UNIT I - ANALOG COMMUNICATION

AM – Frequency spectrum – vector representation – power relations – generation of AM – DSB, **DSB/SC, SSB, VSB** AM Transmitter & Receiver; FM and PM – frequency spectrum – power relations : NBFM & WBFM, **Generation of FM** and DM, **Amstrong method & Reactance modulations** : FM & PM frequency.

INTRODUCTION

Communication:

It is the process of transferring information from one place to another. Communication channel is the media by which the information is sent. The channel could be a wired line such as copper wire or wireless such as free space.

Building Block of Communication Systems:

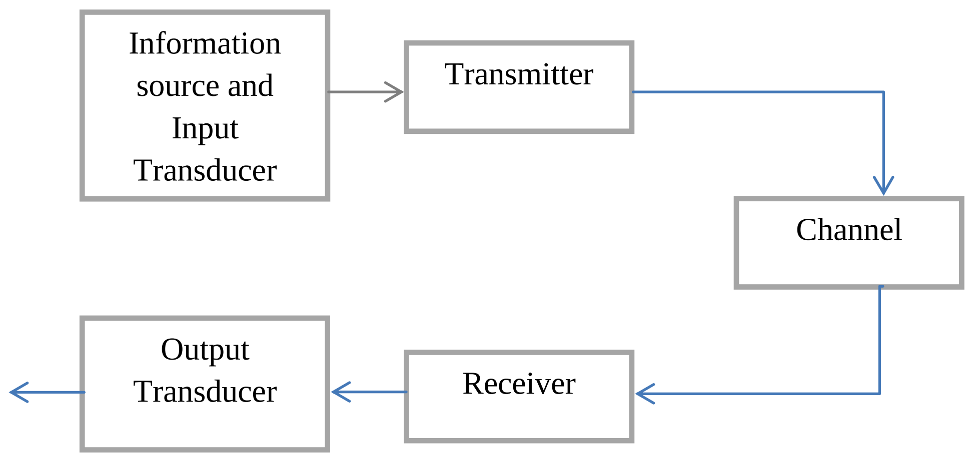


Figure1. Block diagram of communication system

1. Communication systems are designed to send messages or information from a source that generates the messages to one or more destinations. The information generated by the source may be the form of voice (speech source), a picture (image source).
2. A transducer is usually required to convert the output of a source into an electrical signal that is suitable for transmission.
3. The heart of the communication system consists of three basic parts namely, transmitter, channel and receiver.

Transmitter

1. The transmitter converts the electrical signal into a form that is suitable for transmission through the physical channel or transmission medium.
2. In general, the transmitter performs the matching of the message signal to the channel by a process called modulation. (AM, FM, and PM)

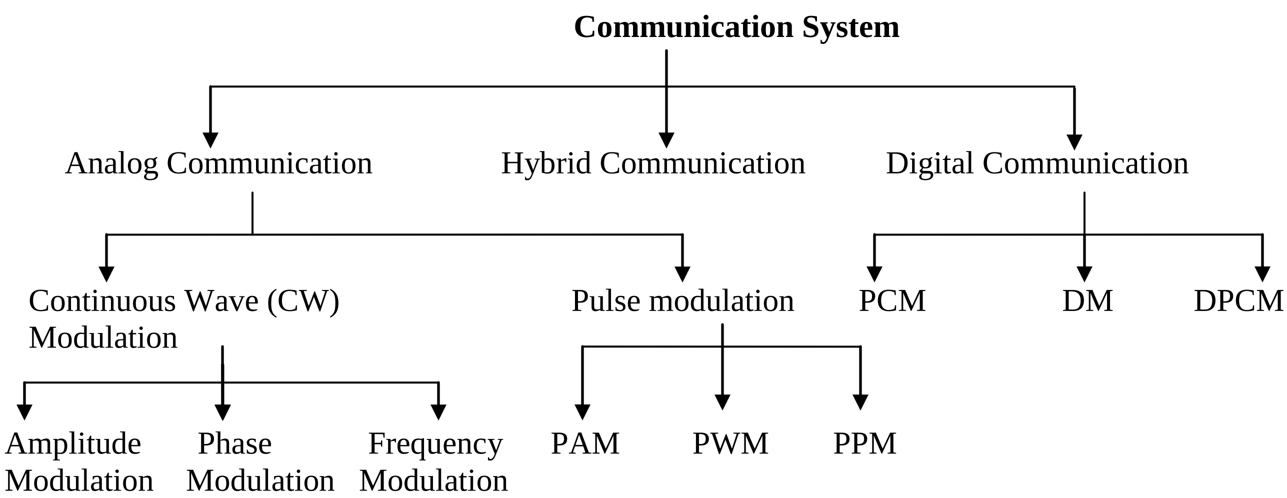
Channel

The communications channel is the physical medium that is used to send the signal from transmitter to the receiver. In wireless transmission, the channel is usually the atmosphere (free space).

Receiver

1. The function of the receiver is to recover the message signal contained in the received signal. If the message signal is transmitted by carrier modulation, the receiver performs carrier demodulation in order to extract the message from the sinusoidal carrier.
2. Besides performing the primary function of signal demodulation, the receiver also performs a number of peripheral functions, including signal filtering and noise suppression.

<u>Types of Communication System:</u>



1. Analog Communication:

- Analog communication involves transferring an analog signal containing information (no digitization at any point) between two users.
- e.g: voice signal, picture signal, and sound signal

2. Digital Communication:

- Digital communication involves transferring an digital signal containing information between two users.
- e.g.: data from computer, text message.

<u>Modulation:</u>

Modulation is the process by which some characteristics of a high frequency carrier signal is varied, in accordance with the instantaneous value of the modulating or message signal.

<u>Demodulation:</u>

Demodulation is the process of separating the message signal from the modulated carrier.

<u>Amplitude Modulation (AM)</u>

Amplitude modulation is the process by which amplitude of the carrier signal is varied in accordance with the instantaneous value of the modulating signal, but frequency and phase of the carrier wave remains constant.

1. Modulating signal is represented by e_m,

$$\boldsymbol{e_m = E_m \sin \omega_m t} \qquad \text{------------------ (1)}$$

2. Carrier signal is represented by e_c,

$$\boldsymbol{e_c = E_c \sin \omega_c t} \qquad \text{---------------------- (2)}$$

Where,

E_m is the maximum amplitude of modulating signal

$\boldsymbol{E_c}$ denotes the maximum amplitude of carrier signal

ω_m represents the frequency of modulating signal

ω_c is the frequency of carrier signal.

3. The amplitude modulated wave is represented by e_{Am},

$$\boldsymbol{e_{Am} = E_{AM} \sin \omega_c t} \qquad \text{------------------------- (3)}$$

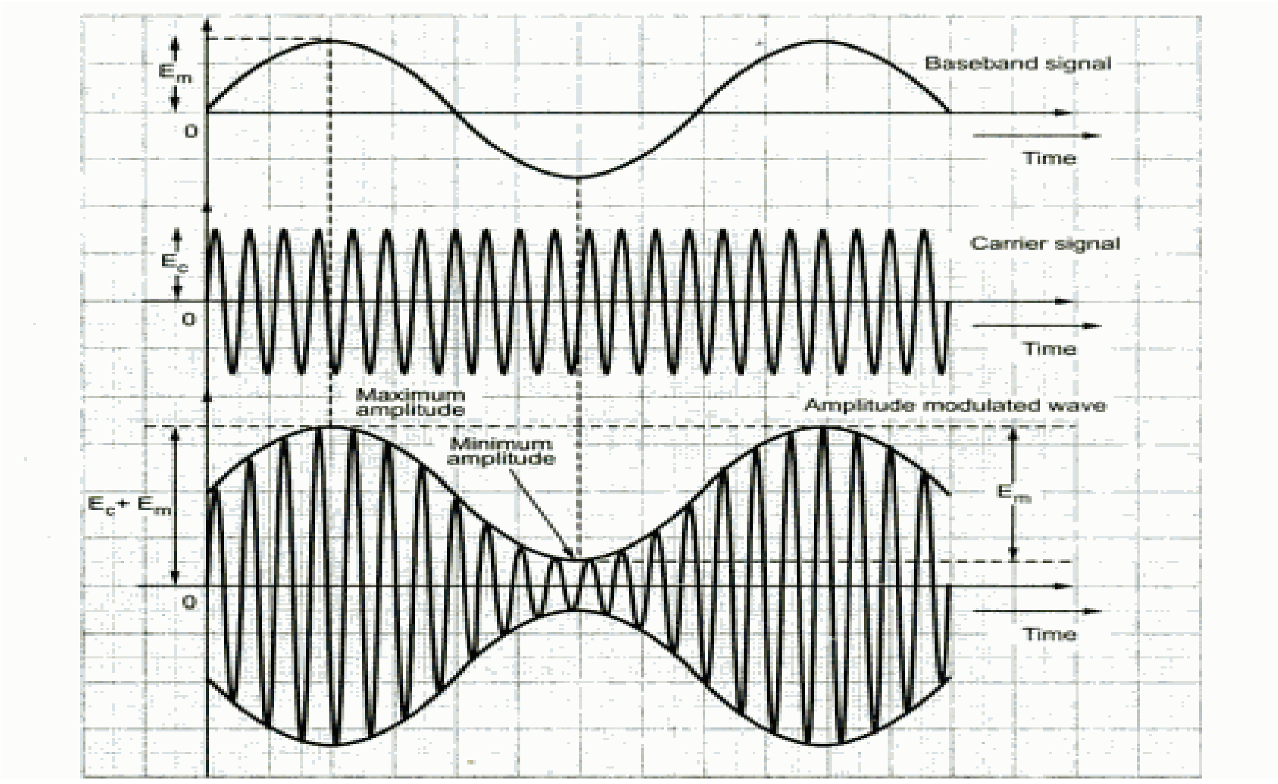


Figure 3: Amplitude modulation (a) modulating signal (information signal); (b) carrier signal; (c) Amplitude Modulated signal.

We know that, $\mathbf{E_{Am} = E_c + e_m}$ -------------------(4)

Sub eqn (1) in eqn (4) $E_{Am} = E_c + E_m \sin\omega_m t$

$= E_c[1+\frac{Em}{Ec}\sin\omega_m t]$ [$\frac{Em}{Ec}$ **= m_a-->modulation index.]**

$\mathbf{E_{Am} = E_c[1+ m_a \sin\omega_m t]}$ ---------------------(5)

Sub eqn (5) in eqn (3)

(3)=> $e_{Am} = E_c[1+ m_a \sin\omega_m t] \sin\omega_c t$

$= E_c \sin\omega_c t + m_a \sin\omega_m t \sin\omega_c t$

Using $\left[\sin a \ \sin b = \frac{1}{2}[\cos(a-b) - \cos(a+b)]\right]$ the above eqn. becomes

$e_{Am} = E_c \sin\omega_c t + m_a E_c [\frac{1}{2}\cos(\omega_c-\omega_m)t - \frac{1}{2}\cos(\omega_c+\omega_m)t]$ --------------------(6)

sub $\omega_c = 2\pi f_c$; $\omega_m = 2\pi f_m$;

$$\boxed{e_{AM} = \underbrace{E_c \sin\omega_c t}_{\text{carrier}} + \underbrace{\frac{mE_c}{2}\cos(\omega_c - \omega_m)t}_{\text{Lower side band}} - \underbrace{\frac{mE_c}{2}\cos(\omega_c - \omega_m)t}_{\text{Upper side band}}}$$

This is the equation of AM wave.

<u>Frequency spectrum:</u>

Amplitude versus frequency plot of the signal is called frequency spectrum.

- the amplitude and frequency of the carrier is higher than the message signal.
- amplitude of upper and lower side bands are equal.

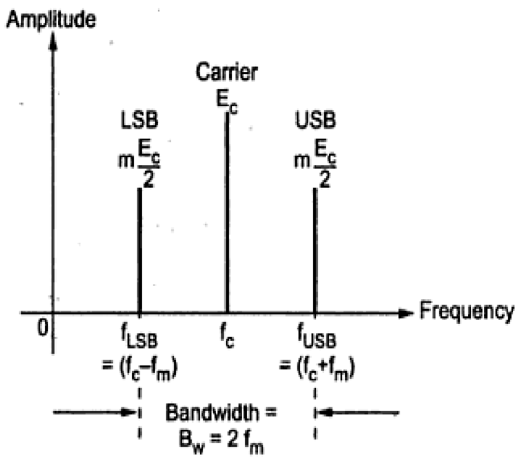


Figure 3: Frequency spectrum

Bandwidth:

- The difference between upper and lower frequency is called bandwidth.

  $\mathbf{B_w = f_{USB} - f_{LSB}}$

  $= (f_c+f_m) - (f_c-f_m)$

  $\mathbf{B_w = 2 f_m}$
- Thus the bandwidth of the AM signal is twice the maximum frequency of message signal.

Modulation Index and percent modulation

The ratio of maximum amplitude of modulating signal to maximum amplitude of carrier signal is called modulation index (m).

Modulation index, $\boldsymbol{m = \frac{E_m}{E_c}}$

- Value of E_m must be less than value of E_c to avoid any distortion in the modulated signal. Hence maximum value of ***modulation index*** will be ***equal to 1*** when, $E_m = E_c$. Minimum value will be zero.
- If modulation index is higher than 1, then it is called ***over modulation***.
- Modulation index is expressed in percentage, it is called ***percentage modulation***.

Calculation of modulation index for AM

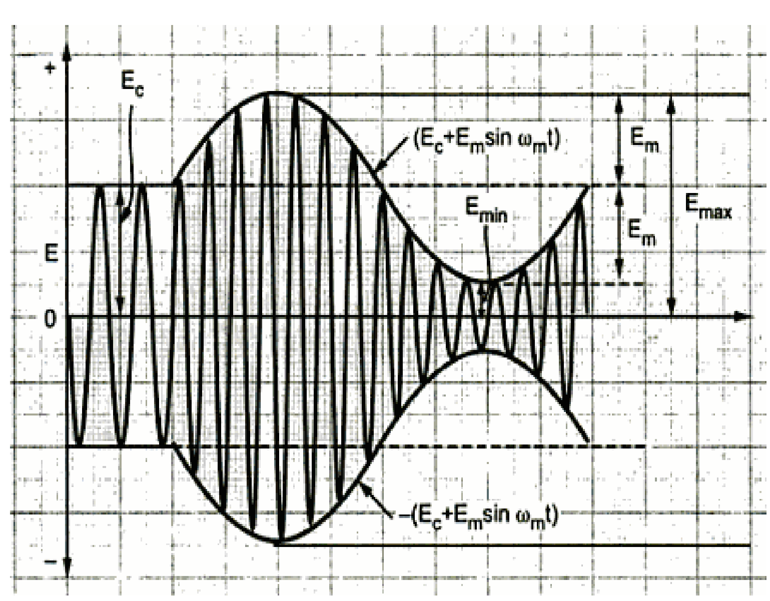
We know that $m = \frac{\mathbf{Em}}{\mathbf{Ec}}$

From the figure $\mathbf{2E_m = E_{max} - E_{min}}$-------(1)

$$E_m = \frac{E_{max} - E_{min}}{2}$$

From fig., $\mathbf{E_c = E_{max} - E_m}$------------- (2)

Sub eqn (1) in eqn (2)



$E_c = E_{max} - [\frac{Emax - Emin}{2}]$

$\mathbf{E_c = \frac{Emax + Emin}{2}}$ ---------------(3)

Therefore $m = \frac{E_m}{E_c} = \frac{(E_{max} - E_{min})/2}{(E_{max} + E_{min})/2}$

$$\boxed{m = \frac{E_{max} - E_{min}}{E_{max} + E_{min}}}$$

AM Power Distribution:

The total transmitted power of an AM wave ***is the sum of the carrier power ,LSB power & USB power***

$$\mathbf{P_{tot} = P_c + P_{LSB} + P_{USB}} \qquad \text{----------------(1)}$$

(i) **Carrier power**

$\mathbf{P_c = E_c^2 / R}$

$\mathbf{E_c}$→maximum amplitude of carrier signal**,**

R→resistance

RMS voltage of $\mathbf{E_c}$ = $\mathbf{E_c / \sqrt{2}}$

Therefore $\mathbf{P_c = (E_c/\sqrt{2})^2/R = E_c^2/2R}$ **--------------- (2)**

(ii) **sideband power**

------------------- (3)

(iii) **Total power**

$$P_{LSB} = P_{USB} = \frac{E_{SB}^2}{R}$$

$$= \left(\frac{\frac{mE_c}{2}}{\sqrt{2}}\right)^2 \times \frac{1}{R} \qquad \because E_{SB} = \frac{mE_c}{2}$$

The

$\frac{^2E_c{}^2}{8R}$

$$\boxed{P_{LSB} = P_{USB} = \frac{m^2 E_c^2}{8R}}$$

2R 2

$$\boxed{P_{Total} = P_c\left(1 + \frac{m^2}{2}\right)} \qquad \text{------------------ (4)}$$

EFFICIENCY:

The efficiency of an AM wave is defined as the **ratio between total sideband power (useful power) to the total transmitting power.**

Transmission efficiency, $\boldsymbol{\eta} = \frac{P_{USB}+P_{LSB}}{P_{Total}} = \frac{m^2}{2+m^2}$

Using eqn. 3 &(4)

$$\eta = \frac{\left[\frac{m^2}{4}P_c + \frac{m^2}{4}P_c\right]}{\left[1 + \frac{m^2}{2}\right]P_c} = \frac{\frac{m^2}{2}}{1 + \frac{m^2}{2}}$$

$$\eta = \frac{m^2}{2+m^2}$$

∴ The percentage transmission efficiency is given as

$$\boxed{\% \ \eta = \frac{m^2}{2+m^2} \times 100 \ \%}$$

If m=1 , then % η = $\frac{1}{2+1}$ * 100 = 33.33%, **% η =33.33%**

Vector Representation of AM:

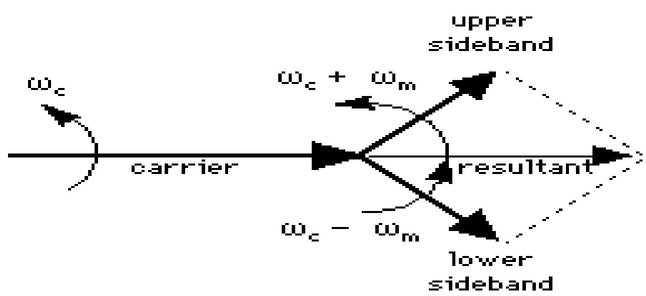


Figure 4: Vector Representation of AM

- The **carrier wave phasor** is taken as the **reference phasor.**
- The **sideband frequency (ω_c+ω_m)** is represented by the **phasor of angular freq +ω_m.**
- The **sideband frequency (ω_c-ω_m)** is represented by the **phasor of angular freq - ω_m.**
- **The resultant phasor is the vector sum of the two sidebands and the carrirer.**

Generation of AM:

- **Low level modulation**
  - Modulation takes place at ***the initial stage*** of Transmitter.
  - First **the carrier is modulated, amplified and then transmitted**.
  - **Less modulating power** is required
- **High level modulation**
  - Modulation takes place at the **final stage** of the transmitter.
  - First the **carrier is fully amplified and then modulated and transmitted.**
  - **Higher modulating signal power** is required.

Low Level AM Modulator (Emitter Modulation):

Class A amplifier is used as a modulator. It has two input 1) carrier signal 2) message signal (modulating signal)

Circuit operation:

- The message signal is applied to the emitter terminal through an isolation transformer
- And the carrier signal is applied directly to the base of the transistor.
- In the absence of message signal the circuit operates as a linear class A amplifier

- So the output is the amplified carrier signal

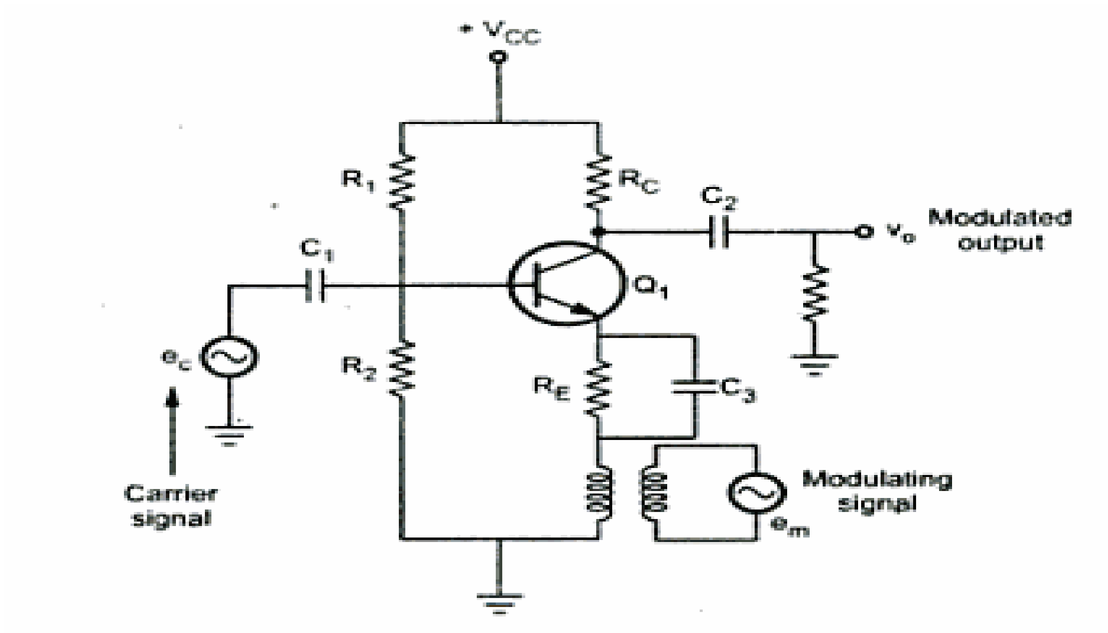
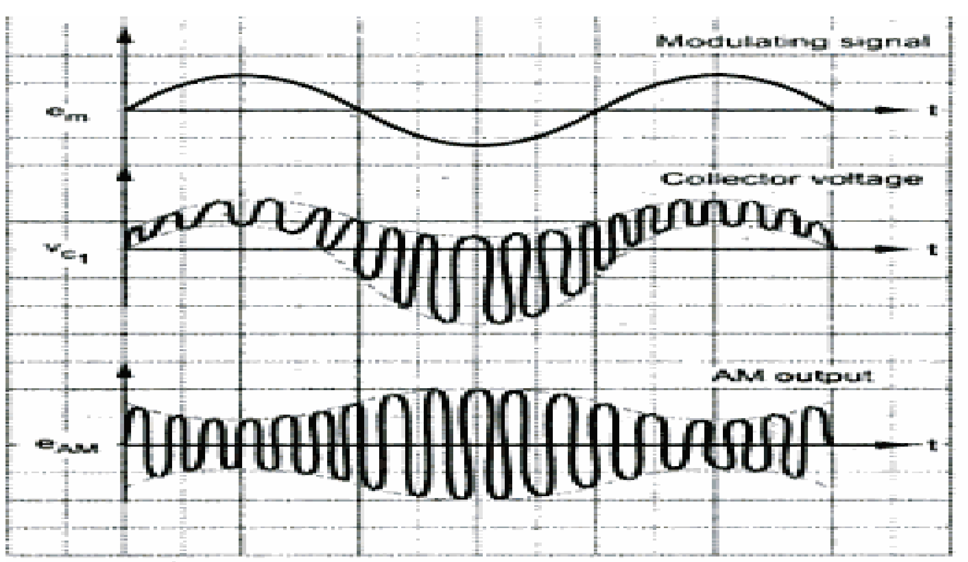


Figure 5 : Emitter modulation

- When the message signal is applied, the amplifier operates nonlinearly and signal multiplication occurs.
- The message signal varies the gain of the amplifier equal to the frequency of the message signal. The depth of modulation is proportional to the amplitude of the message signal.
- The voltage gain is expressed as $A_m = A(1+m\sin\omega_m t)$

  $A_m \rightarrow$ voltage gain without modulation

  $A \rightarrow$ voltage gain without modulation.



(b) Waveforms

Advantages:

- **Simpler circuit.**
- **Less message signal power is required.**

Disadvantages:

- **Low Power efficiency** due to class A operation.

<u>High Level Modulator (collector modulator or Medium Power modulator):</u>

- The modulation takes place at the collector terminal so it is called collector modulator.
- Class C amplifier is used to achieve high efficiency.

<u>Circuit diagram:</u>

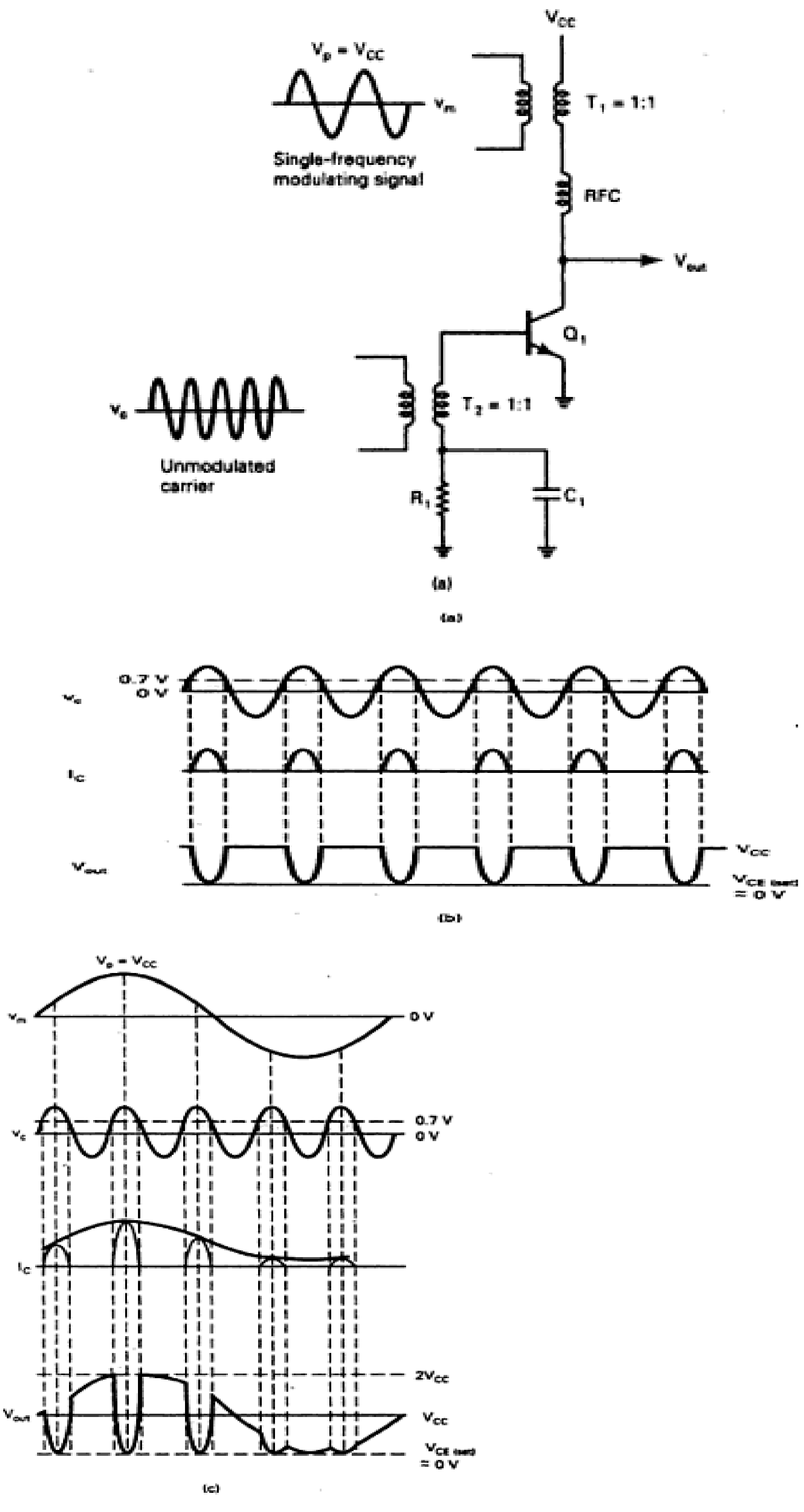


FIGURE 4-16 Simplified medium-power transistor AM DSBFC modulator: (a) schematic diagram; (b) collector waveforms with no modulating signal; (c) collector waveforms with a modulating signal

- The Carrier signalis applied to the base of the transistor through a transformer T1and the message signal is applied to the collector of the transistor through a transformer T2.
- The message signal voltage $v_{m(t)}$ is applied in series with the power supply voltage vcc
- Hence the net collector voltage
    - **$vcc' = vcc + v_{m(t)}$.**
- When the carrier signal voltage is > 0.7 v the transistor turns ON and collector current flows.
- When the message signalvaries ,the net collector voltage vcc' changes.

- Due to this the amplitude of the carrier signalis varied with respect to the variations in message s/l.
- Thus AM signalis produced at the collector terminal.

Advantages:

- Power efficiency is high
- Amplifiers operate at low power level

Disadvantages:

- Higher amplitude message signalis required.
- Amplifier is non-linear ,hence creates noises.

1.1 Generation and Demodulation of AM

AM Modulator circuits (Generation of AM)

1. The device which is used to generate an amplitude modulated wave is known as amplitude modulator.
2. Amplitude modulation may be achieved in a number of ways, the most common method being to use the modulating signal to vary the steady voltage on the output electrode of the amplifier.
3. Vacuum tubes are used for very high power outputs (in the kilowatt and higher ranges), and transistors are used for lower powers.
4. The basic circuit for a BJT modulator is shown in Fig.1. The transistor is normally operated in the class C mode in which it is biased well beyond cutoff.

