}(t) = V_c \cos\{2\pi[f_c + v_m(t)\,\Delta f]t\}$$

where

$v_{fsk}(t)$=binaryFSKwaveform

V_C=peakanalogcarrieramplitude(volts)

f_c=analogcarriercenterfrequency(hertz)

f=peakchange(shift)intheanalogcarrierfrequency(hertz)vm
(t)=binaryinput(modulating)signal(volts)
From Equation 2.13, it can be seen that the peak shift in the carrier frequency (f) is proportional tothe amplitude of the binary input signal (v_m[t]), and the direction of the shift is determined by thepolarity.

Themodulatingsignalisanormalizedbinarywaveformwherealogic1=+1Vandalogic0=-1

V. Thus,foralogiclinput,v_m(t)=+1,Equation2.13 canberewrittenas

$$v_{fsk}(t) = V_c \cos[2\pi(f_c + \Delta f)t]$$

For alogic0input,v_m(t)=-1,Equationbecomes

$$v_{fsk}(t) = V_c \cos[2\pi(f_c - \Delta f)t]$$

WithbinaryFSK,thecarrier centerfrequency(f_c)isshifted(deviated)up and downinthefrequencydomainbythebinaryinputsignalas showninFigure2-3.

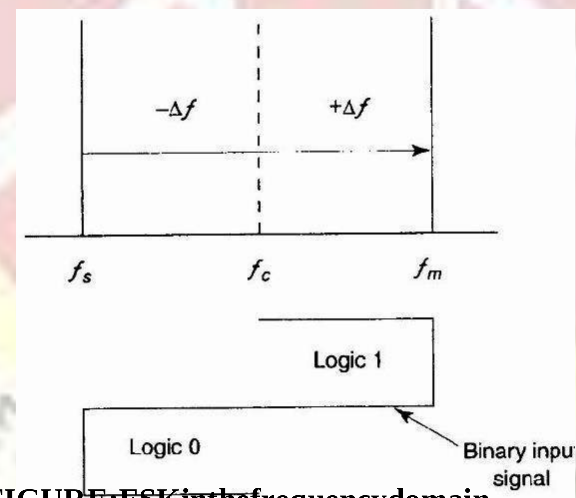


FIGURE:FSKinthefrequencydomain

As the binary input signal changes from a logic 0 to a logic 1 and vice versa, the output frequencyshifts between two frequencies: a mark, or logic 1 frequency (f_m), and a space, or logic 0 frequency(f_s). The mark and space frequencies are separated from the carrier frequency by the peak frequencydeviation(f)andfromeachotherby2f.

FrequencydeviationisillustratedinFigure2-3andexpressedmathematicallyas

$$f=|f_m - f_s|/ 2$$

where f= fre quencydeviation(hertz)

$|f_m–f_s|$=absolutedifferencebetweenthemarkandspacefrequencies(hertz)

Figure2-4ashowsinthetimedomainthebinaryinput toanFSKmodulator andthecorrespondingFSKoutput.

Whenthebinaryinput(f_b)changes fromalogic1toalogic0andviceversa,theFSKoutputfrequencyshiftsfromamark(f_m)toa space (f_s)frequencyandvice versa.

InFigure2-5a,themarkfrequencyisthehigher frequency $(f_c+$ f) andthespacefrequencyisthelowerfrequency(f_c-f),althoughthisrelationship couldbejustthe opposite.

Figure2-5bshowsthetruthtablefora binaryFSK modulator.Thetruthtable showstheinputandoutputpossibilitiesfora givendigitalmodulationscheme.

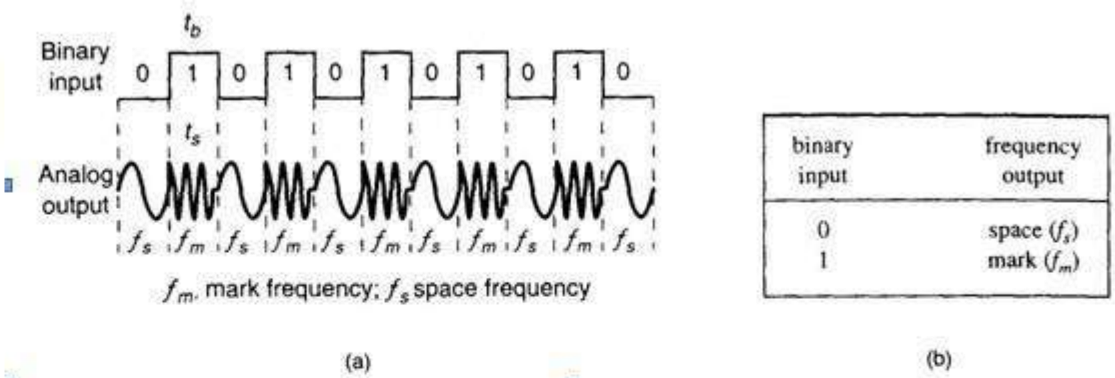


FIGURE 2-4 FSK in the time domain: (a) waveform: (b) truth table

FSK BitRate,Baud,andBandwidth

In Figure 2-4a, it can be seen that the time of one bit *(t_b)* is the same as the time the FSK output is amark of spacefrequency*(t_s)*.Thus,the bit time equals the time of an FSK signaling element,andthe bitrateequals thebaud.

Thebaudfor binaryFSKcanalsobedetermined bysubstituting N=1 inEquation2.11:

baud $= f_b / 1 = f_b$

Theminimumbandwidthfor FSKisgivenas

$$B = |(f_s - f_b) - (f_m - f_b)|$$

$= |(f_s - f_m)| + 2f_b$

andsince$|(f_s - f_m)|$equals2f,theminimumbandwidthcanbeapproximatedas

$$B = 2(f + f_b) \qquad (2.15)$$

where

B=minimumNyquistbandwidth(hertz)f= frequency deviation $|(f_m - f_s)|$ (hertz)f_b=inputbitrate (bps)

Example2-2

Determine (a) the peakfrequency deviation, (b) minimum bandwidth, and (c) baudfor a binaryFSKsignalwithamarkfrequencyof 49kHz,aspacefrequency of51kHz,andaninputbitrateof2kbps.

Solution

a. Thepeakfrequencydeviation isdeterminedfromEquation2.14:

$$f = |149kHz - 51kHz|/2 = 1kHz$$

b. TheminimumbandwidthisdeterminedfromEquation2.15:

B= 2(100+ 2000)

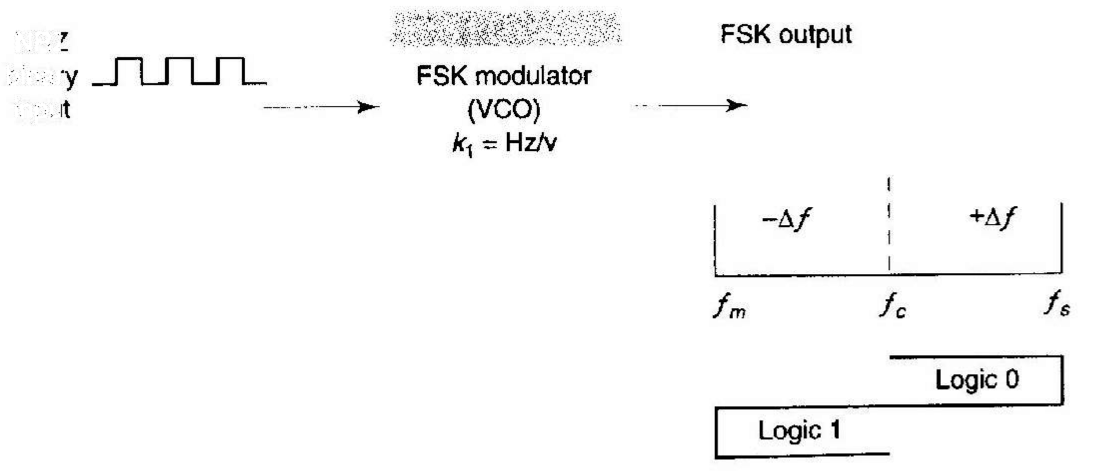
=6kHz

c. For FSK, N=1,andthebaudisdeterminedfromEquation2.11as

baud= 2000/1=2000

FSKTRANSMITTER:

Figure 2.6 shows a simplified binary FSK modulator, which is very similar to a conventional FMmodulator and is very often a voltage-controlled oscillator (VCO).Thecenter frequency (fc)ischosensuchthatitfallshalfwaybetweenthemarkandspacefrequencies.