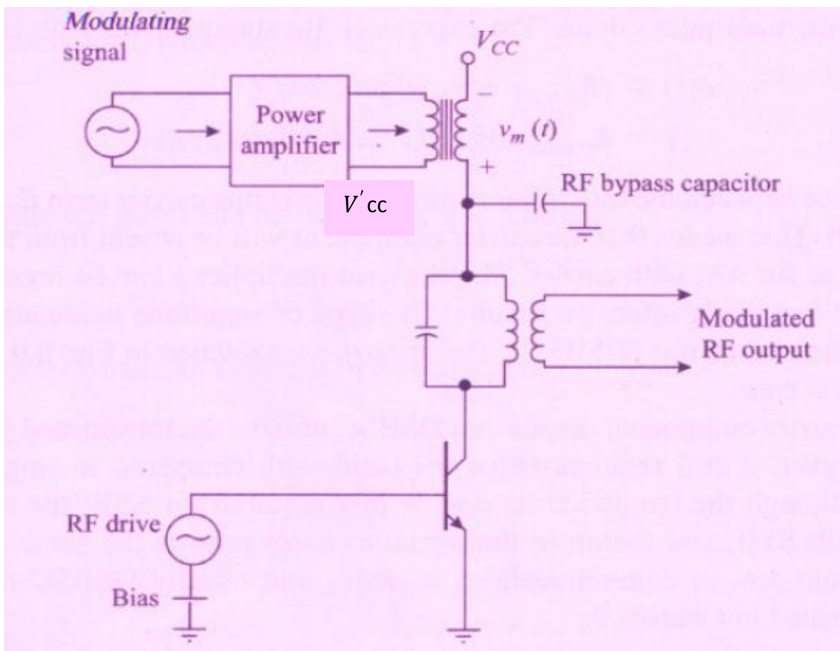


Figure 1. Circuit for a BJT collector modulator

5. The carrier input to the base must be sufficient to drive the transistor into conduction over part of the RF cycle, during which the collector current flows in the form of pulses. These pulses are periodic at the carrier frequency.
6. The tuned circuit in the collector is tuned to resonate at the fundamental component, and thus, to a close approximation, the RF voltage at the collector is sinusoidal.
7. When a modulating voltage is applied, the steady collector voltage changes to a slowly varying voltage (slow compared to the RF cycle) given by $V'_{CC} = V_{CC} + V_m(t)$.
8. The modulating voltage $V_m(t)$ is applied in series with V_{CC} through the low-frequency transformer.
9. The tuned LC circuit associated tuned transformer on the collector receives the AM signal. Because of modulating voltage, the net supply voltage of transistor changes according to slow variations in$V_m(t)$. Hence the RF carrier signal amplitude is also varied according to variations in $V_m(t)$. Thus AM signal is produced across the LC circuit at the collector.

Amplitude Demodulator Circuits (Demodulation of AM)

Envelope Detection

1. At the receiver, a circuit must be provided that recovers the information signal from the modulated carrier. The most common circuit in use is the *diode envelope detector*, which produces an output voltage proportional to the envelope, which is the modulating or information signal.
2. The basic circuit is shown in Fig.2(a). The diode acts as a rectifier and can be considered an ON switch when the input voltage is positive, allowing the capacitor C to charge up to the peak of the RF input.

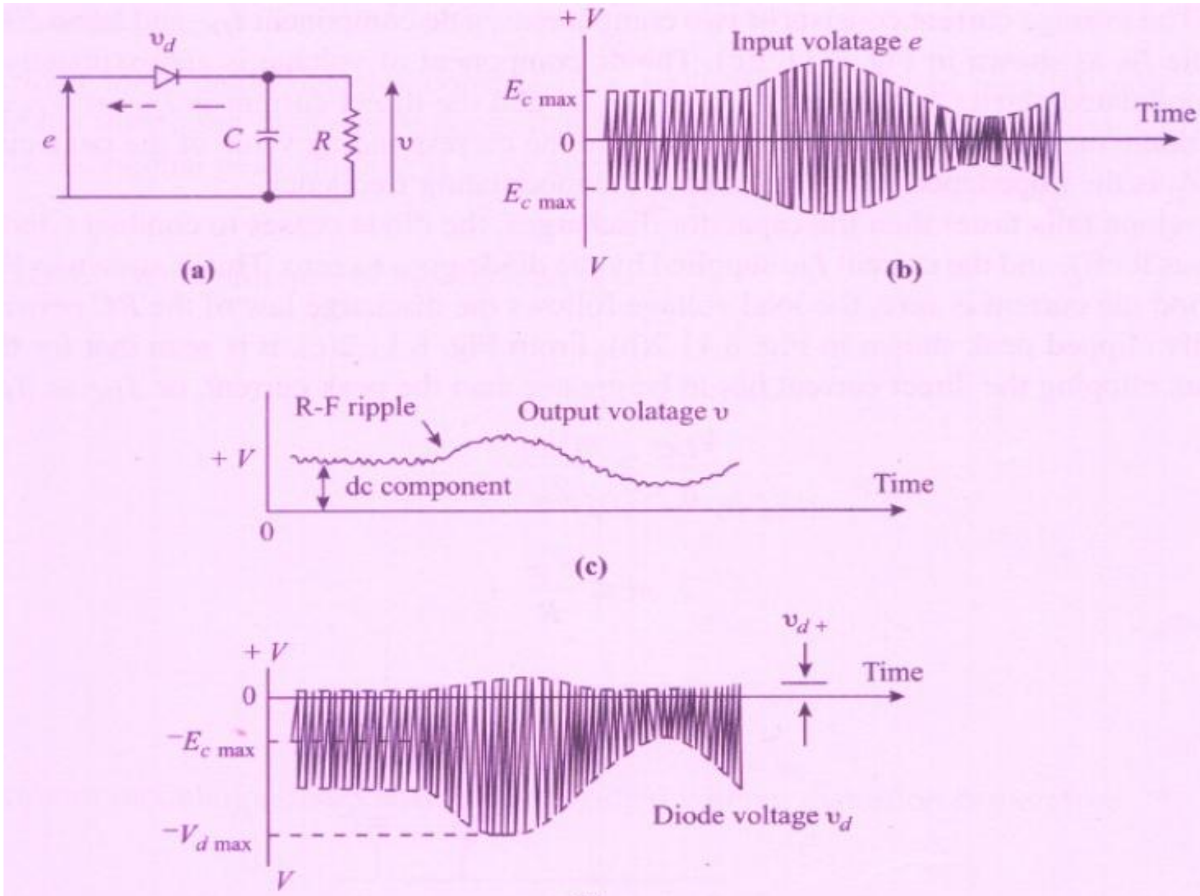


Figure 2.(a) Basic diode envelope detector, (b) voltage input waveform, (c) Voltage output waveform, (d) Voltage across the diode.

3. During the negative half of the RF cycle, the diode is off, but the capacitor holds the positive charge previously received, so the output voltage remains at the peak positive value of RF.
4. There will, in fact, be some discharge of C, producing an RF ripple on the output waveform, which must be filtered out.
5. As theinput voltage rises with the modulating cycle, the capacitor voltage has no difficulty in following this, but during the downward swing in modulation the capacitor may not discharge fast enough unless an additional discharge path is provided by the resistor *R.*
6. The time constant of the *CR* load has to be short enough to allow the output voltage to follow the modulating cycle and yet long enough to maintain a relatively high output voltage.
7. Applying Kirchhoff's voltage law to the circuit, the diode voltage$v_d$is found to be

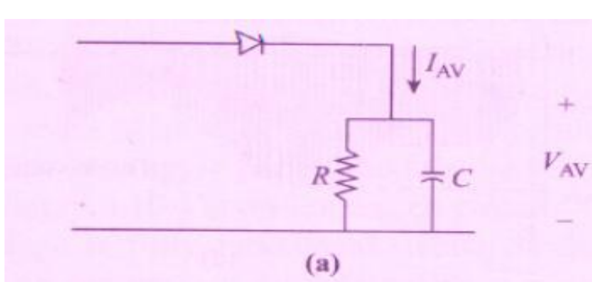
$$v_d = e - v \qquad \text{-------------------------------------} (1)$$

Where*e* is the input voltage and *v* the output voltage. Fig. 2(b) shows *e,* and Fig. 2(c) shows *v*,both for sinusoidal modulation. By graphically subtracting *v* from *e* the graph of $v_d$in Fig. 2(d) isobtained.

8. $v_d$is positive only for short periods, as indicated by the peaks v_d^+, and it is during these peaks that the capacitor is charged to make up for discharge losses.

Diagonal Peak Clipping

1. This is a form of distortion that occurs when the time constant of the RC load is too long, thus preventing the output voltage from following the modulation envelope.
2. The output voltage is labeled V_{AV} in Fig.3 (a). The curve of V_{AV} for sinusoidal modulation is shown in Fig. 3(b).



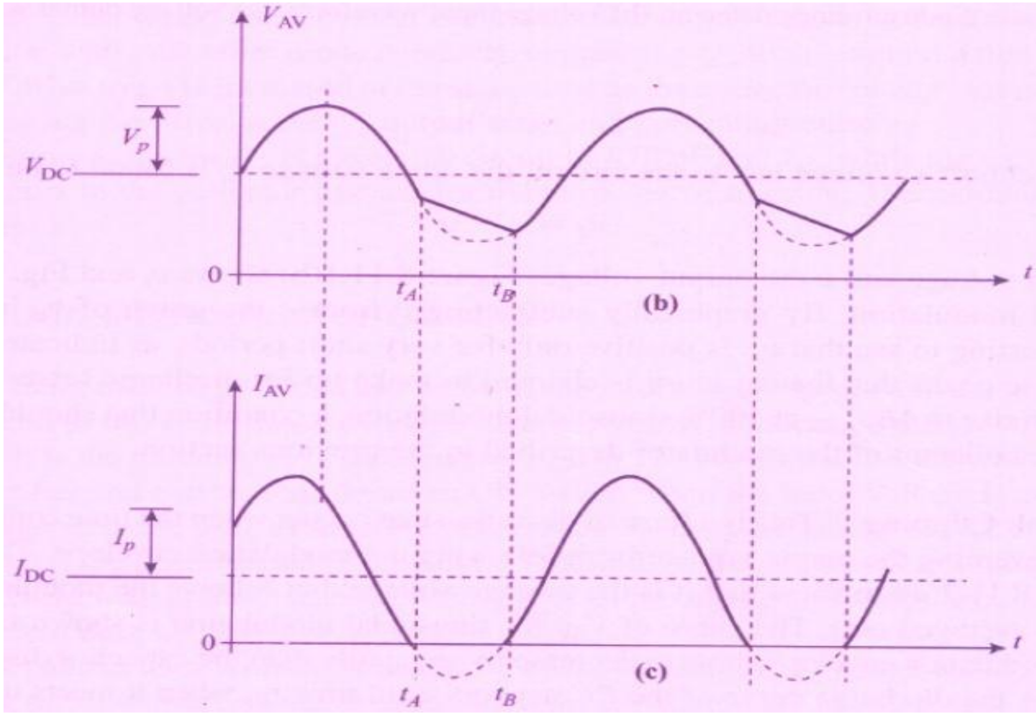


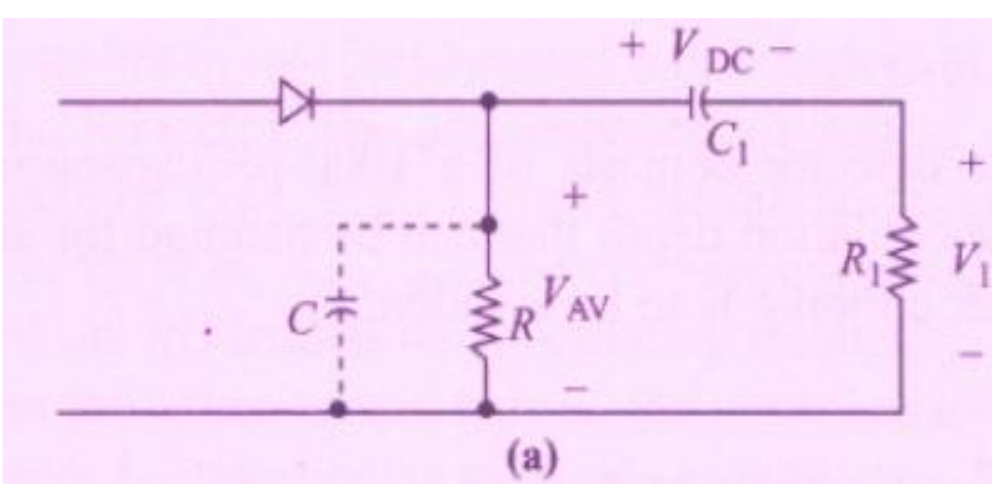
Figure 3.(a) Diode circuit supplying an average current I_{AV} to the RC load, (b) Voltage waveform , illustrating diagonal peak clipping, (c) Current waveform

3. At some time t_A the modulation envelope starts to decrease more rapidly than the capacitor discharges. The output voltage then follows the discharge curve of the RC network until time t_B, when it meets up with the modulation envelope as it once again increases.
4. For sinusoidal modulation the condition necessary for the avoidance of diagonal peak clipping is found as follows. Because of the capacitive nature of the RC load, the current leads the voltage as shown in Fig. 3(c).
5. The average current consists of two components, a dc component I_{DC} and an ac component that has a peak value I_p, as shown in Fig. 3(c). The dc component of voltage is approximately equal to the maximum unmodulated carrier voltage or $V_{DC} \cong E_{c\,max}$ and the direct current is $I_{DC} = \frac{V_{DC}}{R}$.
6. If the envelope falls faster than the capacitor discharges, the diode ceases to conduct (since the capacitor voltage biases it off), and the current I_{AV} supplied by the diode goes to zero. This is shown in Fig. 3(c).
7. During the period the current is zero, the load voltage follows the discharge law of the *RC* network, resulting in the diagonally clipped peak shown in Fig. 3(b).
   The modulation index becomes,

   $$\boldsymbol{m} \leq \frac{|\boldsymbol{Z_p}|}{\boldsymbol{R}} \qquad (Z_p = \text{impedence of the RC load at the modulating frequency})$$

<u>Negative Peak Clipping</u>

1. This is similar in appearance to diagonal peak clipping, but results from the loading effect of the network R_1C_1 following the RC load [Fig. 4(a)].



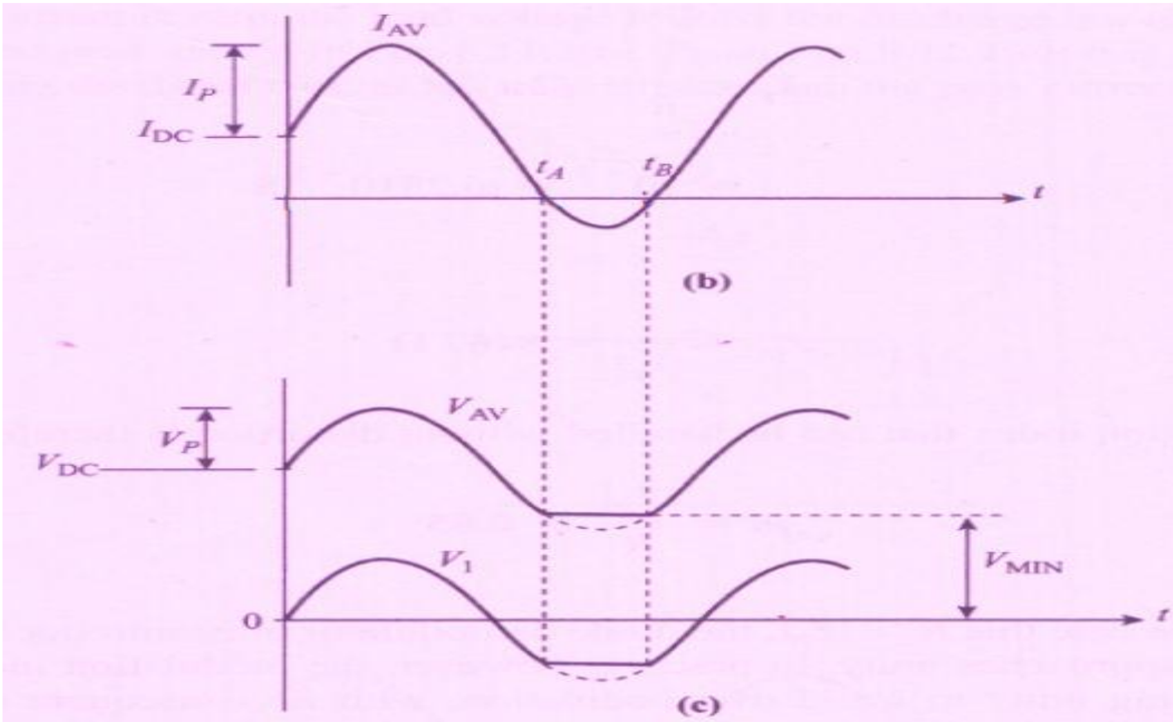


Figure 4. (a) Diode circuit including a dc blocking capacitor C_1 and the input resistor R_1 of the following stage, (b) Average diode current, (c) The voltage across R and R_1.

2. Capacitor C_1 is a dc blocking capacitor, and resistor R_1 represents the input resistance of the following stage.
3. Considering the normal situation where the reactance of C_1 is very small, and that of C is very large at the modulating frequency, the ac impedance is simply R in parallel with R_1 or $|Z_p|$ = R_p= $RR_1/(R + R_1)$. The modulation index ,
$$m \leq \frac{R_p}{R}$$
4. The current I_{AV} is in phase with V_{AV}, and over the period when I_{AV} is zero [Fig.4 (b)], the C_1 capacitor voltage remains approximately constant at V_{DC}, which in turn develops a voltage equal to $V_{MIN} = V_{DC}R/(R + R_1)$ across R. it is this voltage that keeps the diode biased off. The voltage across R and R_1 are shown in Fig. 4(c).

Generation and Demodulation of DSB-SC (Double-Sideband Suppressed-Carrier)

Generation of DSB-SC

Balanced Modulators

1. Balanced modulators are the building blocks from which a wide variety of frequency mixers, modulators, and demodulators are built.
2. Any circuit that multiplies two input signals while canceling the feedthrough of one of these is a singly balanced modulator, and one that cancels both is a doubly balanced modulator. The output contains a double-sideband suppressed carrier signal.

(i) An FET Singly Balanced Modulator Circuit

1. Fig.1 shows two matched FETs connected in a different amplifier, which acts as a singly balanced modulator in which the carrier oscillator signal is canceled from the output, but the modulating signal appears in the output.

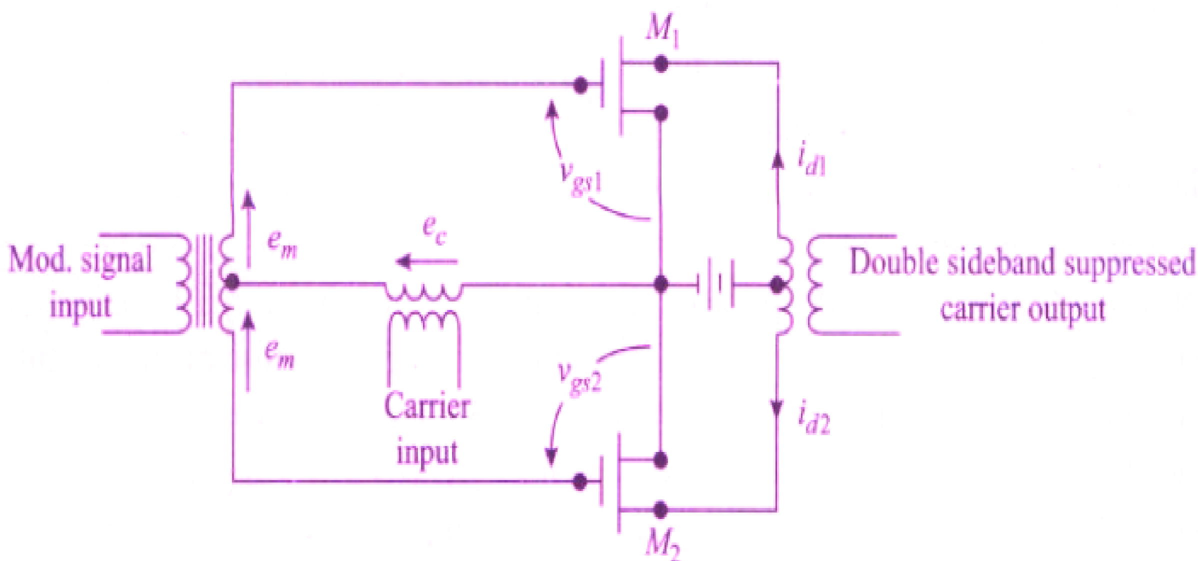


Figure 1. ***FET Singly Balanced Modulator Circuit***

2. The input(modulating) signal is applied in the differential input mode, and the carrier signal is applied as a common-mode signal.
3. The signal applied to the gate of M_1 is the sum of the two input voltages ($e_c + e_m$), while the signal applied to M_2 is the difference ($e_c - e_m$). These two components are squared by the second-order terms of the transistor transfer functions.
4. The common-mode carrier signal remaining is canceled as the two drain currents are subtracted in the output transformer primary.

$$V_{gs1} = e_c + e_m \qquad (1)$$

$$V_{gs2} = e_c - e_m \qquad (2)$$

$$i_{d1} = I_0 + a\, V_{gs1} + b\, V_{gs1}^2 \qquad (3)$$

$$i_{d2} = I_0 + a\, V_{gs2} + b\, V_{gs2}^2 \qquad (4)$$

$$i_p = i_{d1} - i_{d2} = a\,(V_{gs1} - V_{gs2}) + b\,(V_{gs1} + V_{gs2})\,(V_{gs1} - V_{gs2}) \qquad (5)$$

Substituting Eqs.(1) and (2) into (5) gives,

$$i_p = 2a\,(e_m) + 4b(e_m)(e_c) \qquad (6)$$

Substituting cosinusoidal signals in Eqn.(6) yields the output

$$\boldsymbol{i_p = 2aE_{m\,max}\cos\omega_m t + 4b(E_{m\,max}\cos\omega_m t)(E_{c\,max}\cos\omega_c t)}$$

$$\boldsymbol{i_p = 2aE_{m\,max}\cos\omega_m t + 2bE_{c\,max}E_{m\,max}[\cos(\omega_c - \omega_m)t + \cos(\omega_c + \omega_m)t]} \qquad (7)$$

5. This output contains the original modulating signal and the two sidebands about the carrier frequency position. The carrier is absent.
6. Since the output would be fed through a band-pass filter, the low-frequency modulating signal component would be removed at that point.

(ii) Doubly Balanced Diode Ring Modulator

1. A circuit known as the double-balanced ring modulator, which is widely used in carrier telephony, is shown in Fig.2(a). The name comes from the fact that the circuit is balanced to reject both the carrier and modulating signals using a ring of diodes.

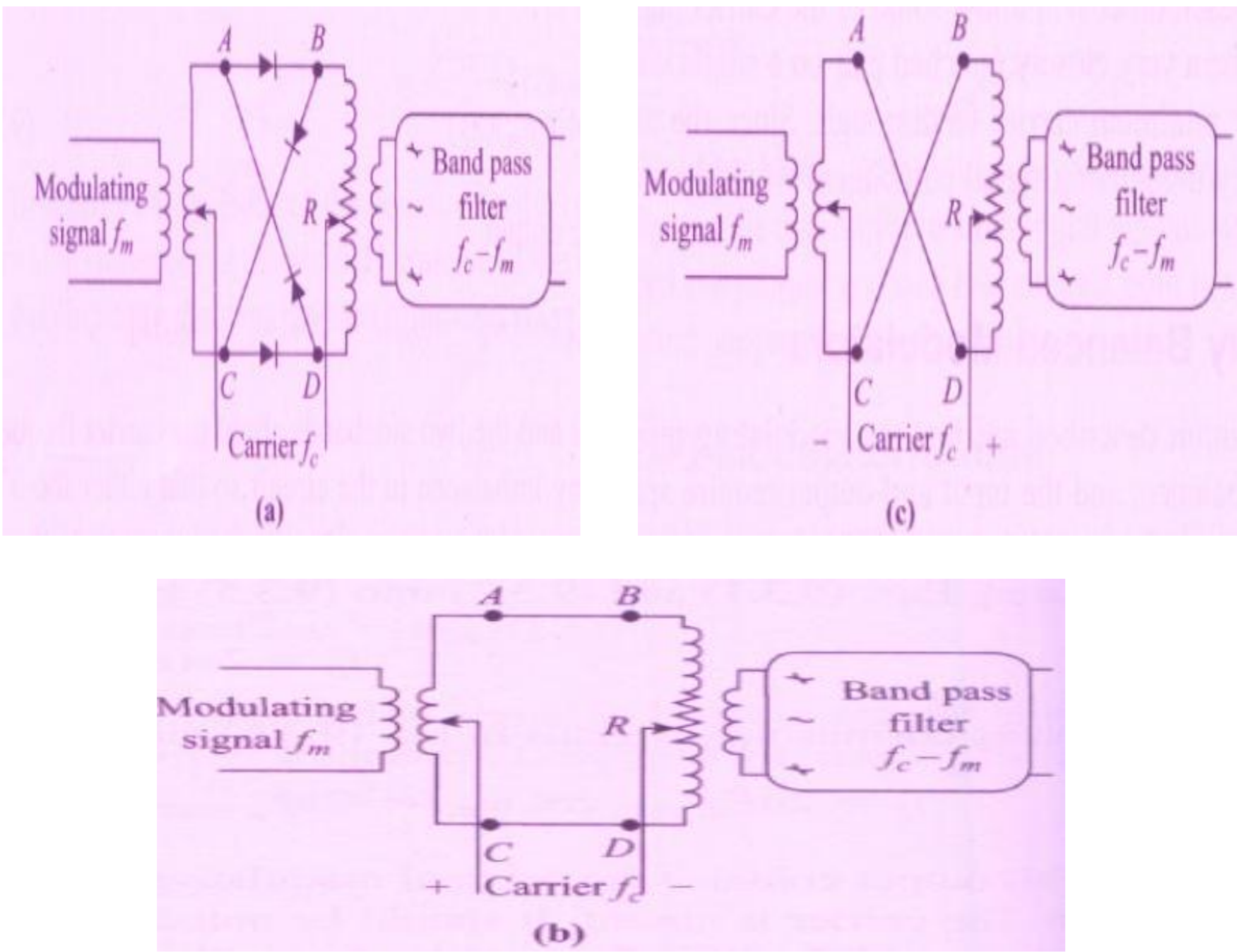


Figure 2.(a)*Doubly Balanced Diode Ring Modulator; (b) The conducting paths when diodes AB and CD are forward biased; (c)The conducting paths when diodes BC and DA are forward biased*

2. The output contains only pairs about the carrier frequency position and several of its harmonics.
3. A large signal carrier acts as a switching signal to alternate the polarity of the modulating signal at the carrier frequency.
4. With a negative carrier voltage V_c applied, diodes AB and CD conduct and diodes AD and BC block to give the effective connection shown in Fig.2(b).
5. With a positive carrier voltage, diodes AD and BC conduct and diodes AB and CD block to give the effective connection shown in Fig.2(c).
6. The effect is to multiply the modulating signal by a fixed-amplitude square wave at the carrier frequency, producing the required DSB-SC signal, with harmonics. Band-pass filters remove the unwanted harmonics from the output.

   Consider the modulating signal $V_m(t)$ and carrier signal $V_c(t)$ such that,

$$V_m(t) = V_m sin\omega_m t; \qquad V_c(t) = V_c \sin\, \omega_c t$$

   The output voltage equals the product of two signals,

$$V_o(t) = V_m sin\, \omega_m t \cdot V_c \sin\, \omega_c t$$
$$= \frac{V_m V_c}{2}[\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t]$$

7. Fig.3 shows the time response waveform for a single sinusoid of modulation and its spectrum. These circuits have been extensively used for low-frequency telephone applications, where they

require balanced input and output transformers and some adjustment of circuit balance for good performance.

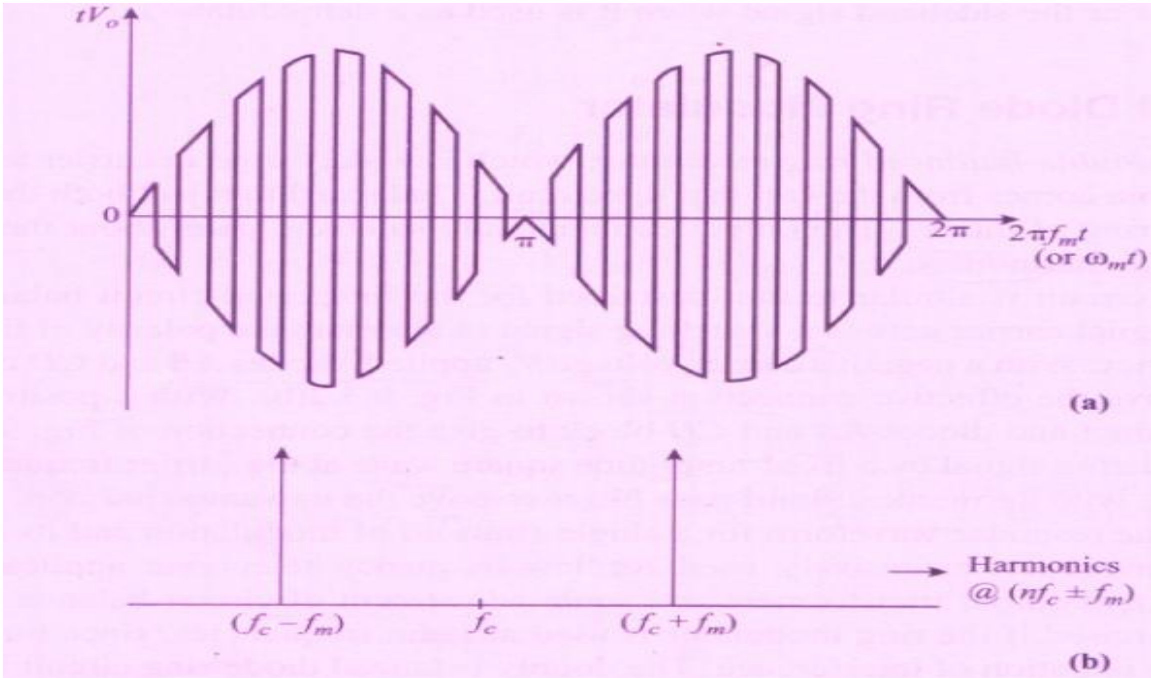


Figure 3.(a)Time waveform of a DSBSC signal for one cycle of modulation; (b)spectrum for the signal of (a).

8. The doubly balanced diode ring circuit is widely used as a mixer in microwave applications where shielded enclosures prevent radiation.

<u>Demodulation of DSB-SC</u>

<u>i) Coherent Detection</u>

1. The baseband signal m(t) can be uniquely recovered from a DSB-SC wave s(t) by first multiplying s(t) with a locally generated sinusoidal wave and the low-pass filtering the product, as in Figure.1.

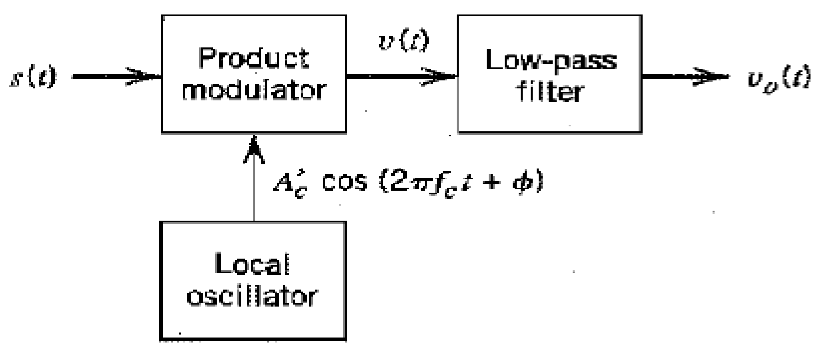


Figure 1. Coherent detector for demodulating DSB-SC modulated wave

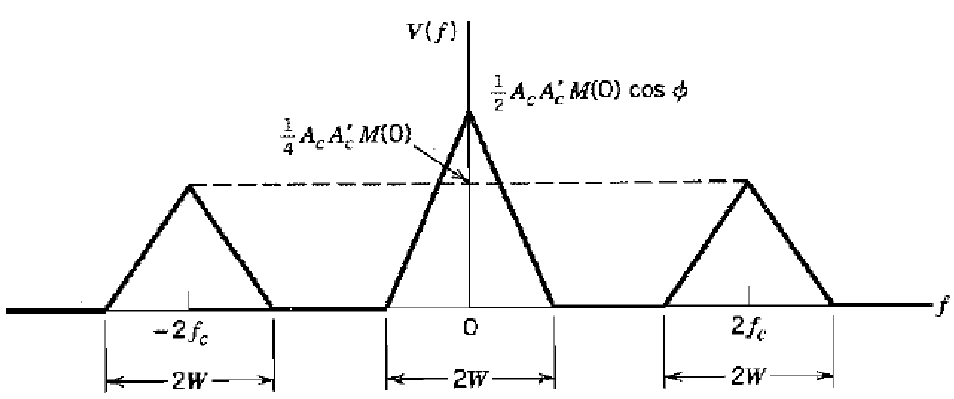


Figure 2.Illustrating the spectrum of a product modulator output with a DSB-SC modulated wave as input.

2. It is assumed that the local oscillator signal is exactly coherent or synchronized, in both frequency and phase, with the carrier wave c(t) used in the product modulator to generate s(t). This method of demodulator is known as coherent detection or synchronous demodulation.

3. Coherent detection is a special case of the more general demodulation process using a local oscillator signal of the same frequency but arbitrary phase difference Φ, measures with respect to the carrier wave c(t).
4. The product modulator output in Figure.1 is,

$$v(t) = A_c^{'} \cos(2\pi f_c t + \Phi) s(t)$$
$$= A_c A_c^{'} \cos(2\pi f_c t) \cos(2\pi f_c t + \Phi) m(t); \text{ Where s(t)= } A_c \cos(2\pi f_c t)\ m(t)$$
$$= \frac{1}{2} A_c A_c^{'} \cos(4\pi f_c t + \Phi)\, m(t) + \frac{1}{2} A_c A_c^{'} \cos \Phi\, m(t) \text{ --------- } \mathbf{(1)}$$

The first term in Equ.(1) represents a DSB-SC modulated signal with a carrier frequency $2f_c$, whereas the second term is proportional to the baseband signal m(t).

5. The baseband signal m(t) is limited to the interval $-W \le f \le W$. The first term in Eqn.(1) is removed by the low-pass filter in Figure.1, provided that the cut-off frequency of this filter is greater than W but less than $2f_c - W$.

   At the filter output we then obtain a signal given by,

$$\boldsymbol{v_0(t)} = \frac{\mathbf{1}}{\mathbf{2}} \boldsymbol{A_c A_c^{'}} \cos \Phi\, \boldsymbol{m(t)}$$

6. The demodulated signal $v_0(t)$ is therefore proportional to m(t) when the phase error Φ is a constant. The amplitude of the demodulated signal is maximum, when $\Phi = 0$, and it is minimum (zero) when $= \pm\frac{\pi}{2}$.

   The zero demodulated signal, which occurs for $\Phi = \pm\frac{\pi}{2}$, represents the *quadrature null effect* of the detector.
7. As long as the phase error Φ is constant, the detector provides an undistorted version of the original baseband signal m(t).

ii) Costas Receiver

1. One method of obtaining a practical synchronous receiver system, suitable for demodulating DSB-SC waves, it to use the Costas receiver shown in Figure 3.

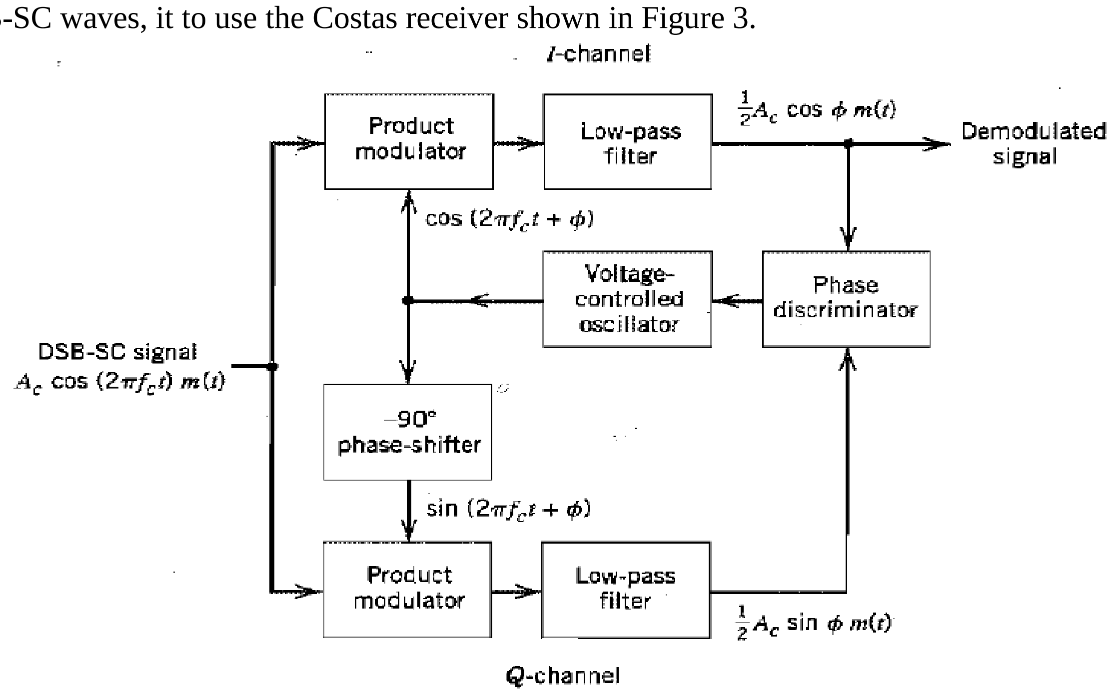


Figure 3. Costas receiver

2. This receiver consists of two coherent detectors supplied with the same input signal, namely, the incoming DSB-SC wave$A_c\cos(2\pi f_c t)\, m(t)$, but with individual local oscillator signals that are in phase quadrature with respect to each other. The frequency of the local oscillator is adjusted to be the same as the carrier frequencyf_c, which is assumed known a priori.
3. The detector in the upper path is referred to as the in-phase coherent detector or I-Channel, and that in the lower path is referred to as the quadrature-phase coherent detector or Q-Channel.

   These two detectors are coupled together to form a negative feedback system designed in such a way as to maintain the local oscillator synchronous with the carrier wave.
4. To understand the operation of the receiver, suppose that the local oscillator signal is of the same phase as the carrier wave $A_c\cos(2\pi f_c t)$ used to generate the incoming DSB-SC wave, the I-channel output contains the desired demodulated signal m(t), whereas the Q-channel output is zero due to the quadrature null effect of the Q-channel.

5. If the local oscillator phase drifts from its proper value by a small angle Φ radians. The I-channel output will remain unchanged, but there will now be some signal appearing at the Q-channel output, which is proportional to $\sin \Phi \approx \Phi$ for small Φ.
6. Thus, by combining the I-channel and Q-channel outputs in a phase discriminator (which consists of a multiplier followed by a low-pass filter), as shown in Figure 3, a DC control signal is obtained that automatically corrects the local phase errors in the voltage-controlled oscillator.
7. Phase control in the Costas receiver ceases with modulation and that phase-lock has to be reestablished with the reappearance of modulation. This is not a serious problem when receiving voice transmission.

Generation and Demodulation of SSB-SC (Single-Sideband Suppressed-Carrier)

SSB Generation

i) Balanced Modulator-Filter Method

1. SSB transmitters used balanced modulator circuits to generate DSB-SC signals followed by sideband filters to remove the unwanted sidebands. Such a transmitter is shown in Fig.(1).

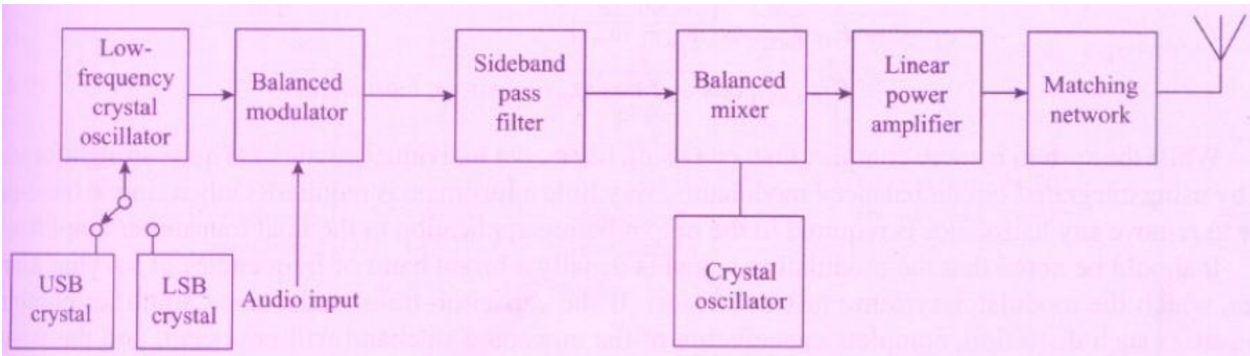


Figure 1. Single sideband suppressed carrier transmitter using band-pass filter to eliminate the unwanted sideband.

2. Initial modulation takes place in the balanced modulator at a low frequency, because of the making adequate filters at higher frequencies.
3. The filter is a band-pass filter with a sharp cutoff at each side of the band-pass to obtain satisfactory adjacent sideband rejection.
4. In this case, a single-sideband filter is used and the carrier oscillator crystal is switched to place the desired sideband in the filter window. Alternatively, two sideband filters could be used with a fixed carrier frequency.
5. The filtered signal is up-converted in a mixer (the second balanced modulator) to the final transmitter frequency and then amplified before being coupled to the antenna. Linear power amplifiers are used to avoid distorting the sideband signal.

ii)Phasing Method

1. Fig.(2) shows a different means of obtaining an SSBSC signal. This circuit does not have any sideband filters, and the primary modulation can be done at the transmitting frequency. It relies on phase shifting and cancellation to eliminate the carrier and the unwanted sideband.

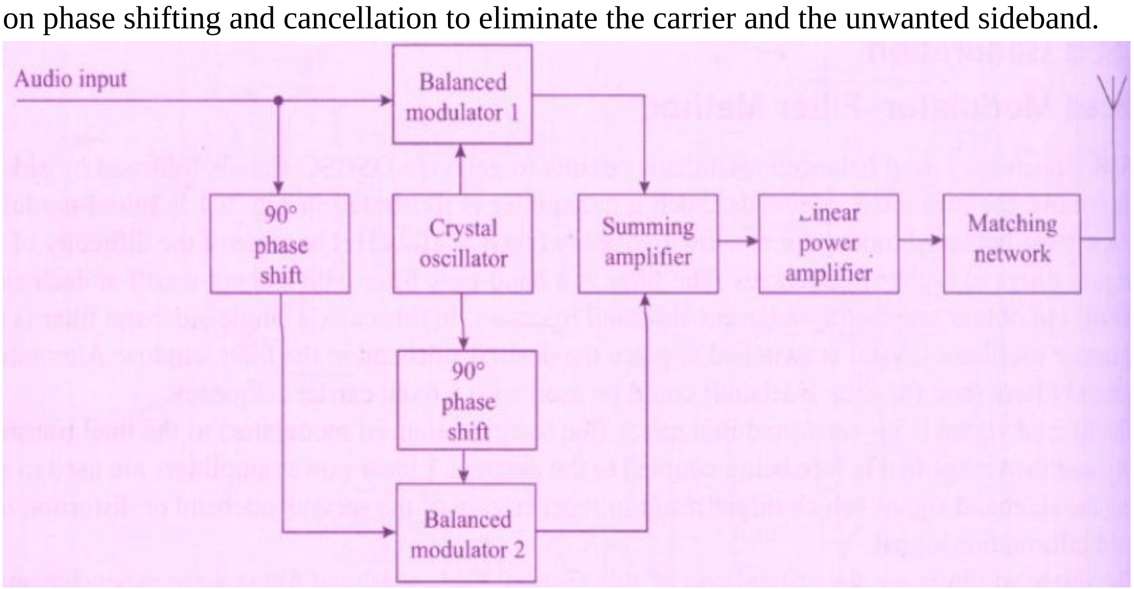


Figure 2.Single sideband suppressed carrier transmitter using phase shift to obtain cancellation of sidebands.

2. Assume cosinusoidal signals for both carrier and modulation and that the circuit shown produces the lower side frequency, given by

$$e_{LSF} = E_{L\,max}\ \cos(\omega_c - \omega_m)t \qquad (1)$$

The standard trigonometric identity for the difference of two angles gives

$$e_{LSF} = E_{L\,max}\ [\cos\omega_c t \cos\omega_m t + \sin\omega_c t \sin\omega_m t\ ] \qquad (2)$$

But, $\sin\omega_c t = \cos(\omega_c t - \frac{\pi}{2})$ and $\sin\omega_m t = \cos(\omega_m t - \frac{\pi}{2})$ (3)

Therefore,

$$e_{LSF} = E_{L\,max}\ [\cos\omega_c t\cos\omega_m t + \cos(\omega_c t - \tfrac{\pi}{2})\cos(\omega_m t - \tfrac{\pi}{2})] \quad (4)$$

3. The first term on the right of Eq.(4) is the result of balanced modulator 1, which multiplies the two unshifted signals. The second term is the result of balanced modulator 2, which multiplies the two signals each shifted by -90°.
4. The -90° shift for the carrier is easily accomplished by feeding the signal through a controlled current source into a capacitor. The phase shifting network for the baseband signal must accurately provide a constant 90° phase shift over a wide frequency range.
5. The carrier signal is canceled out in this circuit by both of the balanced modulators, and the unwanted sidebands cancel at eh output of the summing amplifier.
6. If the two outputs are subtracted instead of added the upper sideband will result,

$$e_{USF} = E_{U\,max}\ \cos(\omega_c + \omega_m)t$$
$$e_{USF} = E_{U\,max}\ [\cos\omega_c t\cos\omega_m t - \sin\omega_c t\sin\omega_m t]$$

iii)Modified Phase Shift method or Weavers method

1. It is similar to the phase shifting method presented previously, but it differs in that the modulating signal is first modulated on a low-frequency subcarrier, which is then modulated onto the high-frequency carrier.
2. The circuit connections for generating an LSB signal are shown in Fig. (3). Modulators BM1 and BM2 both have the unshifted modulating signal as inputs.

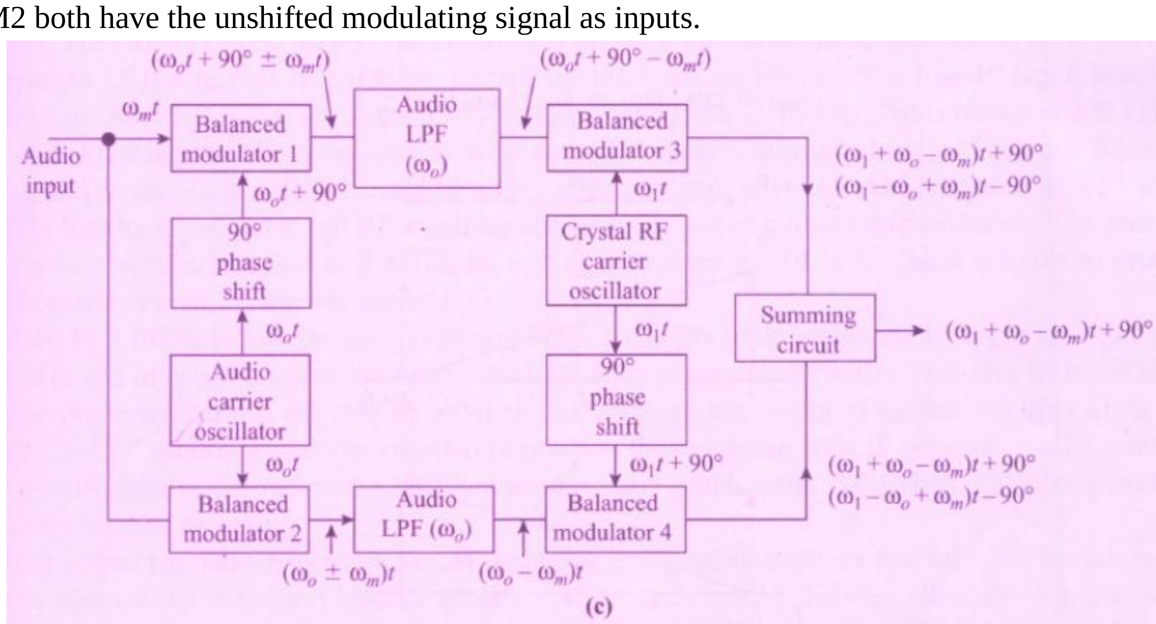


Figure 3. The third method of generating an SSBSC signal

3. BM1 also takes the low-frequency subcarrier with a 90° shift introduced in it from the oscillator signal.
4. BM2 takes the subcarrier signal directly from the oscillator.
5. Assuming unity magnitudes and cosinusoidal single-frequency modulation, the output from BM1 becomes,

$$e_{BM1} = \cos\left(\omega_0 t + \frac{\pi}{2}\right)\cos\omega_m t \quad (1)$$
$$= \frac{1}{2}\left[\cos\left(\omega_0 t + \omega_m t + \frac{\pi}{2}\right) + \cos\left(\omega_0 t - \omega_m t + \frac{\pi}{2}\right)\right] \quad (2)$$

And the output of BM2 becomes

$$e_{BM2} = \cos\omega_0 t\ \cos\omega_m t \quad (3)$$
$$= \frac{1}{2}[\cos(\omega_0 t + \omega_m t) + \cos(\omega_0 t - \omega_m t)] \quad (4)$$

6. Low-pass filters with a cutoff frequency set at the subcarrier frequency f_0 removes the sum term from each of the above signals, leaving only the second terms as inputs to BM3 and BM4.
7. These signals are the lower sidebands on f_0. They are identical except that the signal applied to BM3 is shifted by $+90^\circ$ from that applied to BM4.
8. This process eliminates the need to provide a wideband 90° phase shifting network for the baseband signals, as was the case for the phase shifting method.

9. The high-frequency oscillator signal at f_1 is applied directly to BM3, but it is shifted by $+90^\circ$ before being applied to BM4. The output from BM3 becomes,

$$e_{BM3} = \cos\omega_1 t \cos\left((\omega_0 - \omega_m)t + \frac{\pi}{2}\right) \quad (5)$$

$$e_{BM3} = \frac{1}{2}\left[\cos\left(\omega_1 t + \left((\omega_0 - \omega_m)t + \frac{\pi}{2}\right)\right) + \cos\left(\omega_1 t - \left((\omega_0 - \omega_m)t + \frac{\pi}{2}\right)\right)\right] \quad (6)$$

$$e_{BM3} = \frac{1}{2}\left[\cos\left((\omega_1 + \omega_0)t - \omega_m t + \frac{\pi}{2}\right) + \cos\left((\omega_1 - \omega_0)t + \omega_m t - \frac{\pi}{2}\right)\right] \quad (7)$$

and the output of BM4 becomes

$$e_{BM4} = \cos\left(\omega_1 t + \frac{\pi}{2}\right)\cos(\omega_0 - \omega_m)t \quad (8)$$

$$e_{BM4} = \frac{1}{2}\left[\cos\left(\left(\omega_1 t + \frac{\pi}{2}\right) + (\omega_0 - \omega_m)t\right) + \cos\left(\left(\omega_1 t + \frac{\pi}{2}\right) - (\omega_0 - \omega_m)t\right)\right] \quad (9)$$

$$e_{BM4} = \frac{1}{2}\left[\cos\left((\omega_1 + \omega_0)t - \omega_m t + \frac{\pi}{2}\right) + \cos\left((\omega_1 - \omega_0)t + \omega_m t + \frac{\pi}{2}\right)\right] \quad (10)$$

10. The first terms in Eqs.(7) and (10) are identical lower sidebands on an offset carrier frequency $f_c = f_1 + f_0$. The second terms are the upper sidebands on an offset carrier at $f_c = f_1 - f_0$, but are 180° out of phase with each other.
11. The oscillator frequency f_1 must be adjusted so that the output carrier frequency f_c and the desired sideband fall in the correct position in the output frequency spectrum.
12. Usually, f_0 is chosen to fall at the midpoint of the modulating signal baseband, so that $f_0 = W/2$. The result is that both the USB and LSB spectrums are centered on f_1, with the carrier position f_c for the LSB located at the upper edge of the pass-band and that for the USB at the lower edge. This is illustrated in Fig.(4).

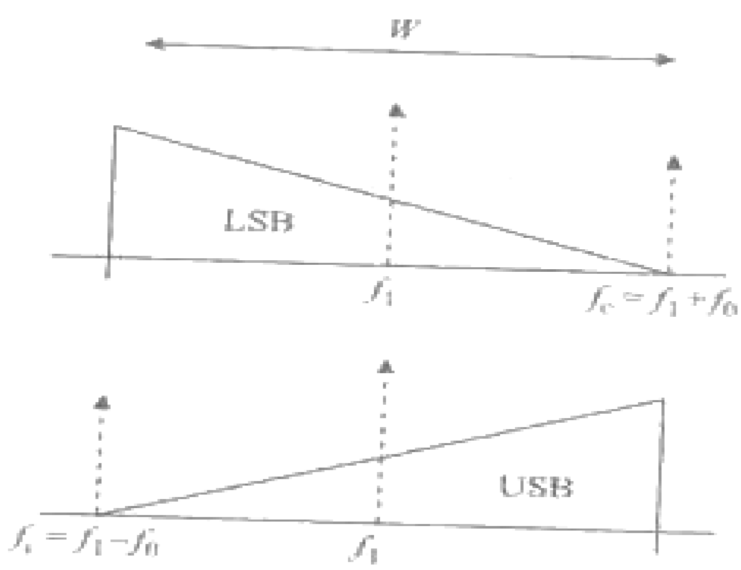


Figure 4. Output spectra for the "Modified Phase Shift" circuit (a) for LSB and (b) for USB

13. The outputs from BM3 and BM4 are added in a summing amplifier to produce the final output. The first two terms add, but the second two cancel, leaving the output as,

$$\boldsymbol{e_{out} = \cos\left((\omega_1 + \omega_0 - \omega_m)t + \frac{\pi}{2}\right)}$$

14. This is the lower sideband on the carrier frequency $(f_1 + f_0)$. The $+90^\circ$ shift in the output is of no consequence since the original carrier has been eliminated. This signal may be applied to a linear power amplifier and antenna for radiation.
15. If the output from BM3 (or BM4) is inverted before the input to the adder, the phasing becomes such that the first terms cancel and the second terms add, giving the upper sideband on the carrier frequency $(f_1 - f_0)$.

SSB Demodulator

Coherent Detection

1. Coherent carrier at receiver-end easily recovers baseband signal m(t) by multiplying and passing it through a low pass filter, (Fig.1)

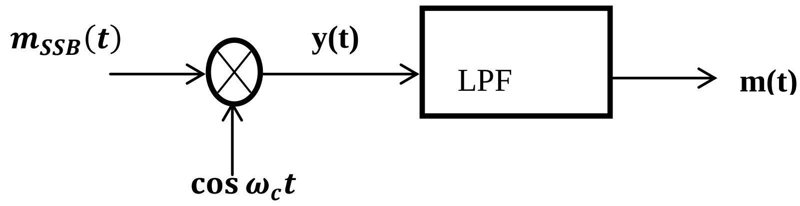


Figure 1.SSB demodulation

$$m_{SSB}(t) = \frac{1}{2}[m(t)\cos\omega_c(t) \pm m_h(t) sin\, \omega_c(t)] \quad (1)$$

$$y(t) = m_{SSB}(t) cos \omega_c(t) \quad (2)$$

$$= \frac{1}{2}[m(t)\cos^2\omega_c(t) \pm m_h(t)\, sin\, \omega_c(t)\cos\omega_c(t)] \quad (3)$$

$$= 0.25\, m(t)[1 + \cos 2\omega_c(t)] \pm 0.25 m_h(t) sin2\, \omega_c(t) \quad (4)$$

2. This signal when passed through a LPF of bandwidth that of m(t) suppresses components around $2\omega_c(t)$ and recovers the first term, the message signal.

   The output as,

   y(t) = $\mathbf{0.25\, m(t)[1 + \cos 2\omega_c(t)]}$

Generation and Detection of Vestigial Sideband Modulation (VSB)

Generation of VSB:

1. In vestigial sideband (VSB) modulation, one of the sidebands is partially suppressed and a vestige of the other sideband is transmitted to compensate for that suppression.
2. VSB modulation is something in between SSB and DSB-SC modulation which provides certain advantages at small cost. This is so called because a vestige or appendage is added here to SSB spectrum.
3. Here the difficulty of SSB generation, i.e. requirement of a sharp cut-off filter or phase shifter is avoided and also the spectrum requirement is not as high as DSB-SC. The additional spectrum required is usually less than one-fourth of SSB requirement.
4. A popular method for generating a VSB-modulated wave is to use the frequency discrimination method.
5. First, we generate a DSB-SC modulated wave and then pass it through a band-pass filter, as shown in Figure.1; it is the special design of the band-pass filter that distinguishes VSB modulation from SSB modulation.

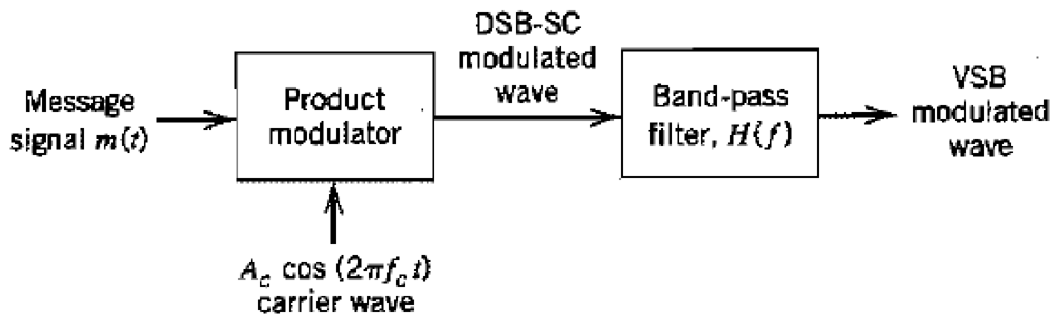


Figure 1. Filtering scheme for the generation of VSB modulated wave.

6. Let us now mathematically represent a VSB signal. We use Figure.2 for it where we consider generating VSB signal by filtering DSB-SC signal with $H(\omega)$.