A logic 1 input shifts the VCO output to the mark frequency, and a logic 0 input shifts the VCOoutput to the space frequency. Consequently, as the binary input signal changes back and forthbetween logic 1 and logic 0 conditions, the VCO output shifts or deviates back and forth betweenthe markandspacefrequencies.

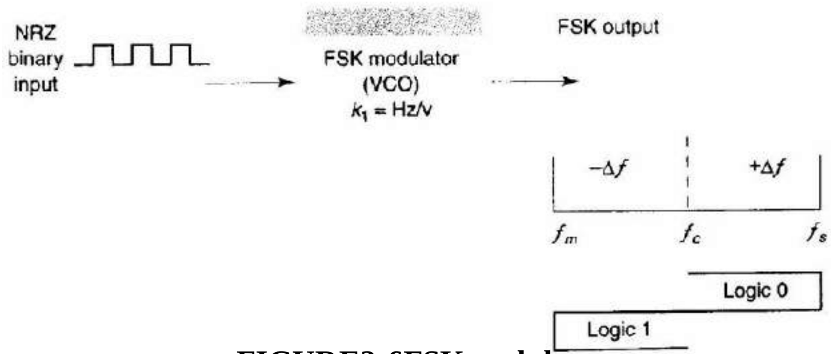


FIGURE2-6FSKmodulator

AVCO-FSKmodulatorcanbeoperated inthesweepmodewherethepeak frequencydeviationissimplythe productofthe binaryinputvoltageandthe deviationsensitivityofthe VCO.

Withthesweepmodeofmodulation,thefrequencydeviationisexpressedmathematicallyas

$$f = v_m(t) k_l \qquad (2\text{-}19)$$

$v_m(t)$ =peakbinarymodulating-signal voltage(volts)

k_l= deviationsensitivity(hertz pervolt).

FSKReceiver

FSKdemodulation isquitesimplewithacircuitsuchastheoneshowninFigure5-7.

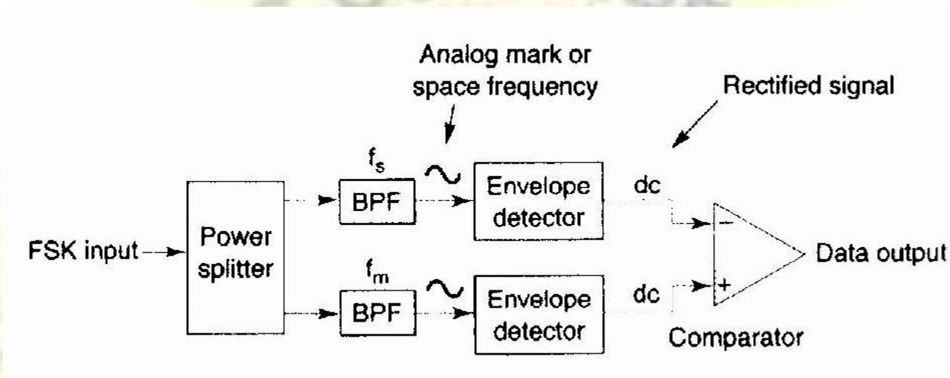


FIGURE5.7NoncoherentFSKdemodulator

The FSK inputsignal is simultaneously applied to the inputs of both bandpassfilters (BPFs)through a power splitter.The respective filter passes only the mark or only the space frequency on toits respective envelope detector.The envelope detectors, in turn, indicate the total power in eachpassband, and the comparator responds to the largest of the two powers.This type of FSK detectionisreferredtoasnoncoherentdetection.

Figure 5-8 shows the block diagram for a coherent FSK receiver.The incoming FSK signal ismultipliedbyarecoveredcarriersignal thathastheexactsamefrequency andphaseasthetransmitterreference.