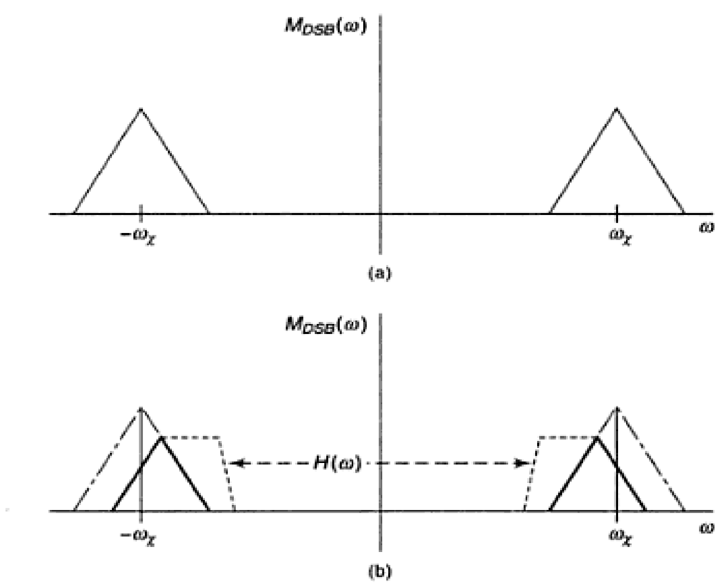


Figure 2. VSB modulation from DSB-SC modulation

DSB-SC signal spectrum, $M_{DSB}(\omega)=\frac{1}{2}[M(\omega+\omega_c)+M(\omega-\omega_c)]$ **(1)**

Then, VSB spectrum, $\mathbf{M_{VSB}(\omega)} = \boldsymbol{H(\omega)}\mathrm{MDSB}(\omega) = \frac{1}{2}\boldsymbol{H(\omega)[M(\omega+\omega_c)+M(\omega-\omega_c)]}$ **(2)**

Detection

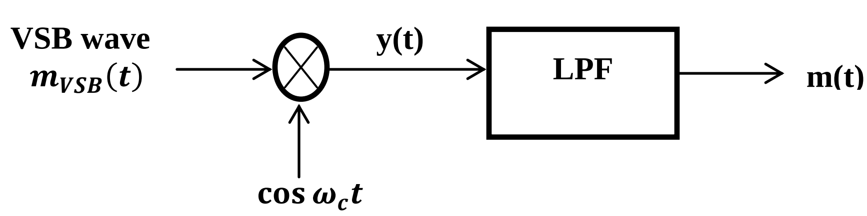


Figure 3.VSB demodulation

1. In the demodulation process, the multiplier output becomes,

$$y(t) = m_{VSB}(t)\cos\omega_c t \quad \textbf{(3)}$$

2. By modulation theorem, the Fourier transform of above equation becomes,

$$Y(\omega) = \frac{1}{2}[M_{VSB}(\omega+\omega_c) + M_{VSB}(\omega-\omega_c)] \quad \textbf{(4)}$$

$$= \frac{1}{4}[H(\omega+\omega_c)M(\omega+\omega_c+\omega_c) + H(\omega+\omega_c)M(\omega-\omega_c+\omega_c)] +$$

$$\frac{1}{4}[H(\omega-\omega_c)M(\omega+\omega_c-\omega_c) + H(\omega-\omega_c)M(\omega-\omega_c-\omega_c)] \text{ (usingEqu.(2))} \quad \textbf{(5)}$$

$$= \frac{1}{4}\,[\,H(\omega+\omega_c) + H(\omega-\omega_c)]M(\omega) +$$

$$\frac{1}{4}[\,H(\omega+\omega_c)M(\omega+2\omega_c) + H(\omega-\omega_c)M(\omega-2\omega_c)\,] \quad \textbf{(6)}$$

3. The low pass filter removes the second term which is centered around $2\omega_c$ and has a transfer function $H_R(\omega)$. Then final demodulated output will be,

$$\boldsymbol{\frac{1}{4}H_R(\omega)[\,H(\omega+\omega_c) + H(\omega-\omega_c)]\,M(\omega)} \quad \textbf{(7)}$$

4. If, $H_R(\omega)[H(\omega+\omega_c) + H(\omega-\omega_c)]$ is constant, i.e. $|\omega|<2\pi B$, then the output of LPF is $km(t)$ where k is a constant. Again for $H_R(\omega) = 1$ in baseband, (usual for LPF) we need $[H(\omega+\omega_c)H(\omega-\omega_c)]$ = constant and this condition requires a band pass filter $H(\omega)$ to be applied on DSB-SC signal which is antisymmetric or odd-symmetric about ω_c.

Comparison of Various AM systems

S.No	Parameter	AM with carrier (DSB)	DSB-SC	SSB-SC	VSB
1.	**Method**	Carrier and both sidebands	Only sidebands	Only one sideband	One sideband and part of the other sideband
2.	**Bandwidth**	$2f_m$	$2f_m$	f_m	f_m<BW< $2f_m$
3.	**Generation**	Easy	Easy	Complex	Complex
4.	**Transmission Efficiency**	33.3%	100%	100%	33.3% <η <100%
5.	**Selective fading**	Heavy distortion in received signal	More distortion compared to SSB-SC	Least distortion	Received signal is distorted.
6.	**Signal to Noise Ratio**	$\left(\frac{S}{N}\right)_0 = \frac{2}{3}\left(\frac{S}{N}\right)_I$	$\left(\frac{S}{N}\right)_0 = 2\left(\frac{S}{N}\right)_I$	$\left(\frac{S}{N}\right)_0 = \left(\frac{S}{N}\right)_I$	$\left(\frac{S}{N}\right)_0 = \left(\frac{S}{N}\right)_I$
7.	**Application**	AM broadcast application	Carrier Telephony	Wireless mobile	Television and high speed data transmission

Generation of FM

1. There are essentially two basic methods of generating frequency modulated signal namely, **direct FM and indirect FM**.
2. In the **direct method** the carrier frequency is directly varied in accordance with the input baseband signal, which is readily accomplished using a voltage-controlled oscillator.
3. In the **indirect method**, the modulating signal is first used to produce a narrowband FM signal, and frequency multiplication is next used to increase the frequency deviation to the desired level.
4. The indirect method is the preferred choice for frequency modulation when the stability of carrier frequency is of major concern as in commercial radio broadcasting.

Direct FM

Direct FM can be obtained by using FET and varactor diode.

FET Reactance Modulator

1. Fig.1 shows the basic circuit of FET reactance modulator. It behaves as reactance across terminals A-B. The terminals A-B of the circuit may be connected across the tuned circuit of the oscillator to get FM output.

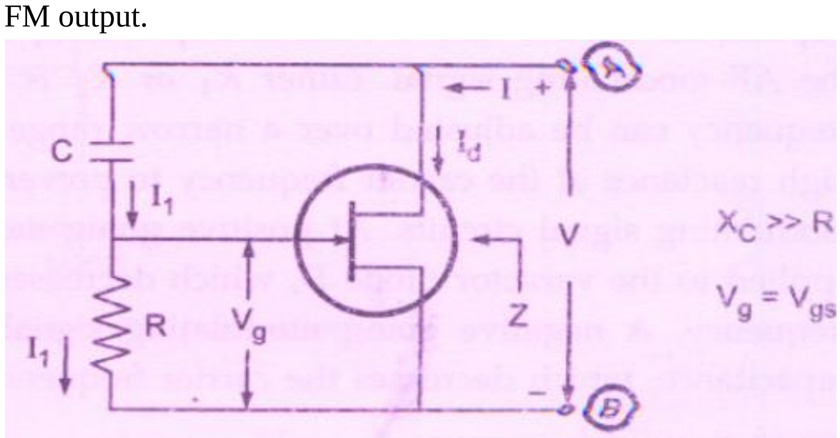


Figure 1. FET reactance modulator

2. The varying voltage (modulating voltage)V, across terminals A-B changes reactance of the FET. This change in reactance can be inductive or capacitive.
3. Neglecting gate current, let the current through C and R be I_1. At the carrier frequency, the reactance of 'C' is much larger than R. We can write equation for I_1 as,

$$I_1 = \frac{V}{R+\frac{1}{j\omega C}} \tag{1}$$

Since $j\omega C \gg R$, we can write above equation as,

$$I_1 = j\omega CV \tag{2}$$

From the circuit, $$V_g = I_1\, R = j\omega CRV \tag{3}$$

For the FET, $$I_d = g_m V_{gs} = g_m V_g \tag{4}$$

Sub Eq.(3) in Eq.(4) $$I_d = j\omega CR g_m V \tag{5}$$

From the circuit, impedance of the FET is,

$$\mathbf{Z}= \frac{V}{I_d} = \frac{V}{j\omega CR\, g_m V} = \frac{1}{j\omega[g_m\, CR]} = \frac{1}{j\omega c_{eq}} \tag{6}$$

Here $c_{eq} = g_m\, CR$. Thus the impedance of FET is capacitive reactance. By varying the modulating voltage across FET, the operating point g_m can be varied. Hence this variesc_{eq} . This change in the capacitance will change the frequency of the oscillator. If we connect inductance instead of capacitor, we get inductive reactance in the circuit.

2.3.1.2. Frequency Modulation using Varactor Diode

1. All the diodes exhibit small junction capacitance in the reverse biased condition. The varactor diodes are specially designed to optimize this characteristic.
2. The junction capacitance of the varactor diode changes as the reverse bias across it is varied. The variations in capacitance of this diode are wide and linear. The varactor diodes provide the junction capacitance in the range of 1 to 200 PF.
3. Fig. 2 shows how varactor diode can be used to generate FM. $L_1$and C_1 form the tank circuit of the carrier oscillator. The capacitance of the varactor diode depends upon the fixed bias set by $R_1$and R_2 and the AF modulating signal.
4. Either R_1 or R_2 is made variable so that the center carrierfrequency can be adjusted over a narrow range.

5. The Radio Frequency Choke (RFC) has high reactance at the carrier frequency to prevent the carrier signal from getting into the modulating signal circuits.
6. At positive going modulating signal adds to the reverse bias applied to the varactor diode D, which decreases its capacitance and increases the carrier frequency.
7. A negative going modulating signal subtracts from the bias, increasing the capacitance, which decreases the carrier frequency.

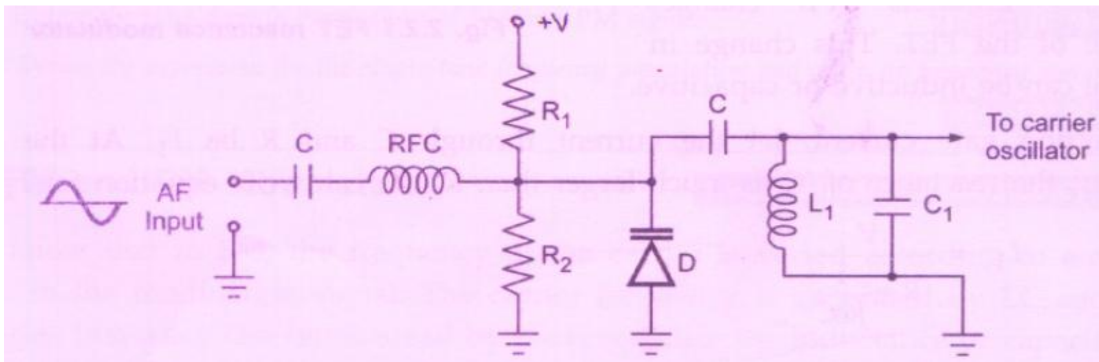


Figure.2 Varactor diode for FM generation

The frequency of the LC oscillator changes due to temperature effects. Hence crystals are used in FM generators to provide frequency stability.

Armstrong Method FM (Wideband FM generation using indirect method)

1. A simplified block diagram of an indirect FM system is shown in Figure 3. The message (baseband) signal m(t) is first integrated and then used to phase-modulate a crystal-controlled oscillator; the use of crystal control provides frequency stability.

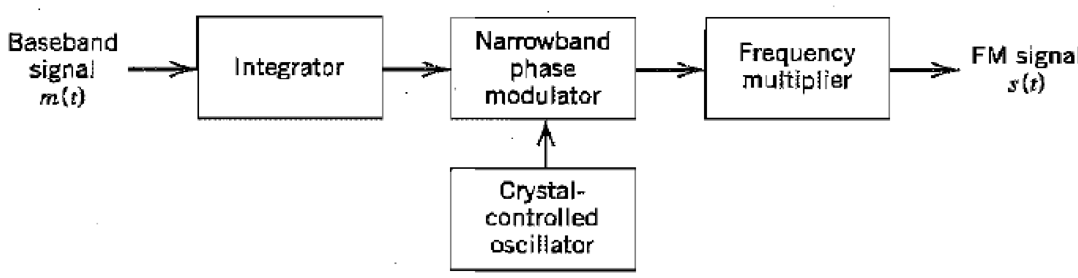


Figure 3. Armstrong method of WBFM generation

2. To minimize the distortion inherent in the phase modulator, the maximum phase deviation or modulation index β is kept small, there by resulting in a narrowband FM signal.
3. The narrowband FM signal is next multiplied in frequency by means of a frequency multiplier so as to produce the desired wideband FM signal.
4. A frequency multiplier consists of a nonlinear device followed by a band-pass filter, as shown in Fig. 4. The implication of the non-linear device being memoryless is that it has no energy-storage elements.

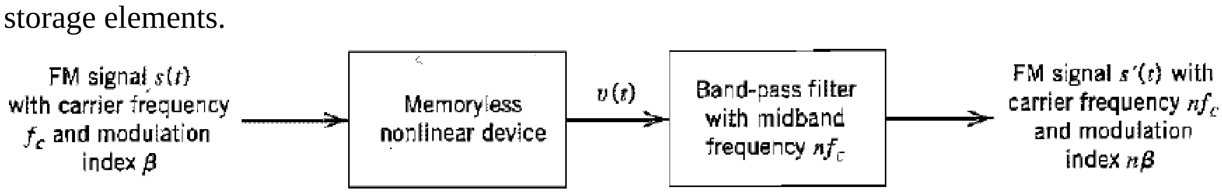


Figure 4. Block diagram of frequency multiplier

5. The input-output relation of such a device may be expressed in the general form,

$$v(t) = a_1 s(t) + a_2 s^2(t) + \cdots . + a_n s^n(t) \qquad (1)$$

Where $a_1, a_2, \dots \dots, a_n$ are coefficients determined by the operating point of the device, and n is the highest order of non-linearity.

6. The input s(t) is an FM signal defined by,

$$s(t) = A_c \cos\left[2\pi f_c t + 2\pi k_f \int_o^t m(\tau)d\tau\right] \qquad (2)$$

Whose instantaneous frequency is,

$$f_i(t) = f_c + k_f m(t) \qquad (3)$$

The mid-band frequency of the band-pass filter in Figure.4 is set equal to nf_c, where f_c is the carrier frequency of the incoming FM signal s(t).

7. Moreover, the band-pass filter is designed to have a bandwidth equal to n times the transmission bandwidth of s(t).
8. After band-pass filtering of the non-linear device's output v(t), we have a new FM signal defined by,

$$\boldsymbol{s'(t) = A'_c \cos\left[2\pi n f_c t + 2\pi n k_f \int_o^t m(\tau)d\tau\right]} \qquad \textbf{\textit{(4)}}$$

Whose instantaneous frequency is,

$$f_i'(t) = nf_c + nk_f m(t) \qquad (5)$$

Thus comparing the Equations.(3) and(5), we see that the nonlinear processing circuit of Figure.4 act as a frequency multiplier.

The frequency multiplication ratio is determined by the highest power n in the input-output relation of Equ.(1), characterizing the memoryless nonlinear device.

Demodulation of FM :

1. The FM receivers are also superheterodyne receivers. But they have different types of demodulators or detectors.
2. FM receivers have amplitude limiters which are absent in AM receivers. The AGC system of FM receiver is different than that of AM receivers. RF amplifiers, mixers, local oscillators IF amplifiers, audio amplifiers etc. all are present inFM receivers.
3. The detection of FM is totally different compared to AM. The FM detector should be able to produce the signal whose amplitude is proportional to the deviation inthe frequency of FM signal. Thus the job of FM detector is almost similar to frequency to voltage converter. FM detectors are Slope detectors, phase discriminator and ratio detector.

2.3.3 Round-Travis Detector or Balanced Slope Detector (Frequency Discriminator)

1. Fig.5. shows the circuit of balanced slope detector. It consists of two identical circuits connected back to back.

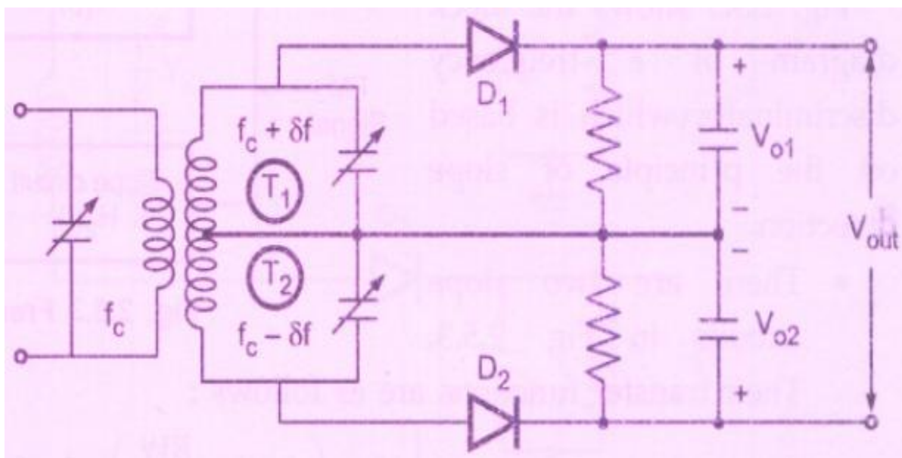


Figure 5. Balanced slope detector

2. The FM signal is applied to the tuned LC circuit. Two tuned LC circuits are connected in series. The inductance of this secondary tuned LC circuit is coupled with, the inductance of the primary (or input side) LC circuit. Thus it forms a tuned transformer.
3. In Fig.5, the upper tuned circuit is shown as T_1 and lower tuned circuit is shown as T_2. The input side LC circuit is tuned to f_c, carrier frequency.

   $T_1$is tuned to$f_c + \delta f$, which represents highest frequency. And lower LC circuit T_2 is tuned to $f_c - \delta f$, which represents the minimum frequency of FM signal.

   The input FM signal is coupled to T_1 and T_2 180° out of phase. The secondary side tuned circuits ($T_1$and T_2) are connected to diodes $D_1$and D_2 with RC loads. The total output V_{out} is equal to difference between V_{01} and V_{02}, since they subtract (See Fig.5).
4. Fig. 6 shows the characteristic of the balanced slope detector. It shows V_{out} with respect to input frequency.

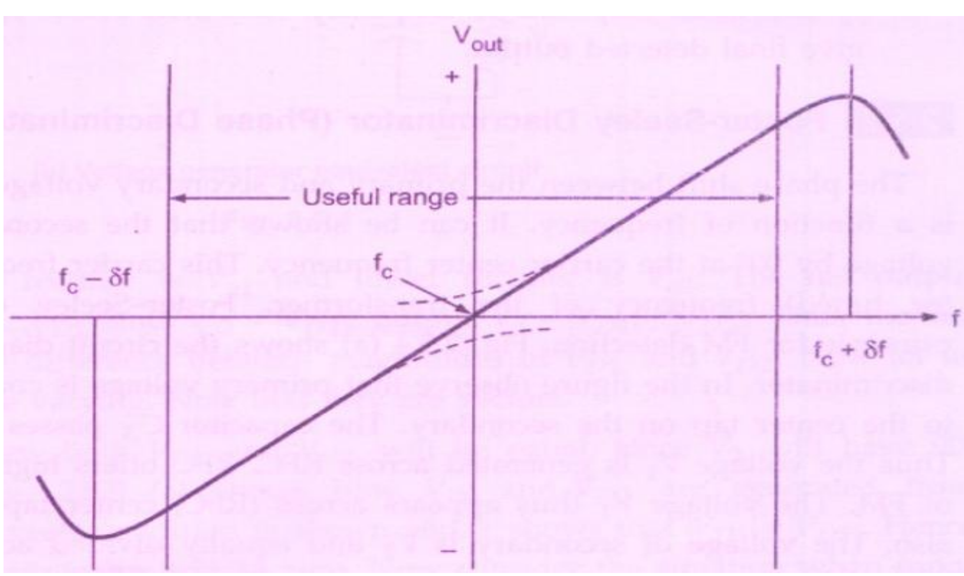


Figure 6. Characteristics of balanced slope detector, or 'S' curve

5. When the input frequency is equal to f_c, both T_1 and T_2 produce the same voltage. Hence V_{01} and V_{02} are identical and they subtract each other. Therefore V_{out} is zero. This is shown in Fig. 6.
6. When the input frequency is $f_c + \delta f$, the upper circuit T_1 produces maximum voltage since it is tuned to this frequency (i.e $f_c + \delta f$).

   Whereas lower circuit T_2 is tuned to $f_c - \delta f$, which is quite away from $f_c + \delta f$. Hence T_2 produces minimum voltage. Hence the output V_{01} is maximum where V_{02} is minimum. Therefore $\boldsymbol{V_{out}}$ = $\boldsymbol{V_{01} - V_{02}}$ **is maximum positive for** $\boldsymbol{f_c + \delta f}$.
7. When the input frequency is $f_c - \delta f$, the lower circuit T_2 produces maximum signal since it is tuned to this frequency (i.e $f_c - \delta f$).

   But upper circuit T_1 produces minimum signal. Hence rectified outputs V_{02} is maximum and V_{01} is minimum. Therefore output $\boldsymbol{V_{out}}$ =$\boldsymbol{V_{01} - V_{02}}$ **is maximum negative for** $\boldsymbol{f_c - \delta f}$. This is shown in Fig. 6.
8. For the other frequencies of input, the output (V_{out}) is produced according to the characteristic shown in Fig. 6. For example if input frequency tries to increase above f_c then V_{01} will be greater than V_{02} and the net output V_{out} will be positive.
9. The linearity of the characteristics depends upon alignment of tuning circuits and coupling characteristics of the tuned coils.

2.3.4 Frequency Discriminator

1. Fig. 7 shows the block diagram of a frequency discriminator which is based Signal on the principle of slope detection.

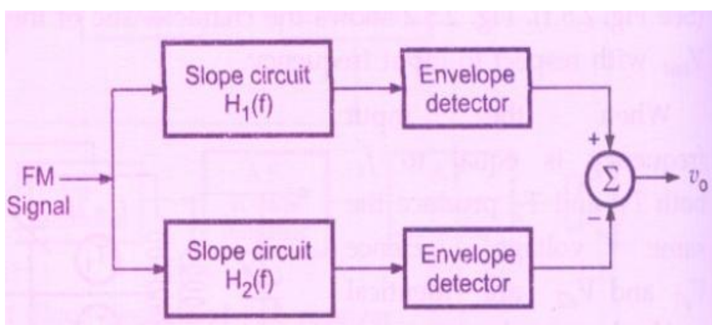


Figure 7. Frequency discriminator

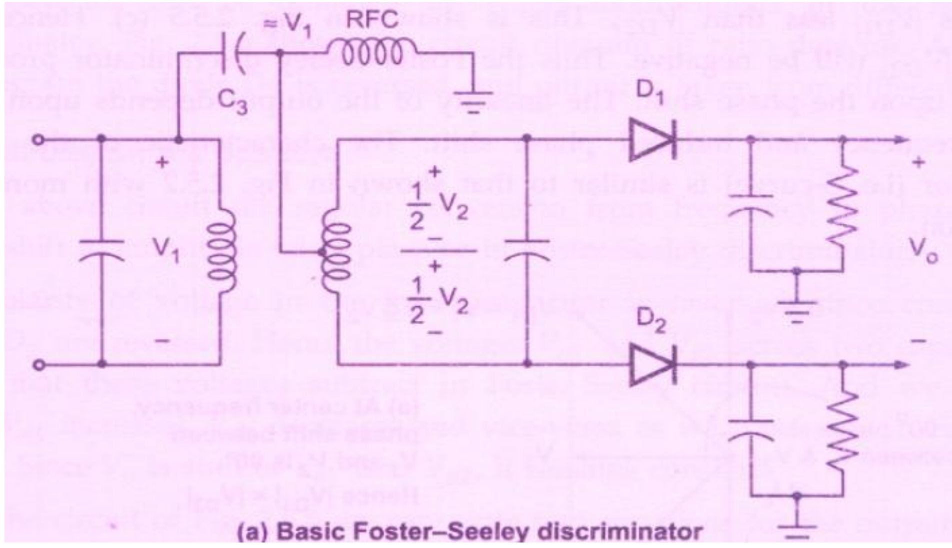
2. There are two slope circuits in Fig. 7. Their transfer functions are as follows:

$$\boldsymbol{H_1(f) = H_2(-f) = \begin{cases} j2\pi a\left(f - f_c + \frac{BW}{2}\right) & for\ f_c - \frac{BW}{2} \le f \le f_c + \frac{BW}{2} \\ 0 & elsewhere \end{cases}}$$

3. When the signal is passed through slope circuits, its amplitude as well as frequency varies as per amplitude of modulating signal $e_m(t)$.
4. This signal is then passed through envelope detectors. It recovers amplitude variations.
5. The outputs of two envelope detectors (one responds to frequency variation above f_c and other responds to frequency variations below f_c) are finally subtracted to give final detected output.

2.3.5 Foster-Seeley Discriminator (Phase Discriminator)

1. The phase shift between the primary and secondary voltages of the tuned transformer is a function of frequency. It can be shown that the secondary voltage lags primary voltage by 90° at the carrier center frequency. This carrier frequency (f_c) is the resonance (or tuned) frequency of the transformer.
2. Foster-Seeley discriminator utilizes this principle for FM detection. Fig. 8(a) shows the circuit diagram of basic Foster-Seeley discriminator. In the figure observe that primary voltage is coupled through C_3 and RFC to the center tap on the secondary. The capacitor C_3 passes all the frequencies of FM.
3. Thus the voltage V_1 is generated across RFC. RFC offers high impedance to frequencies of FM. The voltage V_1 thus appears across (RFC) center tap of secondary and ground also. The voltage of secondary is V_2 and equally divided across upper half and lower half of the secondary coil.
4. Fig. 8 (b) shows the generator equivalent circuit of Foster-Seeley discriminator. In this figure observe that the voltage across diode D_1 is $V_{D1} = V_1 + 0.5V_2$ and that across D_2 is $V_{D2} = V_1 - 0.5V_2$.



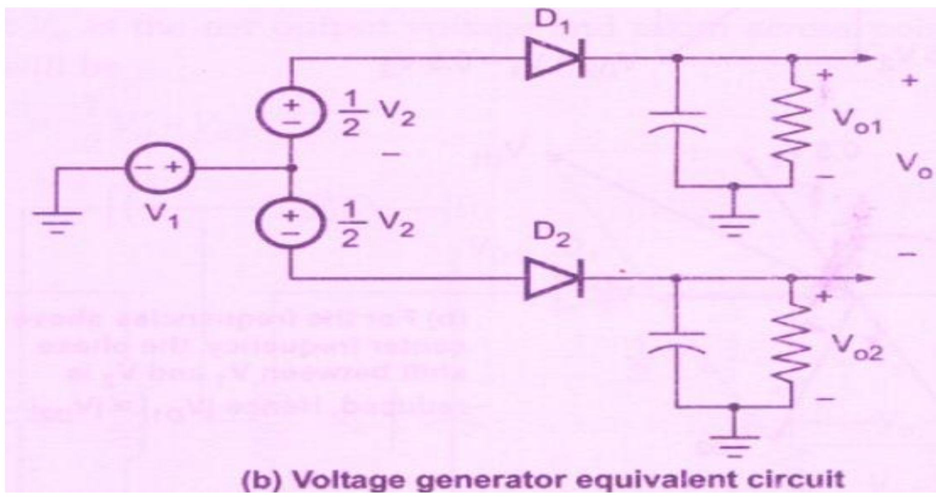
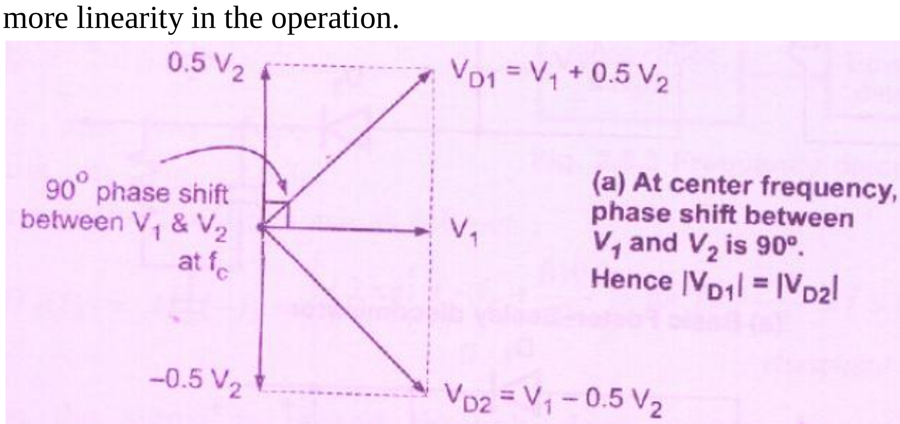
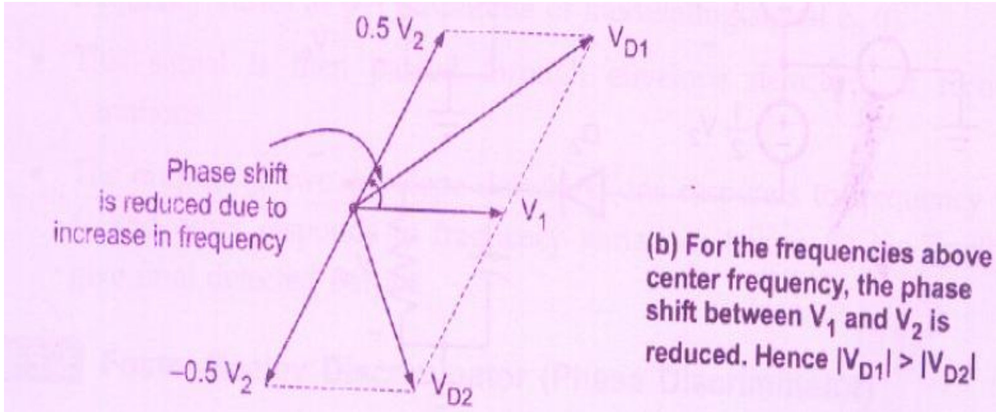


Figure 8.(a) Basic Foster-Seeley discriminator, (b) Voltage generator equivalent circuit.

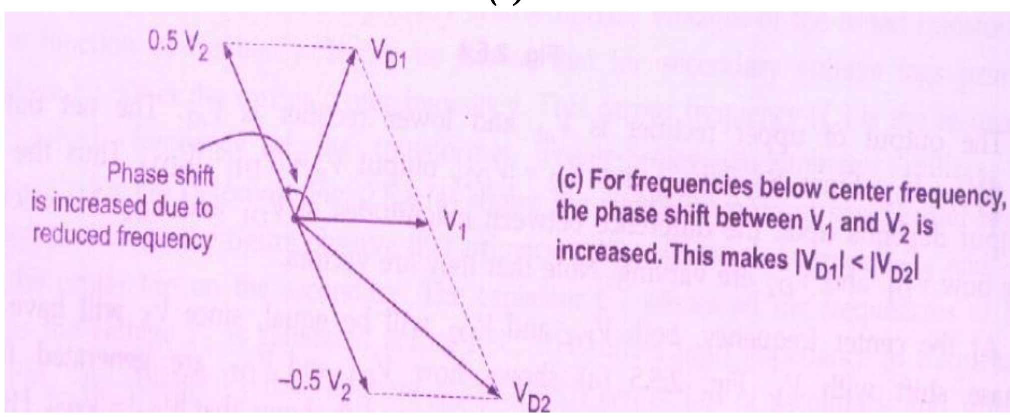
5. The output of upper rectifier is V_{01} and lower rectifier is V_{02}.The net output $V_0 = V_{01} + V_{02}$. Since $V_{01} \approx |V_{D1}|$ and $V_{02} \approx |V_{D2}|$, output $V_0 \approx |V_{D1}| - |V_{D2}|$. Thus the net output depends upon the difference between magnitudes of V_{D1} and V_{D2}.
6. At the center frequency, both V_{D1} and V_{D2} will be equal, since V_2 will have 90° phase shift with V_1. Fig. 9 (a) shows how V_{D1} and V_{D2} are generated from V_1 and V_2. In the figure vector addition is shown and it shows that $|V_{D1}|=|V_{D2}|$.Hence the net output of the discriminator will be zero.
7. Now consider, when input frequency increases above f_c. Hence the phase shift between V_1 and V_2 reduces. Therefore $|V_{D1}|$ is greater than $|V_{D2}|$. This is shown by vector addition in Fig. 9 (b). Hence the net output $V_0 = |V_{D1}| - |V_{D2}|$ will be positive. Thus the increase in frequency increases output voltage.
8. Now consider, when frequency reduces below f_c. This makes $|V_{D1}|$ less than $|V_{D2}|$. This is shown in Fig. 9 (c). Hence the output $V_0 = |V_{D1}| - |V_{D2}|$ will be negative. Thus the Foster-Seeley discriminator produces output depending upon the phase shift.
9. The linearity of the output depends upon the linearity between frequency and induced phase shift. The characteristic of the Foster-Seeley discriminator (i.e. S-curve) is similar to that shown in Fig 6 with more linearity in the operation.



(a)



(b)



(c)

Figure 9. Phasor Representation

2.3.6 Ratio Detector

Ratio detector can be obtained by slight modifications in the Foster-Seeley discriminator. Fig. 10 shows the circuit diagram of ratio detector. As shown in the diagram the diode D_2 is reversed, and output is taken from different points.

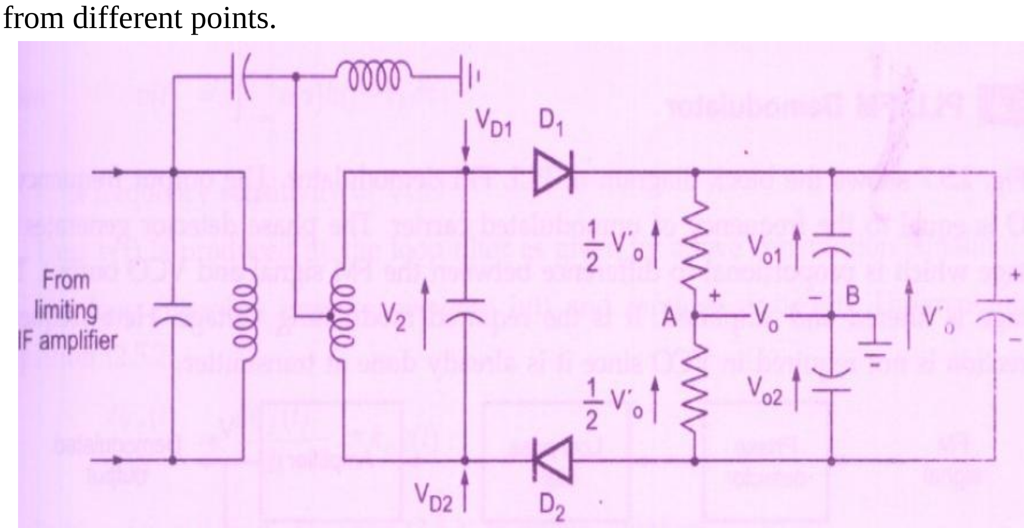


Figure 10. Ratio detector circuit

Circuit diagram and Operation:

1. In the above circuit the regular conversion from frequency to phase shift and phase shift to amplitude takes place as in Foster-Seeley discriminator.
2. The polarity of voltage in the lower capacitor is reversed, since connections of diode D_2 are reversed. Hence the voltages V_{01} and V_{02} across two capacitors add. (Note that these voltages subtract in Foster-Seeley circuit). And we know that when V_{01} increases, V_{02} decreases and vice-versa as we have seen in Foster-Seeley circuit. Since $V_0^{'}$ is sum of V_{01} and V_{02}, it remains constant.
3. From the circuit of Fig.10 we can write two equations for the output voltage V_0 (Note that V_0 is the net output voltage and taken across points A and B). The first equation will be,

$$V_0 = \frac{1}{2}V_0' - V_{02} \qquad (1)$$

And

$$V_0 = -\frac{1}{2}V_0' + V_{01} \qquad (2)$$

Adding the above two equations,

$$2V_0 = V_{01} - V_{02}$$

$$V_0 = \frac{1}{2}\,(V_{01} - V_{02}) \qquad (3)$$

Since, $V_{01} = |V_{D1}|$ and $V_{02} = |V_{D2}|$, above equation will be,

$$\boldsymbol{V_0 = \frac{1}{2}\,(|V_{D1}| - |V_{D2}|)} \qquad (4)$$

4. Here V_{D1} and V_{D2} are obtained as discussed earlier in Foster-Seeley circuit. The above equation shows that the output of ratio detector is half compared to that of Foster-Seeley circuit.
5. We have seen earlier that, as frequency increases above f_c , $|V_{D1}| > |V_{D2}|$, hence output V_0 is positive. Similarly if frequency decreases below f_c, $|V_{D1}| < |V_{D2}|$, hence output V_0 is negative.

Advantages and Disadvantages

Advantages

i) As compared to Foster - Seeley circuit, this circuit does not respond to amplitude variations.

ii) The output is bipolar (i.e. positive as well as negative).

Disadvantages

i) Ratio detector does not tolerate variation in signal strength over performed period.

ii) It requires an ACC signal.

Transmission Bandwidth of FM

1. The number of significant sidebands 'n' produced in an FM waves can be obtained from the plot of bessel function $J_n(m_f)$. For n $> m_f$, the values of $J_n(m_f)$ are negligible particularly when $m_f \gg 1$. Therefore, the significant sidebands produced in wideband FM may be considered to be an integer approximately equal to m_f ie. n$\simeq m_f$ if $m_f \gg 1$.
2. The USB are separated by 'ω_m' and form a frequency span of 'nω_m'. Similar span is produced by the LSB.
3. Therefore transmission bandwidth of FM wave is defined as the separation between the frequencies beyond which none of side frequencies is greater than 1% of the carrier amplitude obtained when the modulation is removed.

i.e., B.W. = $2n\omega_m \, {}^{rad}/_{sec}$; Where n= number of sidebands.

If n$\simeq m_f$ then B.W. = $2m_f\omega_m$ or $m_f = \frac{\Delta\omega}{\omega_m}$

Hence B.W. = $\frac{2\Delta\omega}{\omega_m}\omega_m$ = $2\Delta\omega$ rad. = $2(\Delta f)$ Hz

4. Thus the approximate bandwidth of a wide band FM system is given as twice the frequency deviation. This approximation holds true for $m_f \gg 1$. For smaller values of m_f, the bandwidth may be more than $2\Delta\omega$.
5. The approximate rule for transmission bandwidth of an FM signal generated by a single tone modulating signal is,

B.W. = $2(\Delta\omega + \omega_m)$ we know that $\omega_m = \frac{\Delta\omega}{m_f}$

$$\textbf{B.W.} = \boldsymbol{2\Delta\omega\left(1 + \frac{1}{m_f}\right)} \textbf{ radian.}$$

The empirical relation is known as "Carson's rule".

Bandwidth of PM

1. The PM bandwidth as per Carson's rule $\boldsymbol{(BW)_{pm} = 2\Delta\omega = 2K_pV_m\omega_m}$.
2. Thus the B.W of the PM signal varies tremendously with a change in modulating frequency ω_m.

UNIT II - DIGITAL COMMUNICATION

Pulse modulations – concepts of sampling and sampling theorems, PAM, PWM, PPM, PTM, quantization and coding : DCM, DM, slope overload error. ADM, DPCM, OOK systems – ASK, FSK, PSK, BSK, QPSK, QAM, MSK, GMSK, applications of Data communication.

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INTRODUCTION:

In digital communication the information is processed and transferred by a sequence of digital messages.

BLOCK DIAGRAM OF DIGITAL COMMUNICATION SYSTEM:

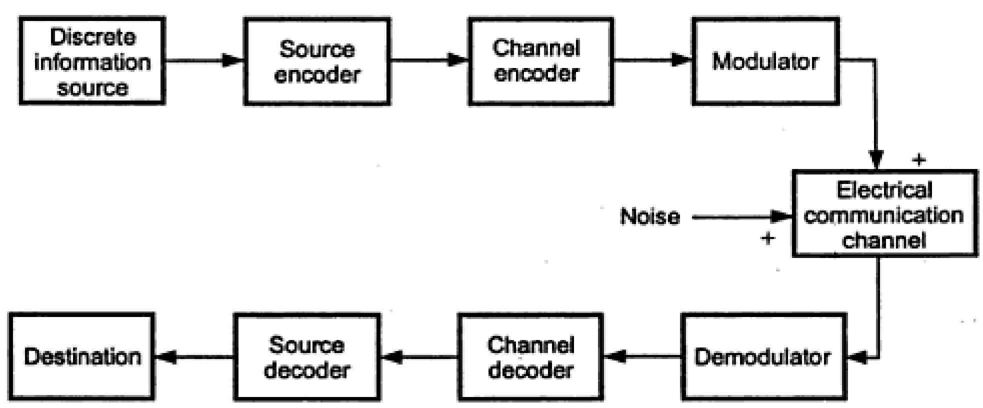


Fig. 1.2.1 Basic digital communication system

- The information may be either analog or digital. If the information is analog then it is converted into digital by sampling and quantizing.
- The source encoder converts the information symbol into a binary sequence 0's and 1's by assigning code words to the input symbols.
- The channel encoder generates redundant symbols.
- Modulator accepts the inputbit stream and convert it into electrical waveform suitable for tx over long distances.
- The communication channel provides the electrical connection between the source and destination.
- Demodulator is the reverse process by which the original message is extracted from the modulated signal .
- The redundant information is decoded by the decoder.
- The source decoder decodes the binary information into original msg signal .
- Thus the original msg is received by the destination.

ADVANTAGES OF DIGITAL COMMUNICATION:

- Noise interference is minimum.
- Multiplexing is easier.
- Secure data transmission.
- Using channel coding errors can be detected and corrected easily.

Disadvantages:

- **More transmission bandwidth** is required.
- **It needs synchronisation** in case of synchronous modulation.

Sampling:

Sampling is the process of converting continuous time signal into a discrete time signal by measuring the signal at periodic intervals of time.

- Analog signal is converted into digital before it is transmitted by digital transmission method.
- So we go for sampling.

Sampling theorem:

- A band limited signal of finite energy ,which has no frequency components higher than 'W' hertz is completely described by specifying the values of the signalat instants of time separated by 1/2w seconds apart.
- A band limited signalof finite energy which has no freq. components higher than w hertz may be completely recovered from its samples taken at the rate of 2w samples per second.
- A continuous time signal can be completely converted into its samples and recovered back if the sampling frequency is twice of the highest frequency content of the signal. Fs≥2w.

Sampling theorem states that the sampling frequency (f_s) should be greater than or equal to twice of the maximum frequency of the modulating signal m(t)

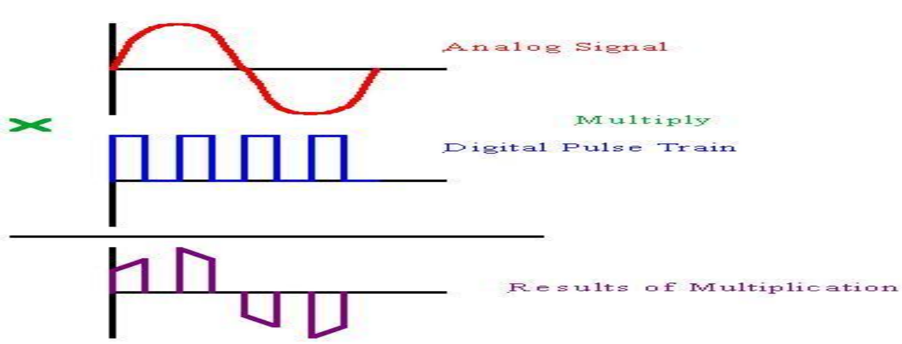


Fig.2 : Sampling

i.e., is $f_s \geq 2f_m$

let the value of m(t)be determined at regular intervals separated by times T_s .

$T_s \leq 1/ 2f_m$

Where, T_s - called sampling time.

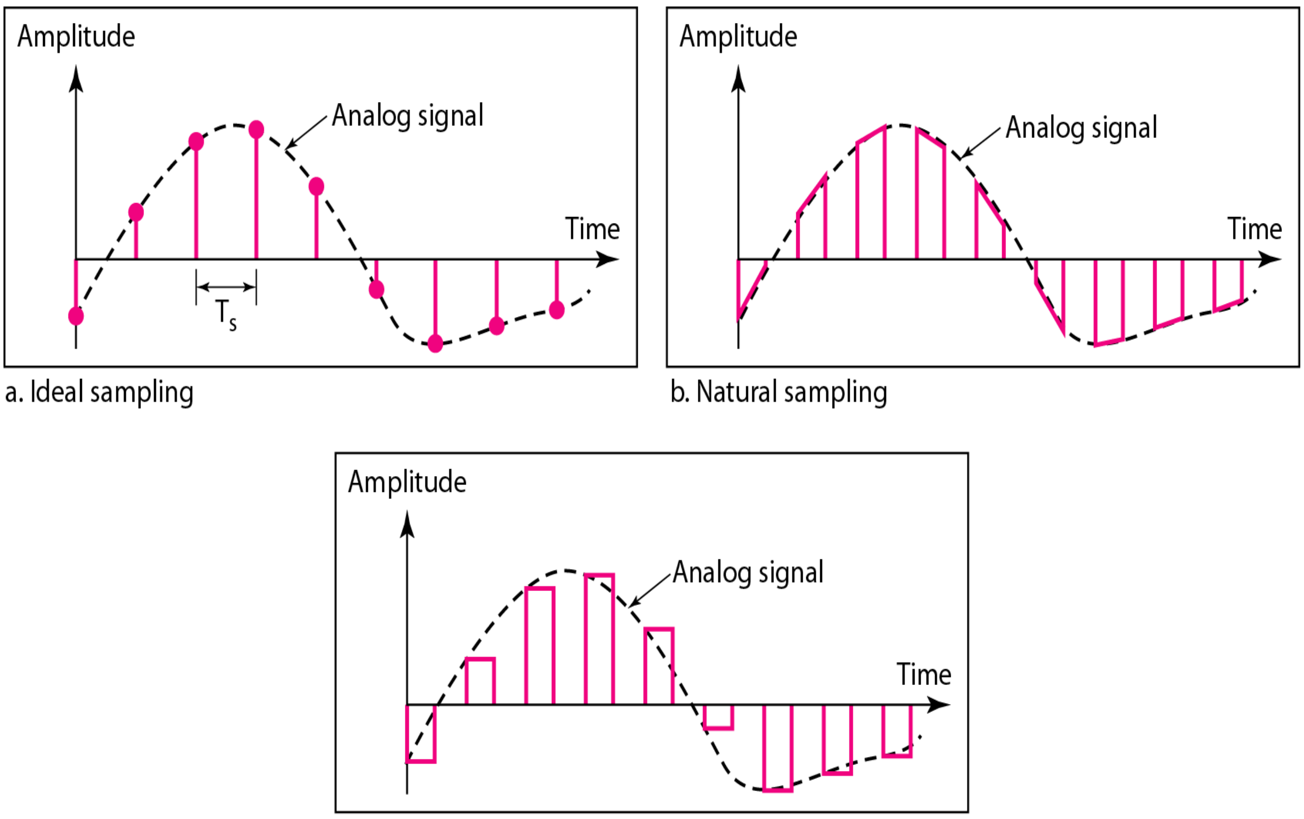
Case i : $f_s = 2f_m$ => critical modulation

Case ii : $f_s > 2f_m$ => under modulation

Case ii : $f_s < 2f_m$ => over modulation

<u>TYPES OF SAMPLING:</u>

- Ideal or instantaneous Sampling.
- Natural sampling.
- Flat top sampling.



a. Ideal sampling

b. Natural sampling

c. Flat-top sampling

Fig.4:Types of Sampling

<u>Proof of sampling theorem:</u>

<u>representation of x(t) in terms of its samples:</u>

the sampled signal **$x_\delta(t)$ is given as, $x_\delta(t) = \sum_{n=-\infty}^{\infty} x(t)\delta(t-nTs)$------------------------(1)**

$x_\delta(t)$ = product of X_δ and impulse train $\delta(t)$,$\delta(t-nTs)$→ samples placed at +Ts,+2Ts,+3Ts -Ts,-2Ts,-3Ts and so on.

Taking fourier transform of eqn(1)

$$X_\delta(f) = FT\left\{\sum_{n=-\infty}^{\infty} x(t)\,\delta(t-nT_s)\right\}$$

$$= FT\ \{\text{Product of } x(t) \text{ and impulse train}\}$$

We know that FT of product in time domain becomes convolution in frequency domain. i.e.,

$$X_\delta(f) = \text{FT}\{x(t)\} * FT\{\delta(t-nT_s)\} \quad \ldots(1.3.2)$$

By definitions, $x(t) \xleftrightarrow{FT} X(f)$ and

$$\delta(t-nT_s) \xleftrightarrow{FT} f_s \sum_{n=-\infty}^{\infty} \delta(f-nf_s)$$

Hence equation (1.3.2) becomes,

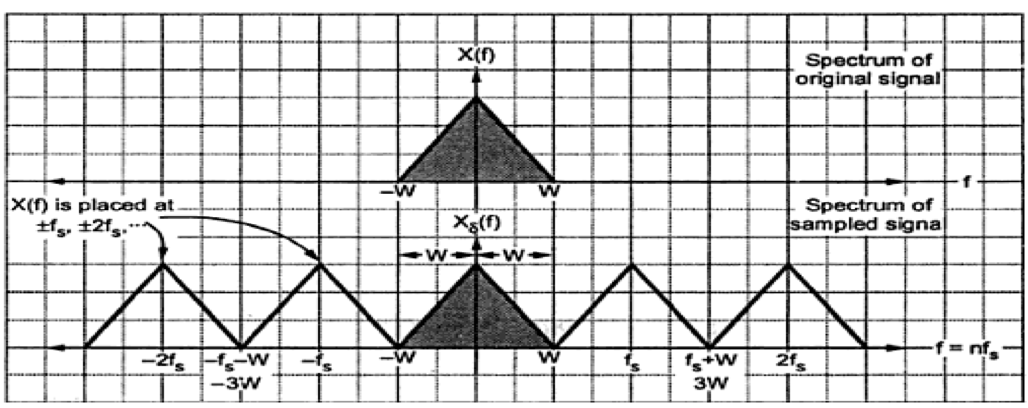
$$X_\delta(f) = X(f) * f_s \sum_{n=-\infty}^{\infty} \delta(f-nf_s)$$

Since convolution is linear,

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

$$= f_s \sum_{n=-\infty}^{\infty} X(f-nf_s) \qquad \textit{By shifting property of impulse function}$$

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



Step 3 : Relation between $X(f)$ and $X_\delta(f)$

Important assumption : Let us assume that $f_s = 2W$, then as per above diagram.

$$X_\delta(f) = f_s X(f) \qquad \text{for } -W \le f \le W \text{ and } f_s = 2W$$

or
$$X(f) = \frac{1}{f_s} X_\delta(f) \quad \ldots (1.3.3)$$

Step 4 : Relation between $x(t)$ and $x(nT_s)$

DTFT is,
$$X(\Omega) = \sum_{n=-\infty}^{\infty} x(n) e^{-j\Omega n}$$

$$\therefore \quad X(f) = \sum_{n=-\infty}^{\infty} x(n) e^{-j2\pi fn} \quad \ldots (1.3.4)$$

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

Step 1 : Take inverse Fourier transform of $X(f)$ which is in terms of $X_\delta(f)$.
Step 2 : Show that $x(t)$ is obtained back with the help of interpolation function.

Step 1 : The IFT of equation (1.3.5) becomes,

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

Here the integration can be taken from $-W \leq f \leq W$. Since $X(f) = \frac{1}{f_s} X_\delta(f)$ for $-W \leq f \leq W$. (See Fig. 1.3.2).

In above equation 'f' is the frequency of DT signal. If we replace $X(f)$ by $X_\delta(f)$, then 'f' becomes frequency of CT signal. i.e.,

$$X_\delta(f) = \sum_{n=-\infty}^{\infty} x(n) e^{-j2\pi \frac{f}{f_s} n}$$

In above equation 'f' is frequency of CT signal. And $\frac{f}{f_s}$ = Frequency of DT signal in equation (1.3.4). Since $x(n) = x(nT_s)$, i.e. samples of $x(t)$, then we have,

$$X_\delta(f) = \sum_{n=-\infty}^{\infty} x(nT_s) e^{-j2\pi f n T_s} \quad \text{since } \frac{1}{f_s} = T_s$$

Putting above expression in equation (1.3.3),

$$X(f) = \frac{1}{f_s} \sum_{n=-\infty}^{\infty} x(nT_s) e^{-j2\pi f n T_s}$$

Inverse Fourier Transform (IFT) of above equation gives $x(t)$ i.e.,

$$x(t) = IFT\left\{ \frac{1}{f_s} \sum_{n=-\infty}^{\infty} x(nT_s) e^{-j2\pi f n T_s} \right\} \quad \dots (1.3.5)$$

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

Interchanging the order of summation and integration,

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

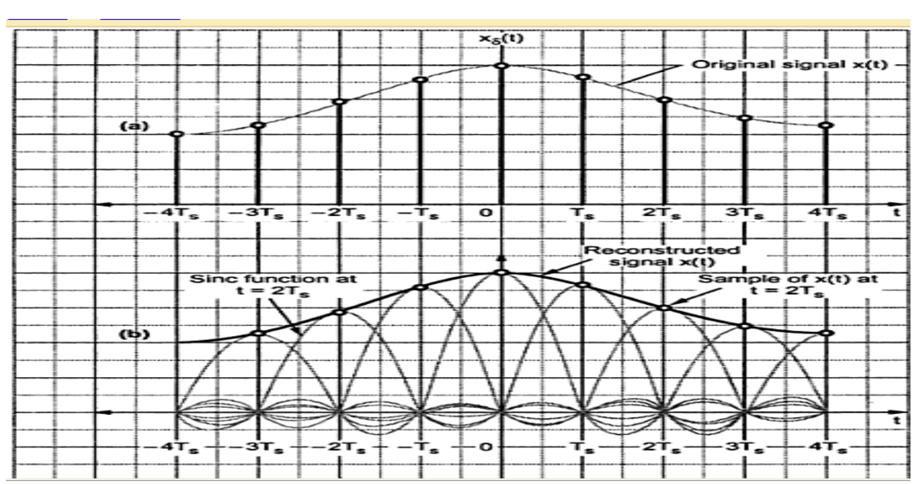
$$= \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \frac{1}{f_s} \cdot \left[\frac{e^{j2\pi f (t-nT_s)}}{j2\pi (t-nT_s)} \right]_{-W}^{W}$$

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

$$= \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \frac{1}{f_s} \cdot \frac{\sin 2\pi W (t-nT_s)}{\pi (t-nT_s)}$$

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

Here $f_s = 2W$, hence $T_s = \frac{1}{f_s} = \frac{1}{2W}$. Simplifying above equation,



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

Here $f_s = 2W$, hence $T_s = \frac{1}{f_s} = \frac{1}{2W}$. Simplifying above equation,

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

$$= \sum_{n=-\infty}^{\infty} x(nT_s) \operatorname{sinc}(2Wt - n) \qquad \text{since } \frac{\sin \pi\theta}{\pi\theta} = \operatorname{sinc}\theta \qquad \ldots(1.3.6)$$

Step 2 : Let us interpret the above equation. Expanding we get,

$$x(t) = \cdots + x(-2T_s)\operatorname{sinc}(2Wt+2) + x(-T_s)\operatorname{sinc}(2Wt+1) + x(0)\operatorname{sinc}(2Wt) + x(T_s)\operatorname{sinc}(2Wt-1) + \cdots$$

When the interpolated signal of equation 1.36 is passed through the low pass filter of bandwidth -W≤f≤W then the reconstructed waveform is obtained.

ALIASING:

When the high frequency components interferes with the low freq components and appears as low freq , the data is lost and it cannot be recovered, then the phenomenon is called aliasing.

DIFFERENT WAYS TO AVOID ALIASING:

- Sampling rate **fs≥2w.**
- **Strictly bandlimit the signal to"W".**

NYQUIST RATE:

The sampling rate f_s =2W samples per second for a signal bandwidth of W Hertz is called Nyquist rate.

NYQUIST INTERVAL:

The reciprocal of Nyquist rate 1/2W is called Nyquist interval.

Nyquist rate =2W hertz

Nyquist interval =1/2W seconds

PULSE MODULATION:

If the characteristics of the carrier wave pulses is varied in accordance with the message signalthen it is called pulse modulation

TYPES OF PULSE MODULATION:

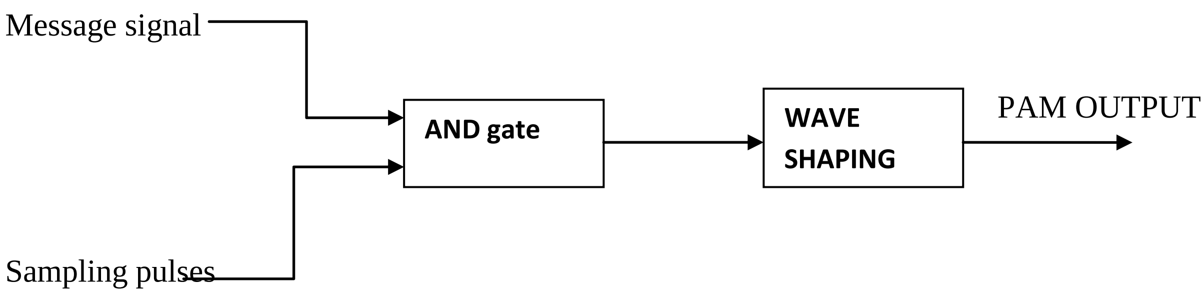
1. **Analog pulse modulation**
   - Pulse amplitude modulation**(PAM)**
   - Pulse width modulation**(PWM)**
   - Pulse position modulation**(PPM)**
2. **Digital pulse modulation**

- Pulse code modulation**(PCM)**
- Delta modulation**(DM)**

PULSE AMPLITUDE MODULATION:

If ***the amplitude of the carrier pulse is changed in accordance with the message signal***then it is called pulse Amplitude modulation.

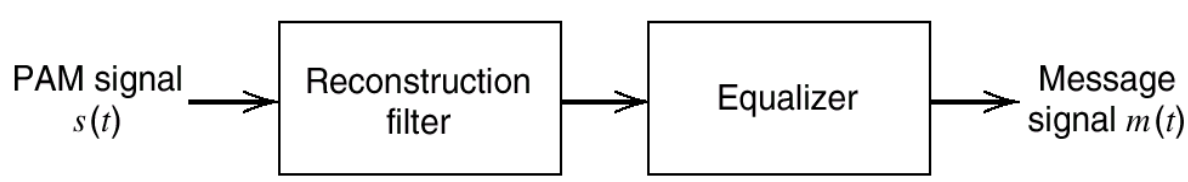
GENERATION OF PAM:



- The analog signalis given as one input to the AND gate along with the sampling pulses.
- The output is the pulses whose amplitude is equal to the amplitude of the msg signalat that instant.
- Then it is given to the pulse shaping network which gives them Flat tops.
- PAM is also generated by CE amplifier.

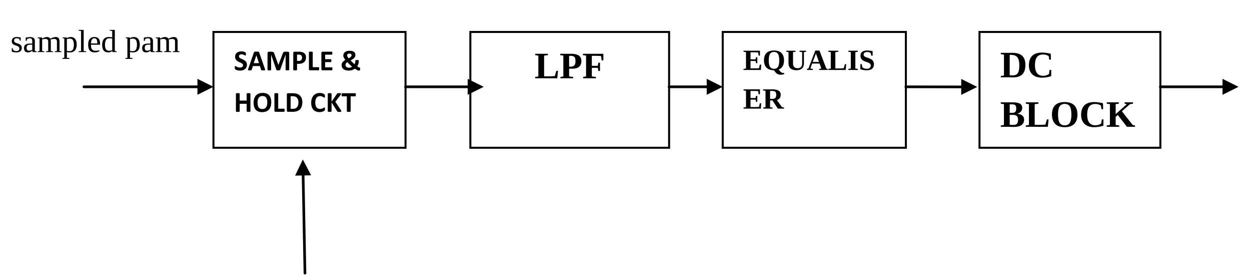
- When either one of the signal is present base voltage is not sufficient to turn ON the transistor T.
- If both the signals are present T1 is switched ON .
- So the output is equal to the message signalamplitude.

PAM DETECTION BY LOW PASS FILTERING:

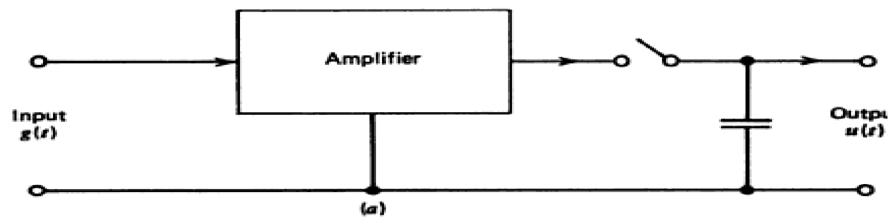


- The PAM signalis applied to the LPF.
- LPF removes all the spectral components outside the bandwidth of the signal.
- The equalization filter is used to remove the distortion present in the signal .
- Since the filter bandwidth is selected 2W ,the filter ideal characteristics will be deviated for a freq >2W.hence this method is imperfect.
- Hence sample and hold circuit is used for reconstruction.