However, the two transmittedfrequencies (themark and spacefrequencies)are notgenerallycontinuous; it is not practical to reproduce a local reference that is coherent with both of them.Consequently,coherentFSKdetectionis seldomused.

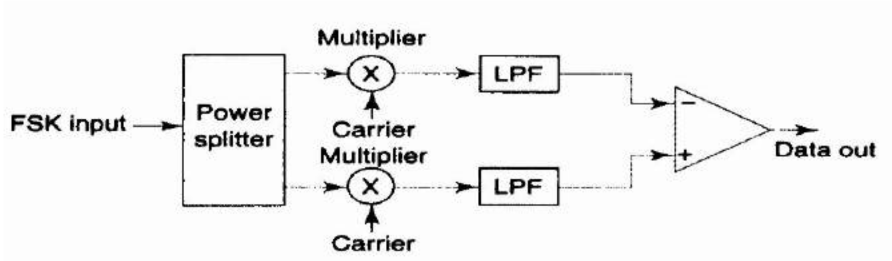
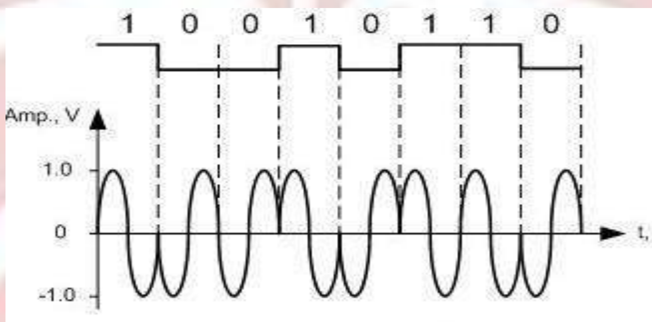


FIGURE5-8CoherentFSKdemodulator

5.7 PHASESHIFTKEYING:

The phase of the outputsignal gets shifted depending upon the input. These aremainly of twotypes, namely BPSK and QPSK, according to the number of phase shifts. The other one is DPSKwhichchanges the phaseaccordingtothepreviousvalue.



Phase shift keying (PSK)

Phase Shift Keying (PSK) is the digital modulation technique in which the phase of the carriersignal is changed by varying the sine and cosine inputs at a particular time. PSK technique is widelyusedforwirelessLANs,bio-metric,contactlessoperations,alongwithRFIDandBluetoothcommunications.

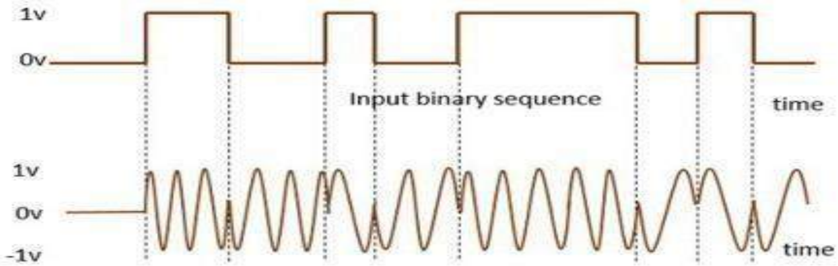
PSKisoftwo types,depending uponthephasesthesignalgetsshifted.Theyare −

BinaryPhaseShiftKeying(BPSK)

This is also called as **2-phase PSK** (or) **Phase Reversal Keying**. In this technique, the sine wavecarriertakestwophasereversalssuchas 0°and180°.

BPSKisbasicallyaDSB-SC(DoubleSidebandSuppressedCarrier)modulationscheme,formessagebeingthe digitalinformation.

FollowingistheimageofBPSKModulated outputwavealongwith itsinput.



BPSK Modulated output wave

BinaryPhase-ShiftKeying

ThesimplestformofPSKis*binaryphase-shiftkeying*(BPSK),whereN=1and*M=2*.Therefore, with BPSK, two phases (2^1 = 2) are possible for the carrier.One phase represents alogic1,andtheotherphaserepresentsalogic0.Astheinputdigital signal changesstate(i.e.,froma 1 to a 0 or from a 0 to a 1), the phase of the output carrier shifts between two angles that areseparatedby180°.

Hence, other names for BPSK are *phase reversal keying* (PRK) and *biphase modulation*. BPSKisaformofsquare-wavemodulationofa*continuouswave (CW)*signal.

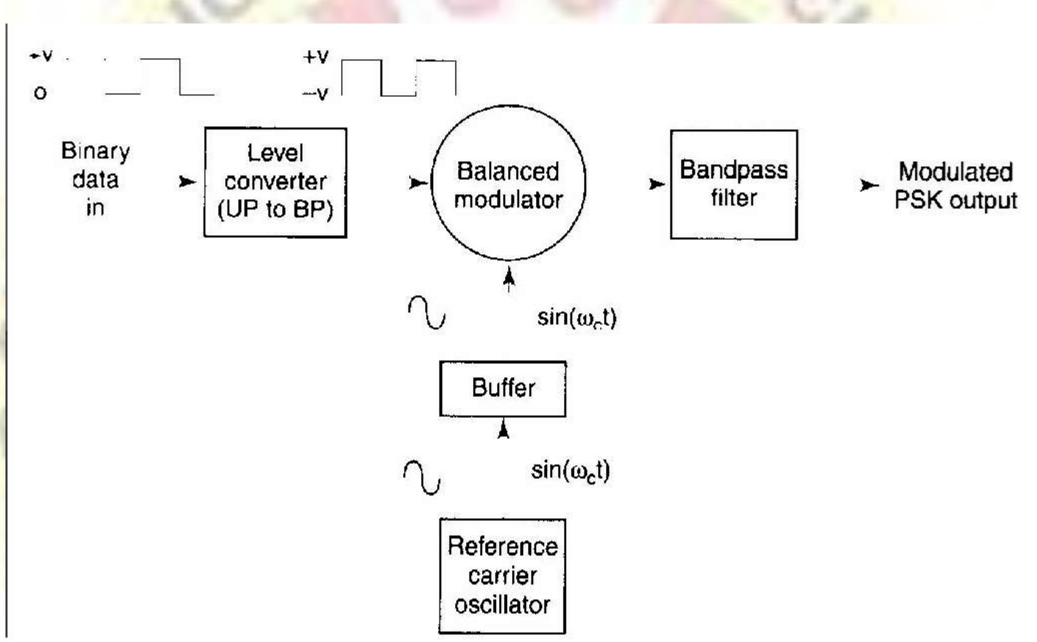


FIGURE5-12PSKtransmitter

BPSKTRANSMITTER:

Figure 5-12 shows a simplified block diagram of a BPSK transmitter. The balanced modulator actsas a phase reversing switch. Depending on the logic condition of the digital input, the carrier istransferred tothe output eitherinphaseor180°outofphase withthe referencecarrieroscillator.

Figure 5-13 shows the schematic diagram of a balanced ringmodulator. The balanced modulatorhas two inputs: a carrier that is in phase with the reference oscillator and the binary digital data. Forthe balanced modulator to operate properly, the digital input voltage must be much greater than thepeakcarriervoltage.

This ensures that the digital input controls the on/off state of diodes D1 to D4. If the binary input isalogic1(positivevoltage), diodesD1andD2areforwardbiasedandon, whilediodesD3andD4

arereversebiasedandoff(Figure2-
13b).Withthepolaritiesshown,thecarriervoltageisdevelopedacrosstransformerT2inphase withthecarriervoltageacrossT

1.Consequently,theoutputsignalisinphasewiththereferenceoscillator.

If the binary inputis a logic 0 (negative voltage), diodes Dl and D2 are reverse biased and off,while diodes D3 and D4 are forward biased and on (Figure 9-13c). As a result, the carrier voltage isdevelopedacrosstransformerT2180°outofphasewiththe carriervoltage acrossT1.