SAMPLE AND HOLD CIRCUIT :



timing



- Assume the inputapplied to the sample and hold circuit is flat top PAM.
- At the sample points the switch closes and output is equal to the inputsample amplitude.
- While the switch is open the capacitor holds the voltage until the next closure.
- So the output remains flat top sampled signalbut its pulse width is varied to Ts from T.

The output of the sample and hold circuit is defined by

U(t) = $\sum_{n=-\infty}^{\infty} g(nTs)h(t-nTs)$--------------------(1)

h(t)-> impulse response representing the action of sample and hold circuit.

h(t) = {1 0<t<Ts}

h(t) = 0 if t<0 and t>TS

the spectrum of the output of the sample and hold circuit is given by

U(f) =fs $\sum_{n=-\infty}^{\infty} H(f)G(f-mfs)$**--------------(2)**

G(f)→ fourier transform of the original analog signalg(t) ;H(f)→transfer fn of sample and hold circuit

H(f)= Ts sinc(fTs)exp(-jπfTs)

- In order to recover the original signalwithout any distortion the output of S/H ckt is passed through low pass filter to remove the components of the spectrum U(f) at multiples of the samling rate fs and an equalizer whose amplitude response equals 1/|H(f)|.

<u>TRANSMISSION BANDWIDTH OF PAM:</u>

- The bandwidth required for the transmission of PAM signalwill be equal to the maximum frequency f_{max}

$$B_T \geq f_{max}; \qquad B_T \geq 1/2\tau$$

- **The tx b.w of PAM signalis B_T>>w.**

<u>DISADVANTAGE OF PAM:</u>

- PAM requires ***larger bandwidth***.
- Since the Amplitude of the pulses is varied ***noise interference is more***.
- The peak power required by the transmitter with the message signalvaries with the message signal .

PULSE WIDTH MODULATION and PULSE POSITION MODULATION:

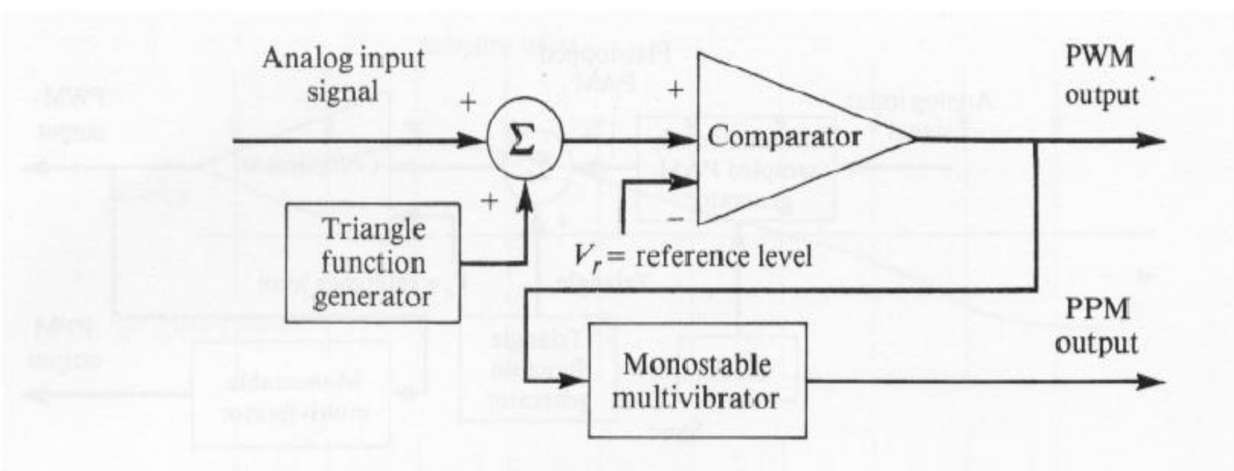
Pulse Width Modulation:

- PWM is the process in which the ***width of the pulse is varied*** in accordance with the message signal

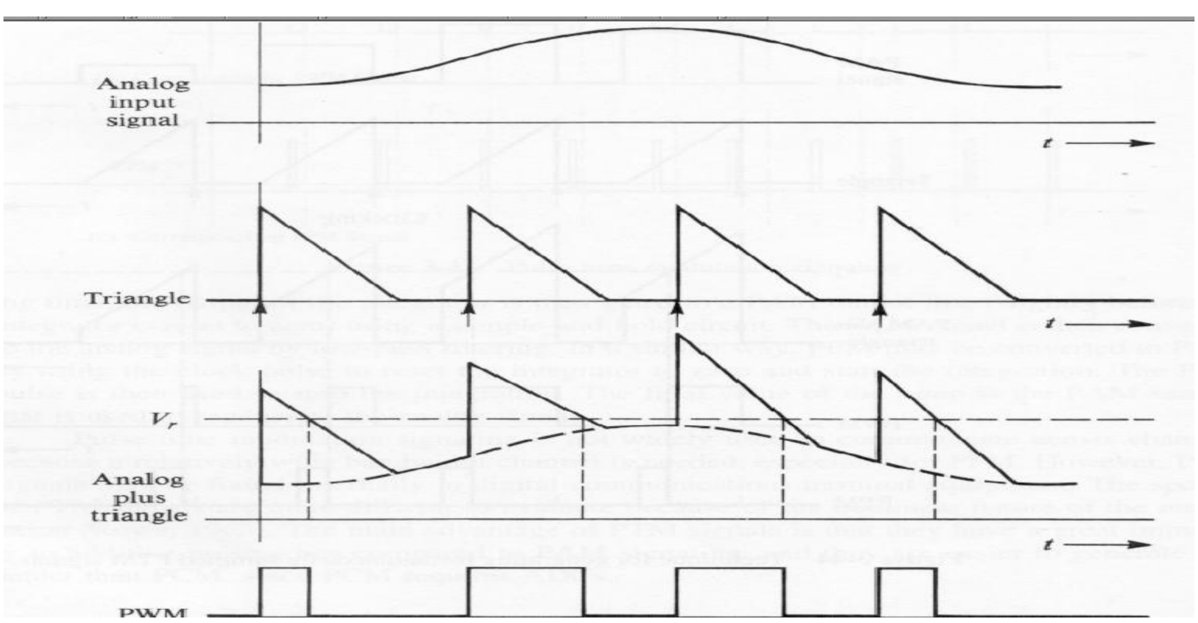
Pulse Position Modulation:

- PPM is the process in which the ***position of the pulse is varied*** in accordance with the message signal .

GENERATION OF PWM AND PPM



1. The Triangle generator generates the triangle waveform of frequency Fs.
2. The triangle signalalso called sampling signalis applied to the comparator.
3. The output of the comparator is high when the analog inputand triangle waveform is greater than reference voltage.
4. When the analog inputand triangle waveform is lesser than reference voltage the output of comparator is zero.
5. PWM signalis nothing but output of the comparator.
6. The Amplitude of PWM signalwill be the positive saturation of the comparator.



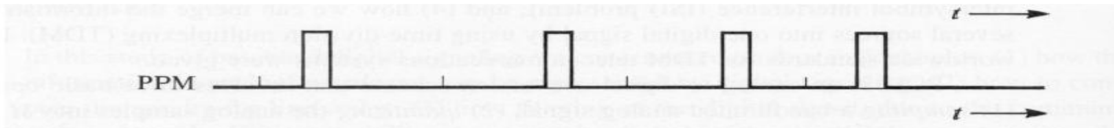


Figure 3-45 Technique for generating naturally sampled PTM signals.

Generation of PPM:

1. To generate PPM PDM signalis used as the trigger inputto one monostable multivibrator.
2. The monostable output remains zero until it is triggered.
3. The monostable is triggerd on the falling edge of PDM.
4. The output of monostable then switches to positive saturation level'A'.
5. This voltage remains high for the fixed period then goes low.
6. The width of the pulse can be determined by monostable.
7. Depending upon the amplitude of inputsignalthe pulse is delayed .

TRANSMISSION BANDWIDTH OF PPM & PWM:

- Since the amplitude is constant in both PPM & PWM non linear amplitude distortion as well as noise interference does not affect the detection at the rx.
- Both PPM & PWM needs a sharp rise time and fall time for pulses in order to preserve the message information.
- Rise time should be very less than Ts.**T_r << T_s .And transmission bandwidth should be $B_T \geq 1/2t_r$.**

ADVANTAGES OF DIGITAL REPRESENTATION OF THE SIGNAL:

- ***Minimum noise intereference.***
- ***Efficient regeneration of coded signal***along the transmission path.
- **Highly privacy and security** through the use of encryption
- **Possibility of uniform format** for different kinds of baseband signals.

PULSE CODE MODULATION:

- Pulse code modulation (PCM) is produced by analog-to-digital conversion process.
- ***PCM is a signal coding in which the msg signalis sampled ,the amplitude of each sample is rounded off to the nearest value so that both time and amplitude are represented in discrete form.***

Basic operations involved in PCM are

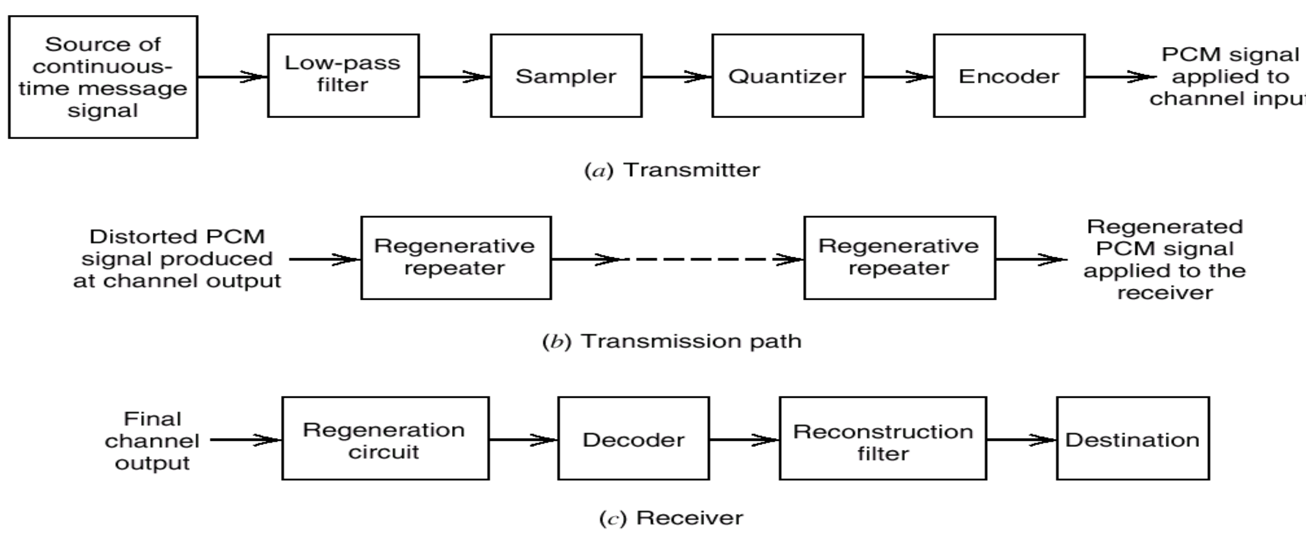
- Sampling
- Quantizing
- Encoding

1)Sampling:

- The incoming message signalis sampled with rectangular pulses by the sampling process.

- For the perfect reconstruction of the message signalat the receiver the sampling rate must be greater than twice the highest frequency component ie[T>1/2Fm].
- A LPF is used at the input in order to limit the frequencies above W .
- Thus sampling permits the reduction of the continuously varying message wave to a limited number of discrete values per second

PCM BLOCK DIAGRAM:



2)Quantizing:

- **quantization is the process of selecting a set of discrete voltage levels and rounding off exact samples to the nearest discrete or quantum level.**
- An analog signalsuch as voice signalhas a continuous range of amplitude and therefore its samples cover a continuous amplitude range.
- it is not necessary to transmit the exact amplitudes of the samples.
- Thus the original continuous signalmay be approximated by a signal constructed of discrete amplitudes selected on minimum error basis from an available input.
- ***Quantization is defined as the process by which analog samples of the original message signal s are transformed into discrete amplitude taken from the finite set of possible amplitudes.***
- Simply quantization is the process of converting the discrete time continuous amplitude signalinto discrete time dicrete amplitude signal.
- The difference between the two discrete levels is called as a quantum or step size.
- Let us consider a continuous signalwhose amplitude varies from -0.5 v to 7.5 v.This range is divided into eight levels

Quantization level	Binary code

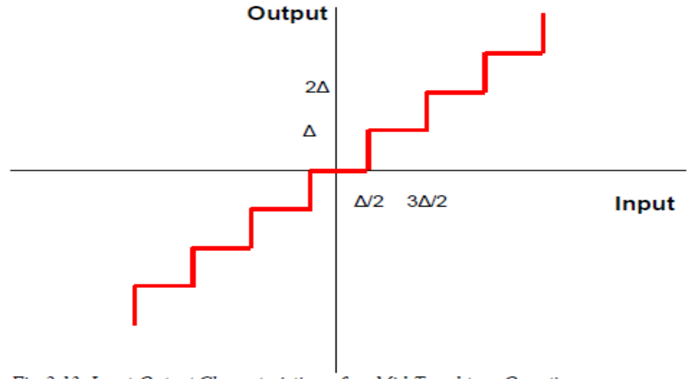
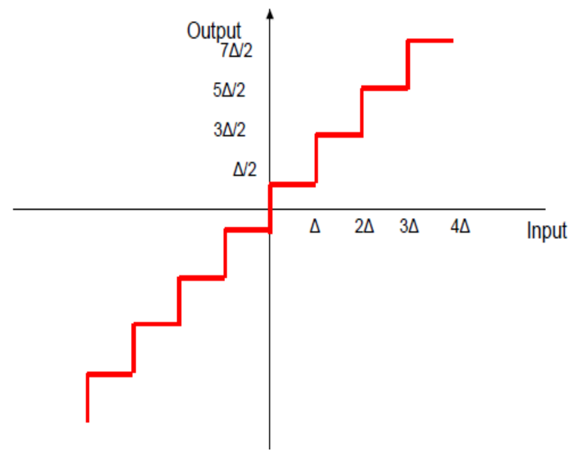


Fig:2.13 Input-Output Characteristics of a Mid-Tread type Quantizer



0	000
1	001
2	010
3	011
4	100
5	101
6	110
7	111

The quantization has two fold effects

- The peak to peak range of inputvalues are subdivided into set of decision levels that are considered as risers of stair case.
- The output is assigned a Discrete value selected from a finite set of representation levels are considered as treads of stair case.
- **If the separation between decision levels and the representation levels of the quantizer are uniform then it is known as uniform quantization.**
- **Types of quantizer:**
  - ✓ **Mid tread quantizer.**
  - ✓ **Mid riser quantizer.**

Mid tread quantizer:

The uniform quantizer in which the decision levels of the quantizer are located at ±Δ/2, ±3Δ/2, ±5Δ/2 and so on and the representation levels are located at 0, ±Δ , ±2Δ, ±3Δ and so on where Δ is the step size,then this type of quantizer is called midtread quantizer.

Mid riser quantizer.

The uniform quantizer in which the decision levels of the quantizer are located at 0, ±Δ , ±2Δ, ±3Δ and so on and the representation levels are located at±Δ/2, ±3Δ/2, ±5Δ/2 and so on where Δ is the step size,then this type of quantizer is called midriser quantizer.

- For example consider the message signal of range 0 to 5 volts.
- If the samples are taken at equal intervals the sample values are 2.85, 4.85 ,3.73 ,3.20 v
- The sample values are approximated to quantum levels such as 3, 5 ,4 and 3 volts.
- i.e. pulses having values between -0.5 to + 0.5 values are approximated to quantum level of '0' volts and the pulses having amplitudes between 0.5 to 1.5 volts are approximated to 1 volts and so on.
- In uniform quantization the the quantizer has linear characteristics thus step size of the quantizer is uniform ,therefore the quantization error is also constant
- Let X(nTs) =continuous amplitude in the range of –Xmax to Xmax.
- If the amplitude range is divided into 'L' levels then the step size

  $\mathbf{\Delta = X_{max} - (-X_{max})/L = 2\, X_{max}/L}$

If X_{max} =1& - X_{max} =-1 then **δ =2 /L** where **L = 2^b; b**→number of bits used to represent the level.

- The step size 'δ' should be as small as possible then it is assumed that the quantization error will be uniformly distributed random variable.
- **Then quantization error = ε_{max} =|δ/2|** **ie – δ/2≤ ε_{max} ≤δ/2**
- For example in PCM let **b= 4bits then L = 2^3 = 8 levels**
- Thus **the step size δ = 2/L =2/8= ¼** hence the error will be **-1/4 ≤ ε_{max} ≤1/4**
- Ie the quantization error is **¼ of full range of input** .
- if the inputrange is 8 volts the error will be(¼)8 = 2 volts.
  - ✓ For low amplitude signals the error rate is maximum
  - ✓ For high amplitude signals the error rate is very small.

Non-uniform quantization:

Step size is made adaptive by distorting the signal before quantization and the distortion is cancelled at the receiver .this process is called non-uniform quantization.

- Quantization error depends on step size.
- When the steps are uniform in size the low amplitude signals have a poorer signal to quantization noise than high amplitude signal.
- In order to maintain the quantization error constant the step size is made smaller for low amplitude signaland larger for high amplitude signal.
- ***non-uniform quantization is performed by companding process .***

3)ENCODING: ***Assigning binary values to each discrete set of samples is called coding.***

- In a binary code each symbol ie ***symbol 0 →represented by presence of pulse .***

  symbol 1→represented by presence of pulse .
- In a binary code each code word consists of n bits .using we can represent 2^n distinct numbers.

- For eg if a signalis quantized into one of 4levels then it is represented by a 2bit code ($2^n = L$). ($2^4 = 16$)

Representation level	Binary numbers
0	0000
1	0001
2	0010
3	0011
4	0100
5	0101
6	0110
7	0111
8	1000
9	1001
10	1010
11	1011
12	1100
13	1101
14	1110
15	1111

There are several formats to represent the binary signal.

- **Non –return to zero unipolar format.:**
  symbol 0 →represented by absence of pulse
  symbol 1→represented by a pulse of constant amp
- **Non-return to zero polar format.:**

  symbol 0 →represented by negative pulses .

  symbol 1→represented by a positive pulses.

4**) REGENERATION:**

- The PCM waveform is distorted due to the mixing up of noise in the channel.
- The distorted PCM is perfectly reconstructed by the regenerative repeaters.
- The basic fns performed by a regenerative repeaters are
- equalization
- Timing.
- Decision making.

- The equalizer shapes the received pulses to compensate the amplitude and phase distortions.
- The timing ckt provides a periodic pulse train for sampling the equalized pulses at the instants of time where the signal to noise ratio is a maximum.
- The decision device is enabled when the amplitude of the equalized pulse plus noise exceeds a predetermined voltage level..
- The repeater makes the decision whether the pulse is present or not.
  - ✓ If the decision is ‘yes ‘a clean new pulse is transmitted to the next repeater.
  - ✓ If the decision is ‘No’ a clean base line is transmitted.
- The regenerated signalis exactly the same as the signal originally transmitted.
- The regenerated signaldeparts from the original signalfor two main reasons:

✓ The presence of channel noise and interference causes the repeater to make wrong decisions , bit errors are introduced in the regenerated signal .

✓ If the spacing between the received pulses deviates from its assigned value ,a jitter is introduced into the regenerated pulse position thereby causing distortion.

Decoding:

- The receiver regenerates the received pulses.
- These pulses are then re -grouped into code words and de coded into a quantized PAM signal.
- The decoding process generates a pulse with an amplitude equal to the linear sum of all the pulses in the codeword.

RECONSTRUCTION:

- the_decoder output is passed through the low pass filter of cut off frequency W, the LPF reconstructs the original signal

SYNCHRONIZATION:

- The PCM Transmitter and the the receiver has to be synchronized perfectly.
- Synchronization is established by transmiting pulse at the end of the frame .
- The receiver search for the pattern of 1's and 0's alternating at half the frame rate and establishes synchronization.

COMPANDING IN PCM:

- It is the process of compressing and expanding the signal.
- The higher amplitude signal s are compressed prior to transmission and then it is expanded at the receiver.
- We do not know how the signalvary in advance.therefore non uniform quantization is difficult to implement.
- Therefore the signalis amplified at low levels and attenuated at high levels.
- The compression and expansion is performed by passing the signalthrough the amplifier having non-linear transfer characteristics.

µ-law companding:

- compressed output

$$\mathbf{Vout = [V_{max} \ln(1+\mu V_{in} / V_{max})]/[\ln(1+\mu)]}$$

V_{max}→ uncompreesed analog i/p

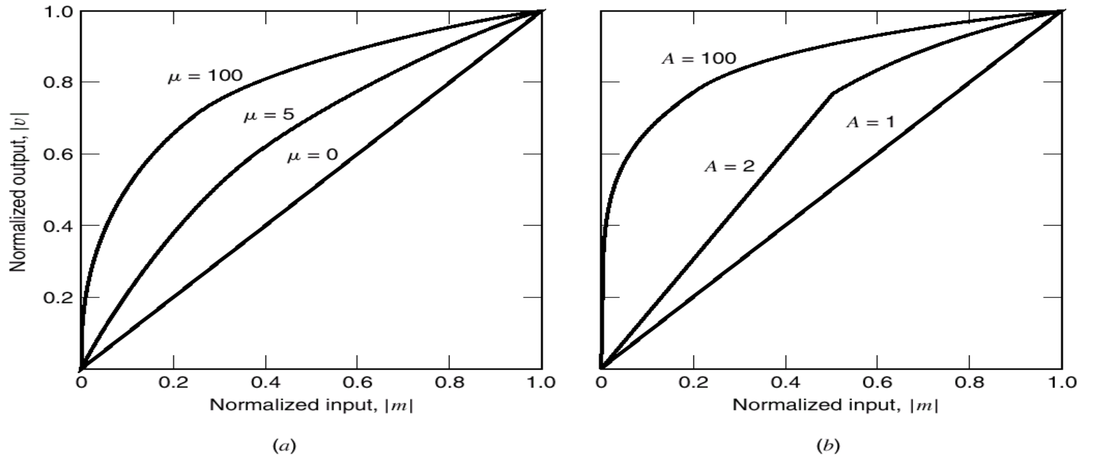
V_{in}→amplitude of the inputsignal.

µ →parameter indicates amount of compression.V_{out} →compressed output .

A-law companding:

$$\mathbf{Vout = [V_{max} \{ AV_{in} / V_{max} \}/[\ln(1+A)]] \qquad 0 \le V_{in} / V_{max} \le 1/A}$$

$$\mathbf{= \{1+\ln(AV_{in} / V_{max})\}/ [\ln(1+A)] \qquad 1/A \le V_{in} / V_{max} \le 1}$$



LIMITATIONS OF PCM:

- ♣ Pcm system are complex compared to analog pulse modulation.
- ♣ The channel bandwidth is increased because of digital coding.

MODIFICATIONS OF PCM:

- PCM can be used for wideband communication since its b.w is more.
- Date rate can be reduced and redundancy can be removed by using data comparison along with pcm.

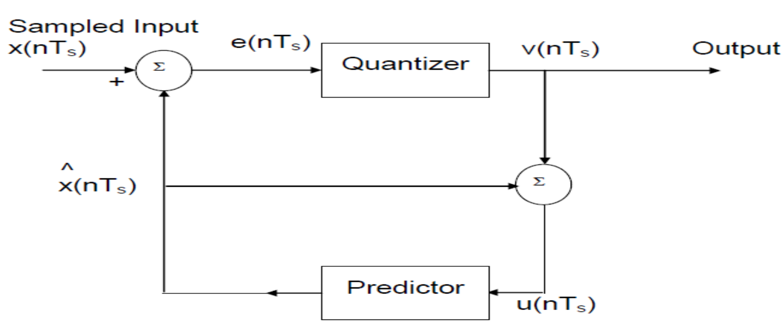
DIFFERENTIAL PULSE CODE MODULATION:

- ***DPCM is a technique in which the difference between two successive samples are transmitted.***
- When a voice signal is sampled at the rate higher than Nyquist rate the resulting sampled signal having a high correlation between the adjacent samples.
- Ie the amplitude difference between the samples is very minimum.
- When these samples are quantized and encoded the resultant PCM contains redundant information.
- To obtain efficient coding the redundant information has to be removed.
- This is done by predicting the future values of the signalif we know some part of the redundant signal .

BLOCK DIAGRAM:

DPCM GENERATION:

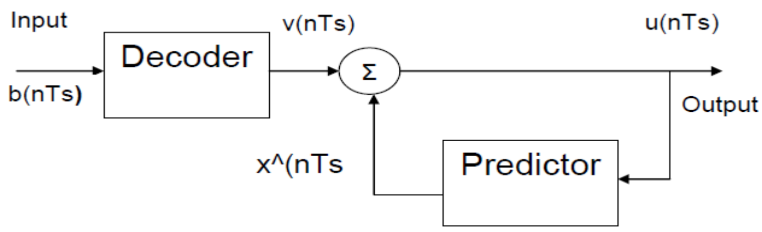
- The original signalm(t) and sampled signal$\hat{m}$ (t) is applied to the difference amplifier.
- The difference amplifier compares m(t) and $\hat{m}$ (t) and the difference between the two signals m(t) -$\hat{m}$ (t) is sampled by the sample and hold circuit.
- Then the sampled signal is quantized and encoded .



Block diagram of DPCM Transmitter

- The predictor is used to predict the past sample values .
- Then this value is given to the difference amplifier through the accumulator.

DPCM RECEIVER:



Block diagram of DPCM Receiver.

- The receiver consists of an accumulator which adds up the received quantized differences.
- The output of the accumulator is $\hat{m}$ (k) which becomes $\hat{m}$ (t) at the filter.
- The filter smooths out the quantization noise.

DPCM TRANSMITTER:

- The msg s.l X(t) is sampled at the rate $F_s = 1/T_s$ to generate the sampled sequence {X(nT_s)}.
- The inputto the quantizer is the error signalwhich is the difference between the unquantized input sample x(nT_s) and its predicted value X^(nT_s)
- **e(nT_S) = x(nT_s)- x^(nT_s)----------------(1)**
- the predicted value **x^(nT_s)**is generated by a predictor whose input is quantized version of the inputsignalx(nT_s).
- the difference signalis called a prediction error .(the amount by which predictor fails to predict the inputexactly.)
- Let the nonlinear function Q(.) denotes the i/p-output characteristics of the quantizer.
- The quantizer output is

  v(nT_s) = Q[e(nT_s)] = e(nT_s) + q(nT_s)---------------(2)

  q(nT_s)→quantization error.

The predictor input**is u(nT_s) = x^(nT_s) + v(nT_s)------------------(3)**

Sub eqn (2) in (3)

u(nT_s) = x^(nT_s) + e(nT_s) + q(nT_s)-------------(4)

from eqn (1)

we can write u(nT_s) = x(nT_s) + q(nT_s)-----------(5)

this eqn represents the quantized version of the inputsignalx(nT_s)

- If the predction is good then the variance of the prediction error e(nT_s) is smaller than the variance x (nT_s).

DPCM RECEIVER:

1. At the receiver the the decoder reconstructs the quantized error signal using the same predictor used at the quantizer.
2. In the absence of noise the encoded signalis identical to the decoded signalat the receiver output .

DELTA MODULATION:

- It uses single bit PCM code to achieve digital transmission of analog signal.
- ***If the current sample is greater than the previous sample logic 1 is txed.***
- ***If the current sample is smaller than the previous sample logic 0 is txed.***
- DM provides the staircase approximation to the oversampled version of an inputbaseband signal .
- The difference between the inputand the approximation is quantized into two levels $+\delta$, $-\delta$ corresponding to positive and negative differences respectively.
    - ***if the approximation falls below the signal ,it is increased by δ.***
    - ***if the approximation falls above the signal ,it is decreased by δ.***
- δ denotes the absolute value of the two representation levels of the one-bit quantizer used in the DM.

BLOCK DIAGRAM:

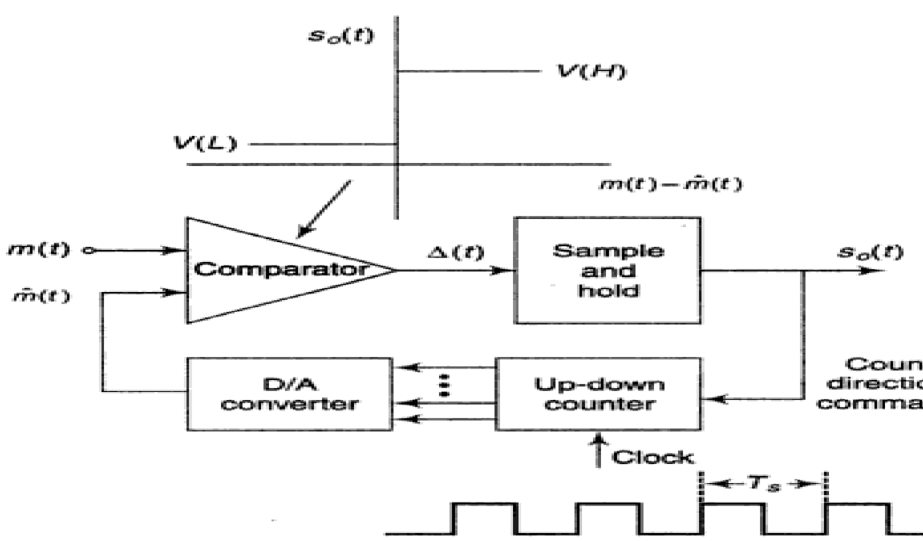


Fig. 5.39 A delta modulator.

1. The baseband signal m(t) and its quantized version $\hat{m}$ (t) are applied as inputs to the comparator.
2. The comparator compares the two input signals and generate the output

    When m(t)> $\hat{m}$ (t) the output is so(t)=V(H)

    When m(t)<$\hat{m}$ (t) the output is so(t)= V(L)

    When m(t)= $\hat{m}$ (t) the output is transitioned between V(H) and V(L).