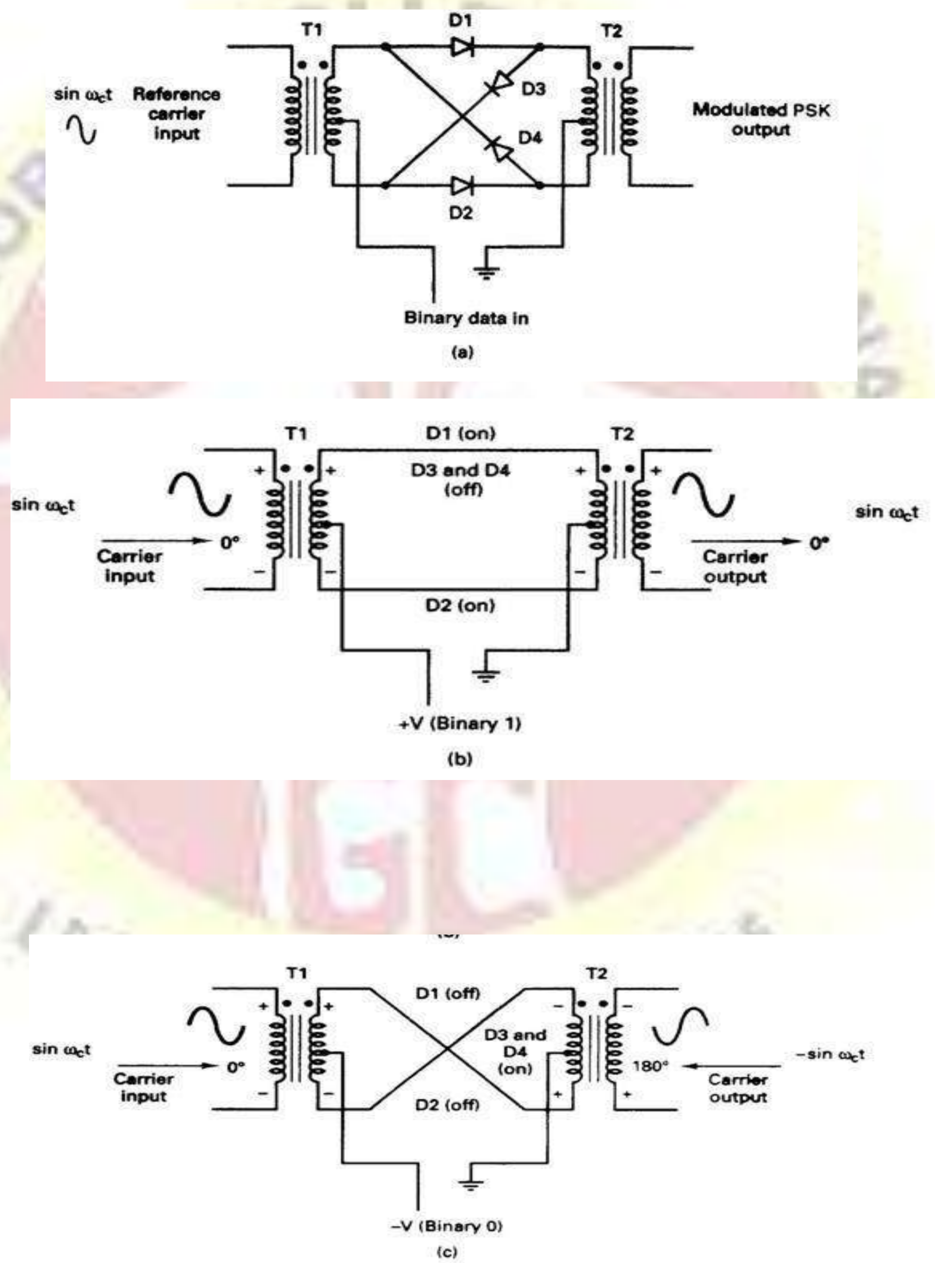


FIGURE9-13(a)Balanced ringmodulator; (b)logic1input;(c)logic0input

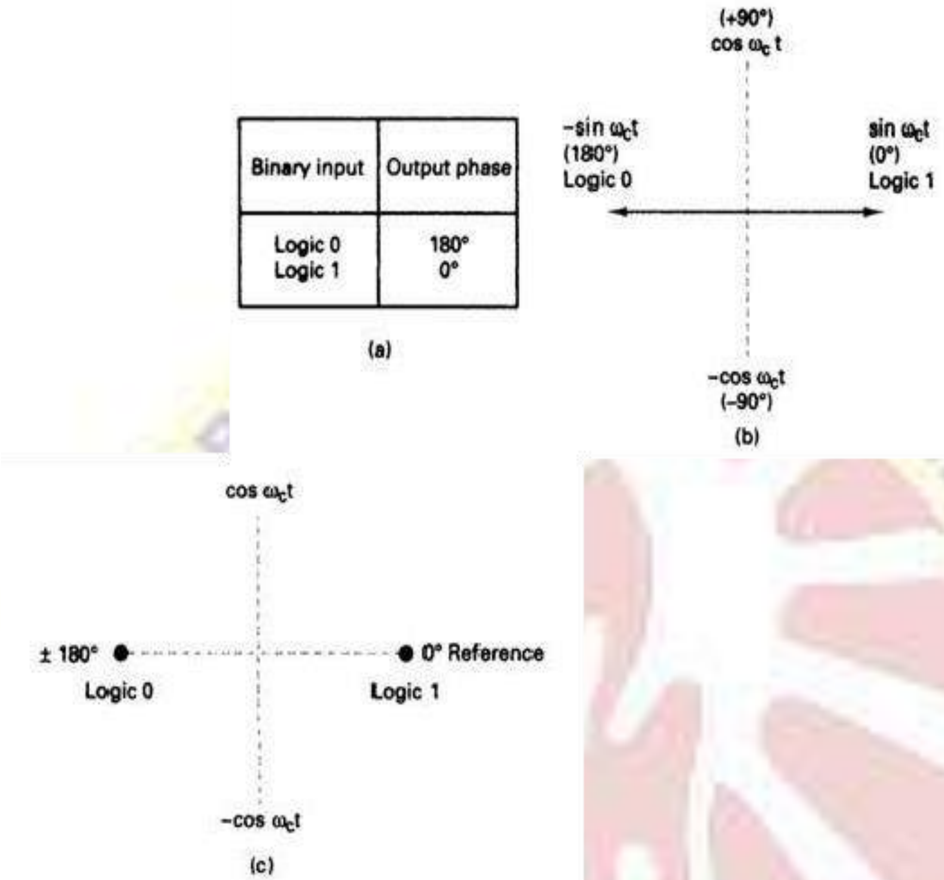


FIGURE2-

14BPSKmodulator:(a)truthtable;(b)phasordiagram;(c)constellationdiagram

BANDWIDTHCONSIDERATIONSOFBPSK:

InaBPSKmodulator.thecarrierinputsignalis multipliedbythebinarydata.

If+1Visassignedtoalogic1and-1Visassignedtoalogic0,theinputcarrier (sinω_ct) ismultipliedbyeithera+or-1.

The output signal is either + 1 sin ω_ct or -1 sin ω_ct the first represents a signal that is *in phase* withthereferenceoscillator,thelatterasignalthatis180°outofphasewiththereferenceoscillator.Eachtime theinputlogic conditionchanges,theoutput phasechanges.

Mathematically, theoutputofaBPSKmodulatorisproportionalto

$$\text{BPSKoutput}=[\sin(2\pi f_a t)]\times[\sin(2\pi f_c t)] \qquad (2.20)$$

where

f_a = maximum fundamental frequency of binary input (hertz)f_c=referencecarrierfrequency(hertz)

Solvingforthetrigidentityfortheproductoftwosinefunctions,0.5cos[2π(f_c–f_a)t]–0.5cos[2π(f_c+f_a)t]

Thus, theminimumdouble-sidedNyquistbandwidth*(B)is*

f_c+f_a $\qquad$ f_c+f_a

$-(f_c+f_a)$ $\qquad$ or $\qquad$ $-f_c\ +f_a$

$$— \frac{}{2f_a}$$

andbecausef_a=f_b/2,where***fb***= inputbitrate,

where*B*istheminimumdouble-sidedNyquistbandwidth.