3. The up-down counter increments or decrements the count at each active edge of the clock pulse.
4. When the the output is so(t)=V(H) the counter counts –up.
5. When the the output is so(t)=V(L) the counter counts down.
6. The counter acts as a accumulator since it add or substract the increment and stores the result.
7. Then the digital output of the counter is converted into analog by D/A converter.

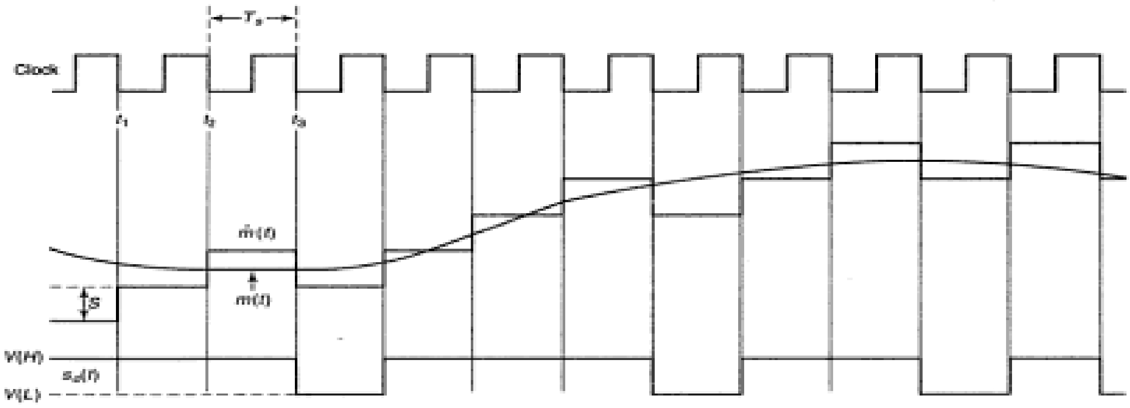
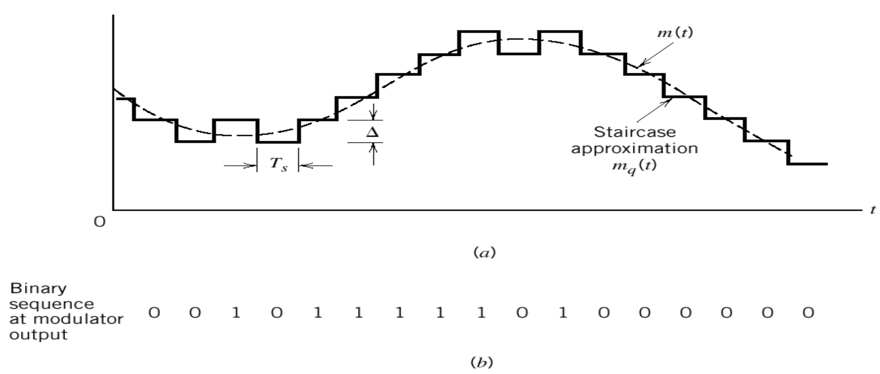


Fig. 5.40 The response of a delta modulator to a baseband signal $m(t)$.

- ***At a time T1 ,m(t)> $\hat{m}$ (t) so that output is so(t)=V(H),when the active clock edge appears counter is incremented and the signal$\hat{m}$ (t) jumps up by an amount equal to step size S.***
- ***At time T3, m(t)<$\hat{m}$ (t) so that output is so(t)=V(L), when the active clock edge appears counter is decremented and the signal$\hat{m}$ (t) jumps down by an amount equal to step size S.***



Delta modulation offers two unique features:

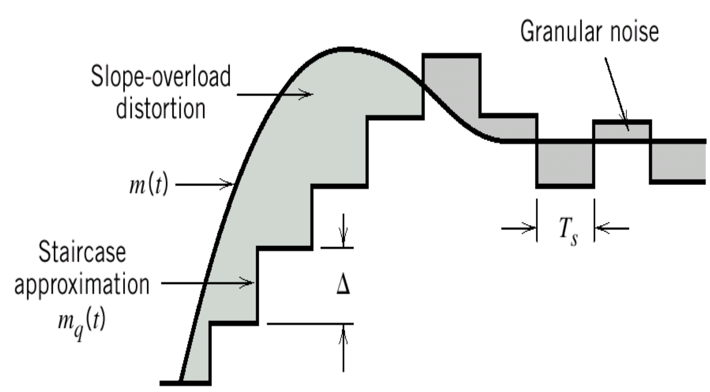
1. No need for Word Framing because of one-bit code word.
2. Simple design for both Transmitter and Receiver

<u>Disadvantage of dm:</u>

Delta modulation systems are subject to two types of quantization error:

(1) slope –overload distortion

(2) granular noise.



(1) <u>slope –overload distortion:</u>

- When the analog signalchanges faster than the stair case approximation we find that the step size $\Delta=2\delta$ is too small .so the staircase approximation does not follow the steep segment of theinputsignal .this condition is called slope-overload and the resulting distortion is called slope-overload distortion
- **To reduce this error the step size should be increased** when slope of signal of x(t) is high.

<u>(2) granular noise:</u>

- Granular noise occurs if the step size is too large .
- For small variations in the input signal, the staircase signalis changed by large amount (δ) because of large step size. This results in an error called granular noise.
- When the input signal is almost flat,the staircase signalu(t) keeps on oscillating by $+\delta,-\delta$.
- **To reduce this error the step size should be decreased** .

- Let $q(nT_s)\rightarrow$quantization error

  then $u(nT_s) = x(nT_s) + q(nT_s)$

prediction error $\mathbf{e(nT_S) = x(nT_s)- x(nT_s-T_s) -q(nT_s-T_s)}$

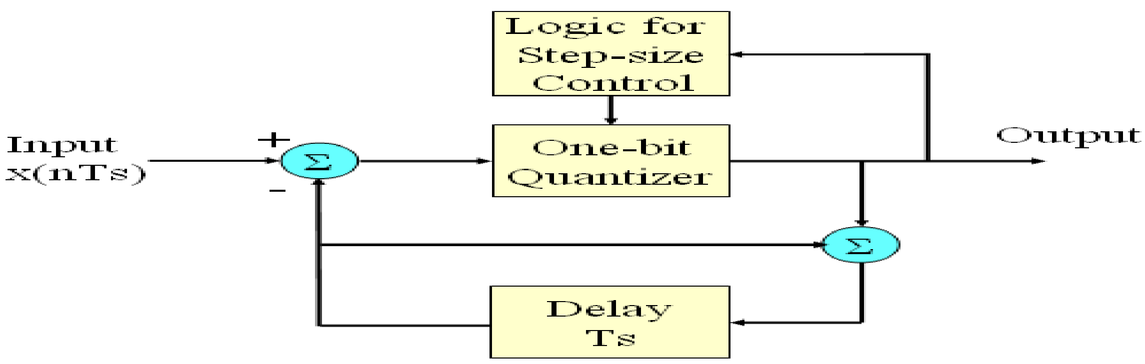
to avoid slope overload distortion the condition

$\boldsymbol{\delta/T_s \geq \max|dx(t)/dt|}$ **should be satisfied.**

ADAPTIVE DELTA MODULATION(ADM):

- To overcome the slope overload distortion and granular noise the step size is made adaptive to the inputsignal
- so that When the inputsignalchanges fastly the step size has to be increased ,and when it changes slowly the step size has to be reduced automatically.
- Then the method is called adaptive delta modulation.
- The adaptive delta modulators take continuous changes in step size or discrete changes in step size.

ADM TRANSMITTER:



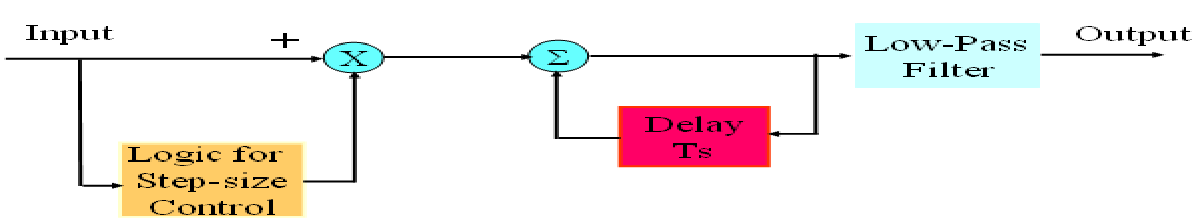
- The step size increases or decrease depending upon the one –bit quantizer.
- For example if the one bit quantizer output is high the step size is doubled .
- if the one bit quantizer output is low the step size is reduced by one step.
- The step size is selected between maximum and minimum values

  $\boldsymbol{\delta_{min}} \leq \boldsymbol{\delta(nT_s)} \leq \delta_{min}$

  δ_{max} controls the slope-overload noise.

  δ_{min} controls the channel noise .

RECEIVER:



1. in the receiver of ADM the first part generates the step size from each incoming bit.

2. The previous inputand present inputdecides the step size.
3. It is then given to the accumulator to construct the staircase waveform.
4. The LPF then smoothens out the reconstructed signal .

advantages of adaptive delta modulation:

1. ***The signal to noise ratio is better than DM*** because of the reduction in slope overload distortion and granular noise.
2. The ***dynamic range is wide*** due to the variable step size.
3. ***Bandwidth utilization is better.***

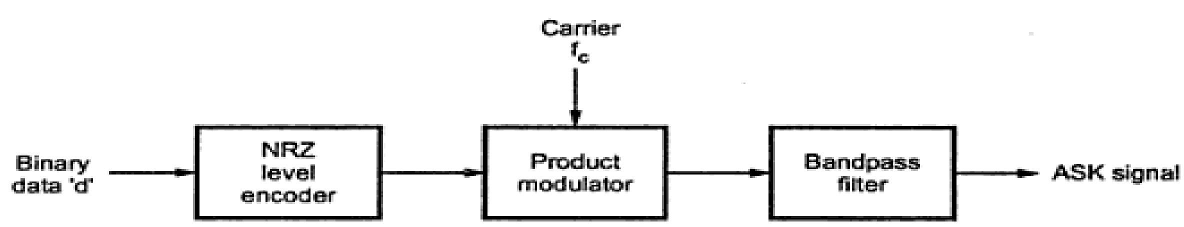
DIGITAL MODULATION:

If digital message signalis modulated by an analog carrier then it is called digital modulation.

AMPLITUDE SHIFT KEYING(ASK) OR OOK:

- ***If the amplitude of the carrier wave is shifted in accordance with the digital message signalthen it is called amplitude shift keying.***
- ***If the digital signalis at logic 1, the output is the carrier.***
- ***If the digital signalis at logic 0, the output is the zero.***

BLOCK DIAGRAM:



1. the NRZ encoder converts the incoming digital data into NRZ code.
2. Then the carrier signalis generated by the local oscillator.
3. The carrier and the NRZ signalare applied to the product modulator.
4. The product modulator multiplies the carrier with the NRZ signal .

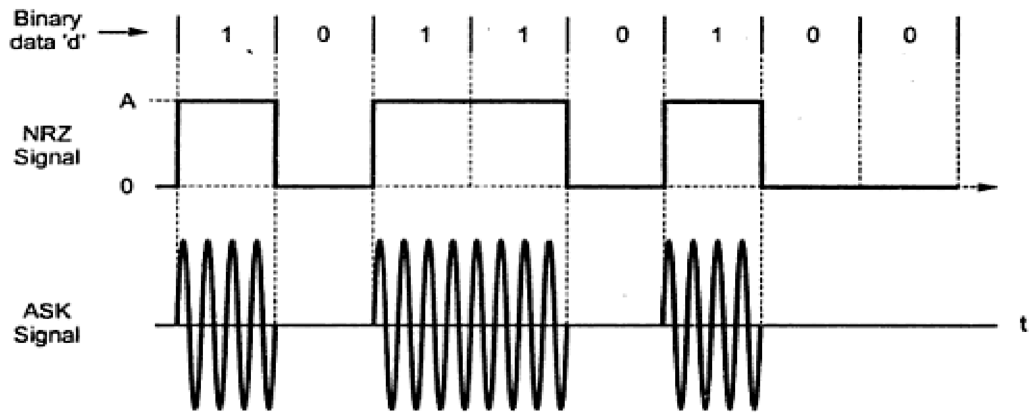
- ***If the digital signalis at logic 1, the output is the carrier.***
- ***If the digital signalis at logic 0, the output is the zero.***
- ***Ie the carrier is 'ON' if the digital is at logic 1.***
- ***The carrier is 'OFF' if the digital is at logic 0.***
- Therefore this method is called ***ON-OFF keying method.***
- The general expression for ASK signalis given as

$$v_{ASK} = d \sin \omega_c t$$

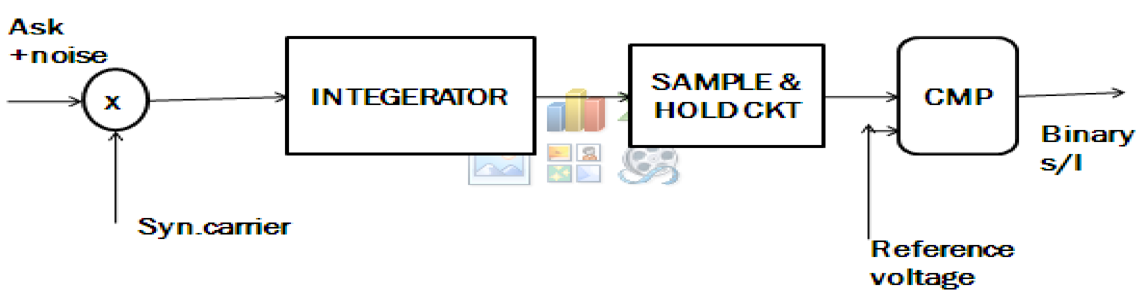
where d→ digital data.

$v_{ASK} = \sin \omega_c t$ **for symbol 1**

$v_{ASK} = 0$ **for symbol** 0



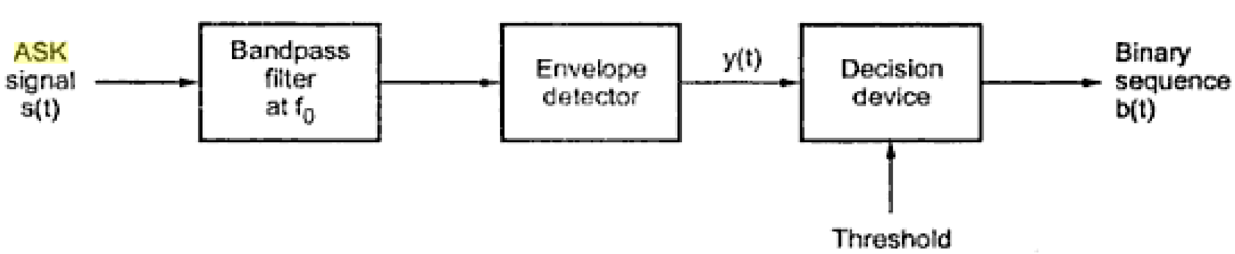
COHERENT RECEPTION OF ASK



1. ***In coherent reception the carrier which has the same frequency and phase as that of the transmitted carrier is used.***
2. First the received ASK signaland the carrier is given to the multiplier.
3. Then the multiplier output is integerated .
4. Then the integerated output is sampled at a particular instant and the output is hold by sample and hold circuit.
5. Then this holded value is compared with the reference voltage by a comparator.
6. ***If S/H o/ p < ref voltage logic 0 is generated.***

   If S/H o/ p > ref voltage logic 1 is generated

NON-COHERENT RECEPTION OF ASK:



1. First the ASK signalis filtered by the BPF to remove the noise .

2. Then the filtered signalis given to the envelope detector which is a rectifier.
3. ***If the ASK signalis <u>present</u> the output of envelope detector is 1.***

   If the ASK signalis <u>absent</u> the output of envelope detector is lesser voltage.
4. The output of the envelope detector is compared in a regenerator.
5. When the envelope detector output is lesser than the ref voltage logic 0 is generated.

<u>Bandwidth of ASK:</u>

Bandwidth = f_b/N = $fb/1 = f_b$ hertz.

$f_b \rightarrow$ no. of bits transmitted per second.

N->no. of symbols changed in a interval of time.

<u>FREQUENCY SHIFT KEYING(FSK) :</u>

Frequency shift keying is a technique in which the frequency of the carrier is shifted between two values f_s or f_m.

- If the digital data is at **logic 1** , the carrier is shifted to **mark frequency(f_m).**
- If the digital data is at **logic 0** ,the carrier is shifted to **space frequency (f_s).**
  - ***If*** the binary digit b(t) = 1 then output **$S_{H(t)}=\sqrt{2Ps}\cos(2\pi f_o+\Omega)t$ --------------**(1)
  - ***If*** the binary digit b(t) = 0 then output **$S_{L(t)}=\sqrt{2Ps}\cos(2\pi f_o-\Omega)t$---------------**(2)

$\Omega \rightarrow$frequency shift.

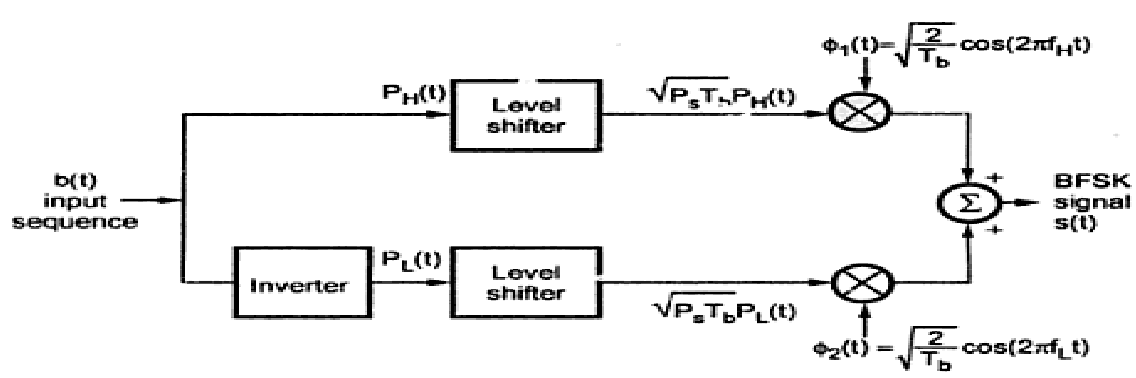
The eqn 1& 2 can be written combinely as **$S(t) = \sqrt{2Ps}\cos(2\pi f_o+d(t)\Omega)t$** -----------------------------(3)

- ***If*** b(t) = 1 then d(t)= +1v the carrier frequency will be **$f_o+\{\frac{\Omega}{2\pi}\}$**
- ***If*** b(t) = 0 then d(t)= -1v the carrier frequency will be **$f_o-1\{\frac{\Omega}{2\pi}\}$**
- ***The*** values P_H(t) and P_L(t) are related to the voltage values of d(t) as

b(t)	**d(t)**	**P_H(t)**	**P_L(t)**
1	**+1v**	**+1v**	**0v**
0	**-1v**	**0v**	**+1v**

When d(t) changes from +1v to -1v , P_H changes from 1 to 0 & P_L **changes from 0 to 1.**

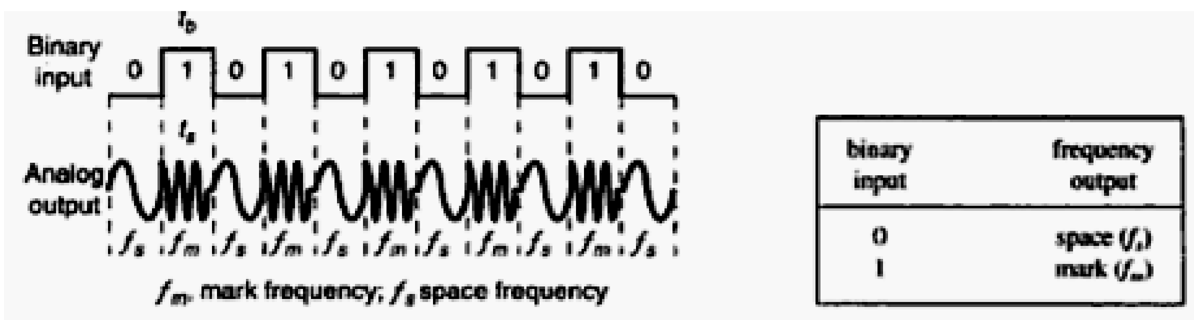
<u>FSK TRANSMITTER:</u>



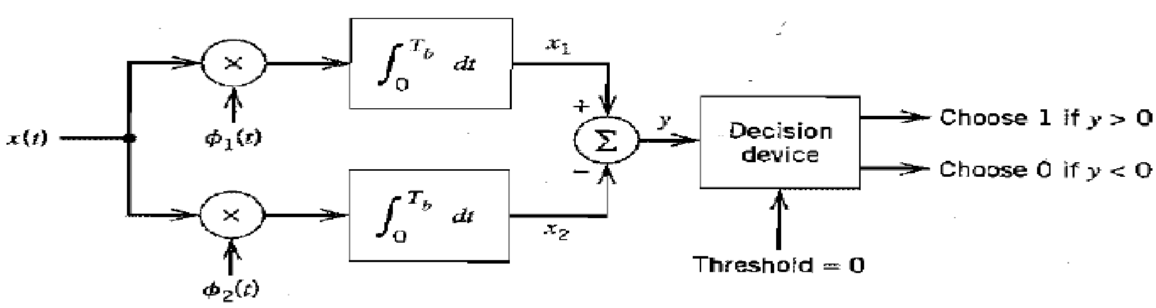
1. The binary sequence b(t) same as ϕ1(t)in is directly applied to the top level shifter to get P $_H$(t)

2. The binary sequence b(t) is applied to the inverter to get $P_L(t)$ and is applied to bottom level shifter.
3. The level shifter converts the +1 level to $\sqrt{Ps\,Tb}$ and zero level to 0.
4. Then the level shifter signals are multiplied with the carrier ϕ1(t)& ϕ2(t)in the product modulator.
5. Then the top and bottom modulator signal s are added in the summer to get BFSK signal .
6. If $P_H(t)$ =1 ,the output is due to the top modulator and the output due to lower modulator is 0(since $P_{L(t)}$=0).
7. The Fsk output is $\mathbf{s(t) = \sqrt{2Ps}\, P_H(t) COS(2\pi f_H t) + \sqrt{2Ps}\, P_L(t) COS(2\pi f_L t)}$

FSK WAVEFORM:

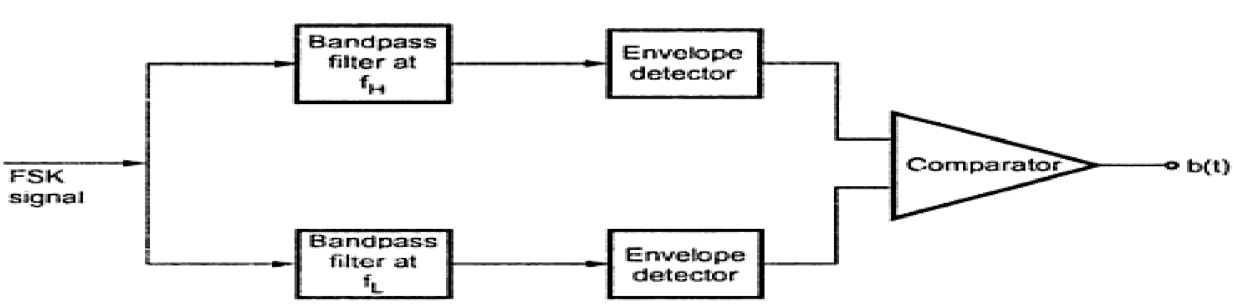


COHERENT FSK RECEIVER:

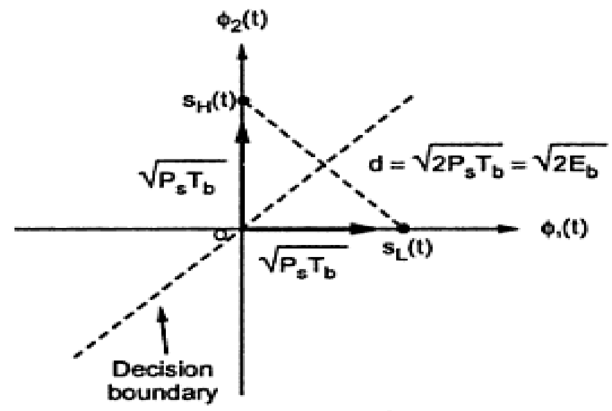


1. The received FSK+noise signalis applied to correlator.
2. The correlator contains product modulator followed by an integerator.
3. The received signalx(t) is multiplied by **ϕ$_1$(t) in the top modulator and integerated**
4. The received signalx(t) is multiplied by **ϕ$_2$(t) in the bottom modulator and integerated.**
5. The correlator output s are then substracted and then it is applied to the decision device.
6. The device compares the difference output y with a threshold of zero volts.
7. if the transmitted frequency is f_H ,the output x1is higher than x2.hence y is greater than zero.
   - ***If y>0 the receiver decides in the favor of 1.***
   - ***If y<0 0the receiver decides in the favor of 0.***
   - ***If y=0 the receiver decides in the favor of 1 or 0.***

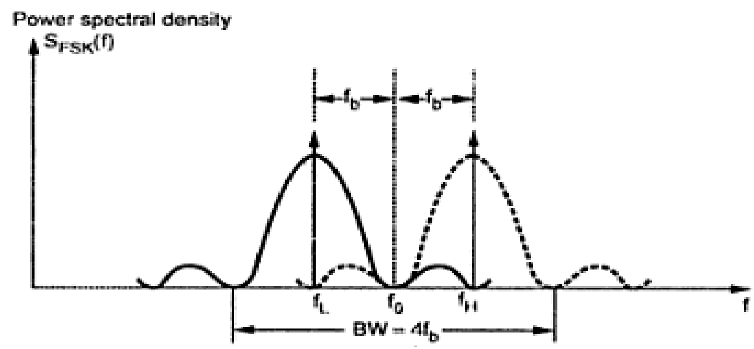
NON-COHERENT FSK RECEIVER:



1. The receiver consists of two bandpass filters one with centre frequency f_H and other with centre frequency f_L.
2. The received signalis filtered and applied to the envelope detector.
3. The output of the detectors are compared by the comparator.
4. The comparator responds to the largest value.



SIGNAL SPACE REPRESENTATION OF FSK:



- for geometrical representation of FSK signal two orthogonal carriers are required **$\phi_1(t)$ & $\phi_2(t)$.**
- the frequencies f_H and f_L are integer multiples of msg freq.

$$f_H = m f_b \qquad f_L = n f_b$$

Here $f_b = \frac{1}{T_b}$, then the carriers will be

$$\phi_1(t) = \sqrt{\frac{2}{T_b}} \cos(2\pi m f_b t)$$

$$\phi_2(t) = \sqrt{\frac{2}{T_b}} \cos(2\pi n f_b t)$$

---------------------(1)

The fsk signalis given as

$$s_H(t) = \sqrt{P_s T_b} \sqrt{\frac{2}{T_b}} \cos(2\pi f_H t)$$

$$s_L(t) = \sqrt{P_s T_b} \sqrt{\frac{2}{T_b}} \cos(2\pi f_L t)$$

---------------(2)

Here $f_H = f_0 + \frac{\Omega}{2\pi}$ and $f_L = f_0 - \frac{\Omega}{2\pi}$

sub eqn (1) in (2)

$$s_H(t) = \sqrt{P_s T_b} \cdot \phi_1(t)$$

$$s_L(t) = \sqrt{P_s T_b} \cdot \phi_2(t)$$

Distance between signal points :

There are two signal points $S_H(t)$ & $S_L(t)$.

The distance between two points

$$d^2 = (\sqrt{P_s T_b})^2 + (\sqrt{P_s T_b})^2$$

$$= 2P_s T_b$$

$$d = \sqrt{2P_s T_b}$$

Sub $P_s T_b = E_b$ then d= $\sqrt{2}$E b

SPECTRUM AND BANDWIDTH OF FSK:

f_H and f_L are selected. Such that,

$$f_H - f_L = 2f_b$$

BANDWIDTH OF FSK:

The width of one lobe is $2f_b$.two lobes are present

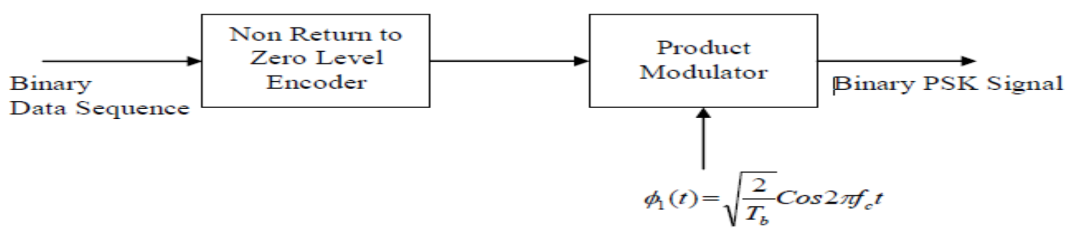
therefore **bandwidth =$2f_b$ +$2f_b$**

b.w = $4f_b$

BINARY PHASE SHIFT KEYING:

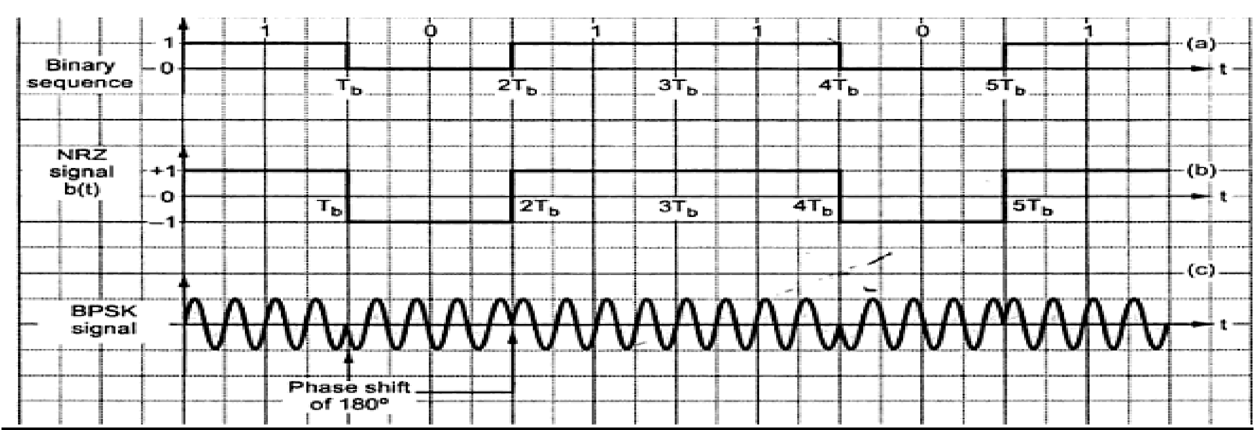
- The phase of the carrier is shifted between two values according to the original data
  - *If the digital data is at **logic 1** ,the **output phase is 0.***
  - *If the digital data is at **logic 0** ,the **output is 180 deg.***
- *Since there are two phases it is called **biphase modulation or phase reversal keying.***

Generation of BPSK signals or BPSK transmitter:



- First the binary sequence is converted to polar form using NRZ encoder.
- **Symbol 1 → represented by an amplitude of $+\sqrt{E_b}$**
- **Symbol 0 → represented by an amplitude of $-\sqrt{E_b}$.**
- This binary wave and a sinusoidal carrier wave $\emptyset_1(t)$ are multiplied in a product modulator
- The desired PSK wave is obtained at the modulator output.