Figure 5.8 shows the output phase-versus-time relationship for a BPSK waveform. Logic 1 inputproduces an analog output signal with a 0° phase angle, and a logic 0 input produces an analogoutputsignalwitha180°phaseangle.

Asthebinaryinputshifts betweenalogic1andalogic0conditionandviceversa, thephaseoftheBPSKwaveformshiftsbetween0°and180°,respectively.

BPSK signaling element *(t_s)* is equal to the time of one information bit *(t_b),* which indicates that thebitrateequals thebaud.

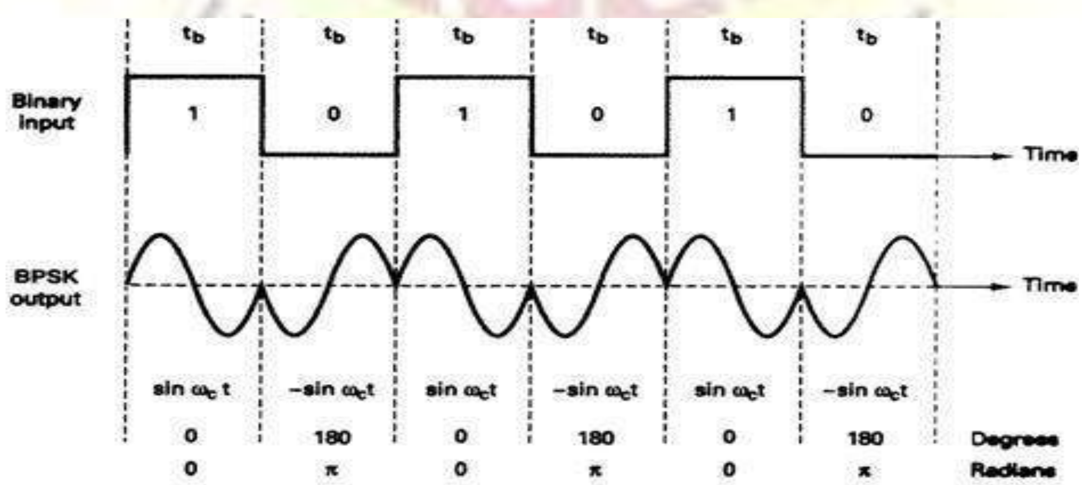


FIGURE5.8Outputphase-versus-time relationshipforaBPSKmodulator

Example:

For a BPSK modulator with a carrier frequency of 70 MHz and an input bit rate of 10 Mbps,determine the maximum and minimum upper and lower side frequencies, draw the output spectrum,de-termine the minimumNyquistbandwidth,andcalculatethe baud..

Solution

Substituting intoEquation2-20 yields

output $=[\sin(2\pi f_a t)] \times [\sin(2\pi f_c t)]; f_a = f_b / 2 = 5\text{MHz}$

$=[\sin 2\pi(5\text{MHz})t)] \times [\sin 2\pi(70\text{MHz})t)]$

$=0.5\cos[2\pi(70\text{MHz}-5\text{MHz})t] - 0.5\cos[2\pi(70\text{MHz}+5\text{MHz})t]$

lowersidefrequency uppersidefrequency

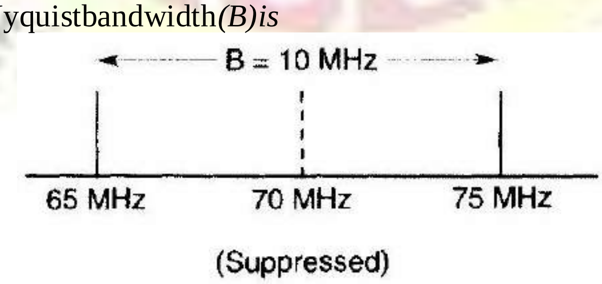
Minimumlowersidefrequency(LSF):

LSF=70MHz-5MHz=65MHz

Maximumuppersidefrequency(USF):

USF=70MHz+ 5MHz= 75MHz

Therefore,theoutputspectrumfortheworst-casebinaryinputconditions*is*as follows:TheminimumNyquistbandwidth*(B)is*



B= 75MHz-65MHz= 10MHz

andthebaud=$f_b$or 10megabaud.

*****