BPSK WAVEFORMS:



MATHEMATICAL ANALYSIS:

- The carrier is defined as **$s(t)= A\cos(2\pi f_0 t)$.**

  Where A=$\sqrt{2P}$

- In a coherent PSK system the pair of signals

$s_1(t)$→Represents binary logic 1

$s_2(t)$→Represents binary logic 0

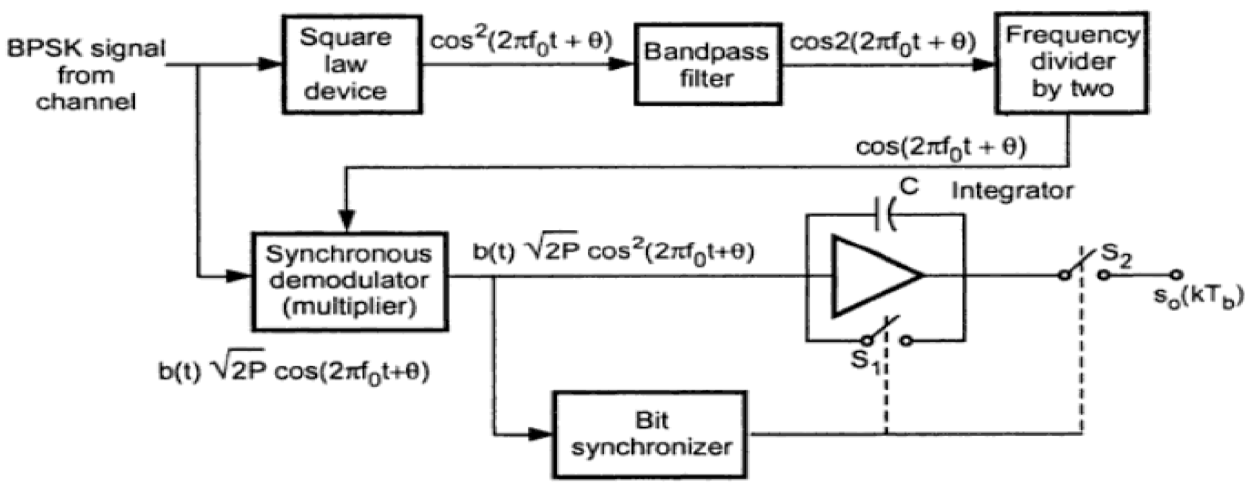
$$s_1(t) = \sqrt{2P}\cos(2\pi f_o t) \quad \text{..................(1)}$$

$$s_2(t) = \sqrt{2P}\cos(2\pi f_0 t + \pi) = -\sqrt{2P}\cos(2\pi f_o t) \text{.......(2)}$$

- The PSK signal is represented as

  S(t) =b(t) $\sqrt{2P}\cos(2\pi f_o t)$
  - b(t)=1 for symbol 1
  - b(t) =0 for symbol 0

RECEPTION OF PSK SIGNAL:



- The Carrier is separated from the received PSK signalby coherent detection.
- The signalat the inputof receiver is **S(t) =b(t)** $\sqrt{2P}\cos(2\pi f_o t + \theta)$
- The received signalis passed through square law device.
- The output of square law device is **COS^2(2πf$_o$t+θ)**

$$\cos^2\theta = \frac{1+\cos 2\theta}{2}$$

$$\cos^2(2\pi f_0 t+\theta) = \frac{1+\cos 2(2\pi f_0 t+\theta)}{2}$$

$$= \frac{1}{2}+\frac{1}{2}\cos 2(2\pi f_0 t+\theta)$$

- this signal is passed through a bandpass filter of cut off frequency **2 f$_o$**.
- the BPF removes the Dc level ½ and gives only **cos2(2πf$_o$t+θ).**
- The frequency divider divides the **frequency 2fo to fo** ie the output is **cos (2πf$_o$t+θ).**
- The synchronous modulator multiplies the inputsignaland the recovered carrier .
- The output of the multiplier is
- The multipler output is given to the bit synchronizer and the integrator.
- The integrator integerates the signal over one bit period .
- The bit synchronizer decideds the starting and ending times of a bit.
- At the end of bit duration T_b ,the bit synchronizer closes switch s2 to connect the output of an integrator to the decision device.
- Then the synchronizer opens the switch s2 and connect s1.
- This resets the the integrator voltage to zero.the integrator then integerates the next bit.
- The phase change of the carrier occurs only at the zero crossing.

SIGNAL SPACE REPRSENTATION OF PSK SIGNALS:

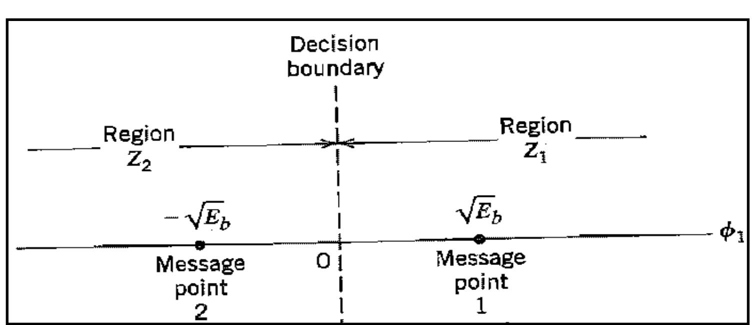
- The PSK signal is represented as s(t) = b(t) $\sqrt{2P}\cos(2\pi f_o t)$
- **Re-arranging,**

$$S(t) = b(t)\sqrt{P\,Tb}\sqrt{\frac{2}{Tb}}\cos(2\pi f_o t)$$

- Let $\phi_1(t) = \sqrt{\frac{2}{Tb}}\cos(2\pi f_o t)$ **be the carrier signal**

Therefore ,

$$S(t) = b(t)\sqrt{PTb}\,\phi_1(t)$$



- The bit energy E_b is defined in terms of power P and bit duration T_b , $\boldsymbol{Eb = P\,T_b}$
- Therefore $s(t) = \pm\sqrt{Eb}\,\phi_1(t)$
- **Therefore the axis $\phi_1(t)$ contains two points** $+\sqrt{Eb}, -\sqrt{Eb}$ corresponds to the symbol 1 and 0

❖ **The distance** $d = +\sqrt{Eb} - (-\sqrt{Eb})$

$$= 2\sqrt{Eb}$$

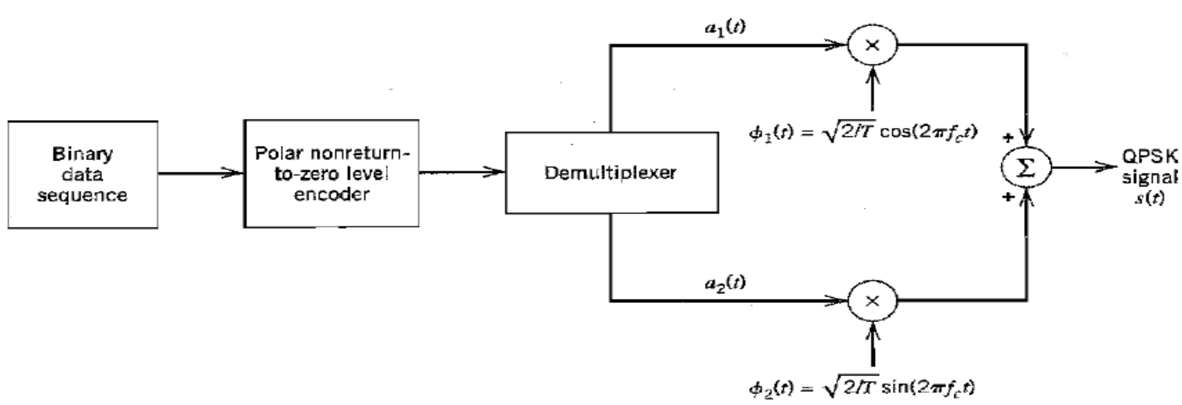
BANDWIDTH OF PSK SIGNAL :

- The spectrum of psk signal is centerd at the carrier frequency f_o and extends from $f_o - f_b$ to $f_o + f_b$
- Therefore **bandwidth = highest frequency –lowest frequency =** $f_o + f_b - (f_o - f_b) = 2f_b$.
- **Thus the minimum bandwidth of PSK signal is equal to twice the highest frequency of msg signal .**

QPSK (QUARTERNARY PHASE SHIFT KEYING):

- **quadriphase shift keying is a technique in which the information is contained in the phase**
- It is a M-ary technique with **M=4; N=2.**
- **Therefore 4 –output phases are possible for a single carrier freq.**

QPSK TRANSMITTER:



(a)

- the inputbinary sequence is converted into polar form with

  symbol 1 → represented by $+\sqrt{Eb}$

  symbol 0 → represented by $-\sqrt{Eb}$
- the demultiplexer divides the binary wave into even and odd numbered input bits .
- these two binary waves are denoted by $b_1(t), b_2(t)$.
- the amplitude of $b_1(t)$ & $b_2(t)$ equals s_{i1} & s_{i2} respectively.
- Then this binary i/ps are multiplied by the carrier $\phi_{1(t)}$ & $\phi_2(t)$ where

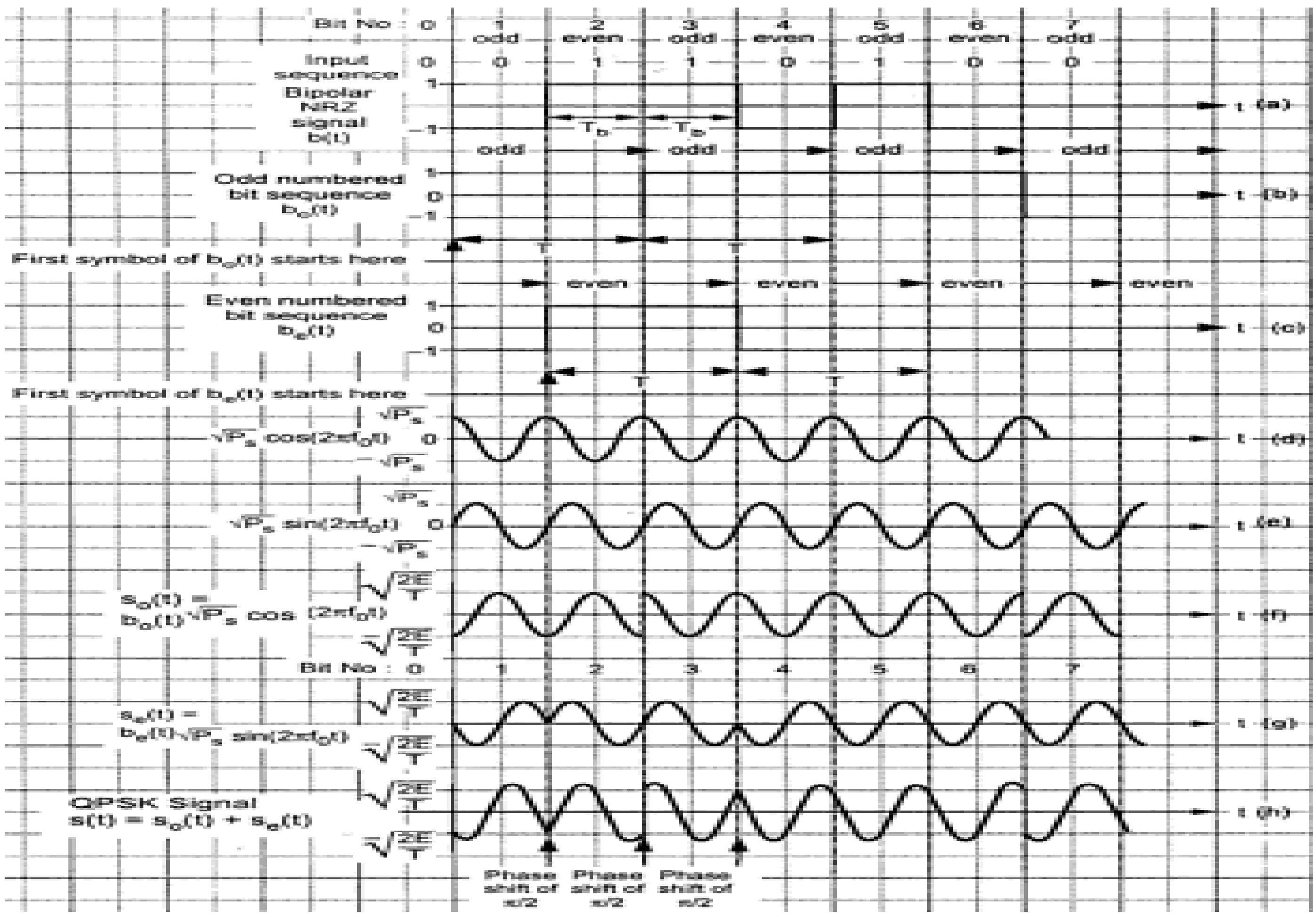
$$\phi_1(t) = \sqrt{\frac{2}{T}}\cos(2\pi f_c t), \qquad 0 \le t \le T$$
$$\phi_2(t) = \sqrt{\frac{2}{T}}\sin(2\pi f_c t), \qquad 0 \le t \le T$$

- Finally the two binary waves are added to produce the desired QPSK waveform.
- QPSK carries twice as many bits of information for a given bandwidth.

QPSK WAVEFORMS:

- The even bit occurs after the odd bit, therefore even bit sequence $b_e(t)$ is delayed by a time period T_b.
- The first symbol of $b_e(t)$ is delayed by a time period T_b with respect to the odd sequence bo(t).
- This time delay is called offset.
- Therefore $b_e(t)$ and $b_o(t)$ doesnot occur at the same time ,so the max phase shift is $\boldsymbol{\pi/2}$.

s.no	**inputbits**		**symbol**	**Phase shift in carrier**
i=1	**1**	**0**	**S_1**	**π/4**
i=2	**0**	**0**	**S_2**	**3π/4**
i=3	**0**	**1**	**S_3**	**5π/4**
i=4	**1**	**1**	**S_4**	**7π/4**



Mathematical expression:

- ✓ The even bit stream $b_e(t)$ modulates the carrier $\sqrt{\mathbf{Ps}}\,\mathbf{sin}(\mathbf{2\pi f_o t})$
- ✓ The odd bit stream $b_o(t)$ modulates the carrier $\sqrt{\mathbf{Ps}}\,\mathbf{cos}(\mathbf{2\pi f_o t})$.
- ✓ **Therefore the output of the two modulators are**

$$s_e(t) = \mathbf{be(t)}\,\sqrt{\mathbf{Ps}}\,\sin(2\pi f_o t)$$

$$s_o(t) = \mathbf{bo(t)}\,\sqrt{\mathbf{Ps}}\,\cos(2\pi f_o t)$$

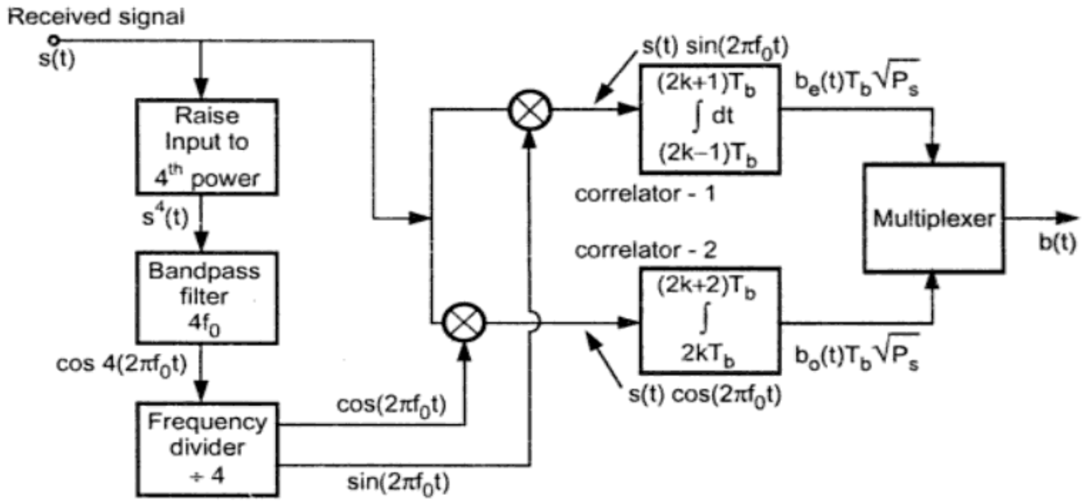
- ✓ The adder add these output to give QPSK signal,

$$\mathbf{S(t)} = s_e(t) + s_o(t)$$

$$= \mathbf{be(t)}\sqrt{\mathbf{Ps}}\,\sin(2\pi f_o t) + \mathbf{bo(t)}\,\sqrt{\mathbf{Ps}}\,\cos(2\pi f_o t)$$

QPSK RECEIVER(COHERENT DETECTION):

1. The carrier is recovered from the received signals(t).
2. First the received signals(t) power is raised to 4^{th} power ie $s^4(t)$.
3. Then it is passed through the bandpass filter of cut-off frequency $4f_o$.
4. The BPF allows only the carrier frequency $4f_o$.
5. Then the filtered output is divided by 4 and the output s are $\cos(2\pi f_o t)$ and $\sin(2\pi f_o t)$.
6. This coherent carriers are applied to synchronous demodulators.
7. The demodulator consists of multiplier and integrator.
8. The incoming signal is applied to both the multipliers .the integeratoe integerates the multiplier output over the time period 2 T_b.



9. Then the output of integrator is sampled at the time instant T_b.
10. The even and odd sequences obtained are multiplexed.
11. The top multiplier output is

$$\boxed{s(t)\sin(2\pi f_o t) = be(t)\sqrt{Ps}\,\sin 2(2\pi f_o t) + bo(t)\sqrt{Ps}\cos(2\pi f_o t)\sin(2\pi f_o t) \text{---------------}(1)}$$

This signalis integrated by the top integrator $\int_{(2k-1)Tb}^{(2k+1)Tb} s(t)\ \sin(2\pi f_o t)$

$= bo(t)\sqrt{Ps}\int_{(2k-1)Tb}^{(2k+1)Tb}\cos(2\pi f_o t)\sin(2\pi f_o t)\,dt + be(t)\sqrt{Ps}\int_{(2k-1)Tb}^{(2k+1)Tb}\sin 2(2\pi f_o t)\,dt$ ---------

(2)

Since **1/2 sin(2x) = sin x.cos x**

$\sin^2 x$ = ½[1-cos2x]

using these formula the above eqn becomes

eqn 2→ $= \frac{bo(t)\sqrt{Ps}}{2}\int_{(2k-1)Tb}^{(2k+1)Tb}\sin(4\pi f_o t)\,dt + \frac{be(t)\sqrt{Ps}}{2}\int_{(2k-1)Tb}^{(2k+1)Tb} dt - \frac{be(t)\sqrt{Ps}}{2}\int_{(2k-1)Tb}^{(2k+1)Tb}\cos(4\pi f_o t)\,dt$ -----

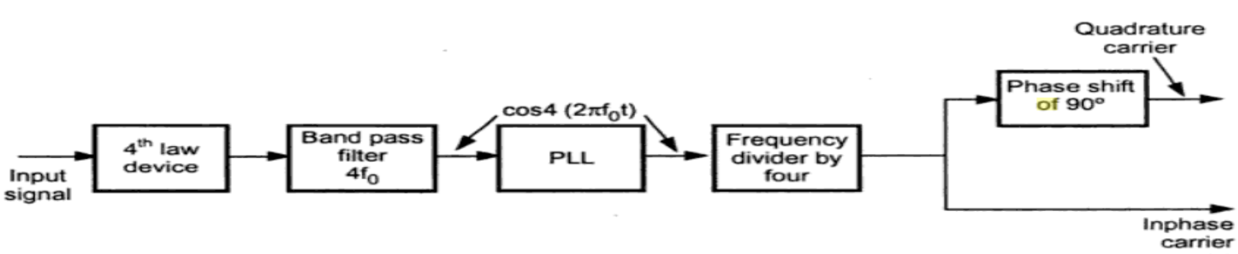
-(3)

first and third term becomes zero therefore

$$\boxed{\int_{(2k-1)Tb}^{(2k+1)Tb} s(t)\ \sin(2\pi f_o t) = \frac{be(t)\sqrt{Ps}}{2}[t] = b_e(t)\sqrt{Ps}Tb}$$

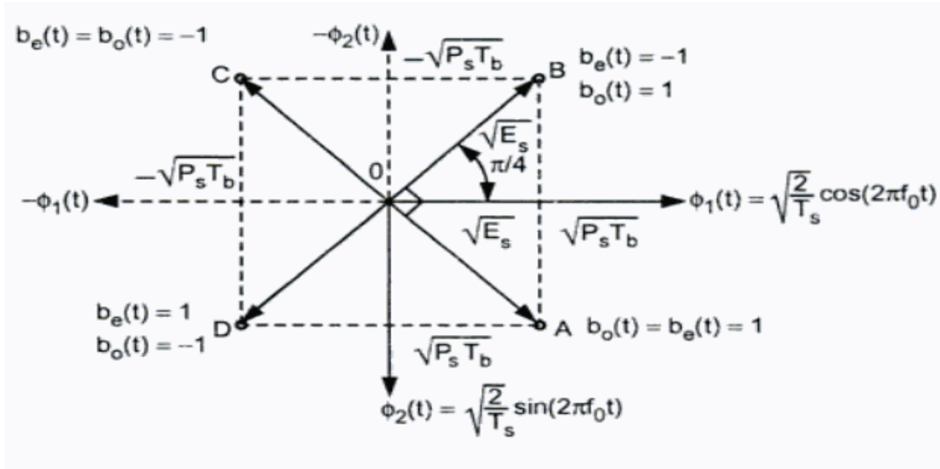
similarly we can obtain the output of bottom integrator as $\boxed{b_o(t)\sqrt{Ps}Tb}$

<u>Carrier synchronization in qpsk:</u>



- in coherent detection the carrier are to be synchronized properly.
- For carrier synchronization we are using PLL system.
- The inputsignalis raised to fourth power and filtered by BPF .

- The output of the filter contains the frequency component $4f_o$.
- We know that **$\cos 4(2\pi f_o t) = \cos(8\pi f_o t + 2\pi N)$.**
  $N \rightarrow$ integer value, no. of cycles.
- **When the output is divided by 4 the eqn becomes $\cos[2\pi f_o t + N\pi/2]$**
- The output has a fixed phase error of $N\pi/2$.



SIGNALSPACE REPRESENTATION OF QPSK:

- ✓ The QPSK signalis given as $s(t) = \sqrt{2Ps}\cos[(2\pi f_o t) + \frac{(2m+1)\pi}{4}]$ **m= 0,1,2,3….---------------(1)**
- ✓ expand this using **cos(A+B)= cosAcosB-sinAsinB**

$$s(t) = \sqrt{2Ps}\cos(2\pi f_o t)\cos\frac{(2m+1)\pi}{4}] - \sqrt{2Ps}\,s\,in(2\pi f_o t)\sin\frac{(2m+1)\pi}{4}] \text{------(2)}$$

$$s(t) = \{\sqrt{PsTs}\cos\frac{(2m+1)\pi}{4}\}\sqrt{\frac{2}{Ts}}\cos(2\pi f_o t)] - \{\sqrt{PsTs}\sin\frac{(2m+1)\pi}{4}\}\sqrt{\frac{2}{Ts}}\sin(2\pi f_o t)] \text{-----(3)}$$

let $\phi_1(t) = \sqrt{\frac{2}{Ts}}\cos(2\pi f_o t)$ **and**

$$\phi_2(t) = \sqrt{\frac{2}{Ts}}\sin(2\pi f_o t) \text{ -----------------------------(4)}$$

and $b_o(t) = \sqrt{2}\cos\frac{(2m+1)\pi}{4}$

$$b_o(t) = -\sqrt{2}\sin\frac{(2m+1)\pi}{4} \text{ --------------------(5)}$$

sub eqn 4,5 in eqn 3

eqn3$\rightarrow$ $s(t) = \sqrt{PsTs}\,bo(t)\,\phi 1(t) + \sqrt{PsTs}\,be(t)\,\phi 2(t)$

$$= \sqrt{PsTs/2}\,bo(t)\,\phi 1(t) + \sqrt{PsTs}/2be(t)\,\phi 2(t) \text{ ------------(6)}$$

Since $T_s = 2T_b$; $T_b = T_s/2$

$S(t) = \sqrt{PsTb}\,bo(t)\,\phi 1(t) + \sqrt{PsTb}be(t)\,\phi 2(t)$ -----------------(7)

Since $E_b = P_s T_b$

$S(t) = \sqrt{Eb}\,bo(t)\,\phi 1(t) + \sqrt{Eb}\,be(t)\,\phi 2(t)$ -----------------------(8)

Distance OB = $\sqrt{PsTb + PsTb}$

$= \sqrt{2PsTb}$

$=\sqrt{PsTs}$ since $T_s = 2T_b$

$=\sqrt{Es}$ since $E_s = P_s T_s$

The distance between signal points is given as

$d^2 = (\sqrt{Es})^2 + (\sqrt{Es})^2$

$d = \sqrt{2Es}$: $d = 2\sqrt{PsTb} = 2\sqrt{Eb}$

<u>BANDWIDTH</u>:

B.W= highest frequency –lowest frequency

$= [1/T_s] - [-1/T_s]$

$= 2/T_s = 2/2T_b$

$BW = 1/T_b = f_b$

<u>Advantages of QPSK:</u>

1. Reduced bandwidth.
2. High transmission rate.
3. Constant carrier power.

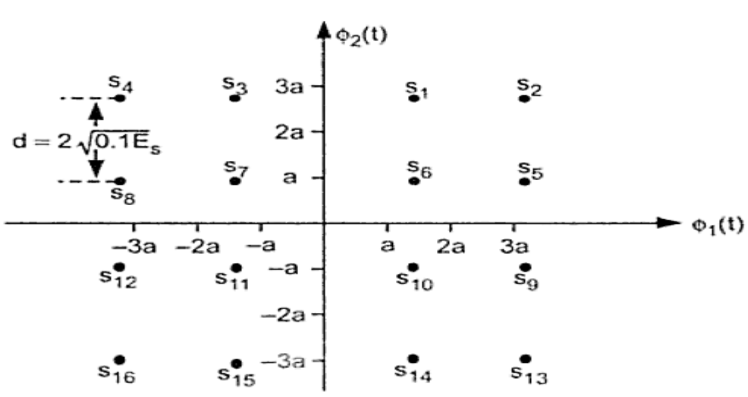
<u>Disadvantage :</u>

- ★ **Phase ambiguity** is present due to the squaring of received signalat the rxer.

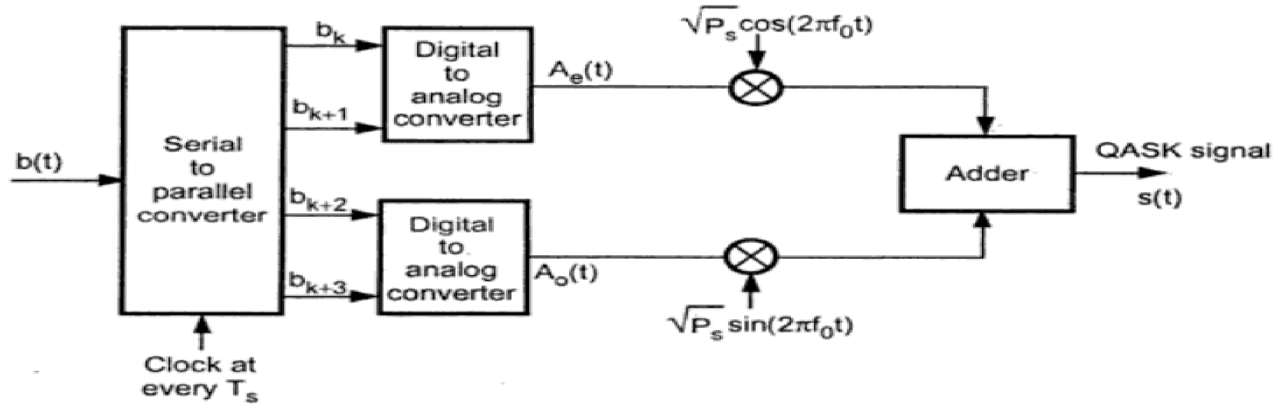
<u>QUADRATURE AMPLITUDE SHIFT KEYING (QASK) OR QUADRATURE AMPLITUDE MODULATION (QAM):</u>

QAM is a technique in which both the amplitude and phase of the carrier is varied in accordance with the digital data.

<u>Signal pace representation of 16- QASK signal:</u>

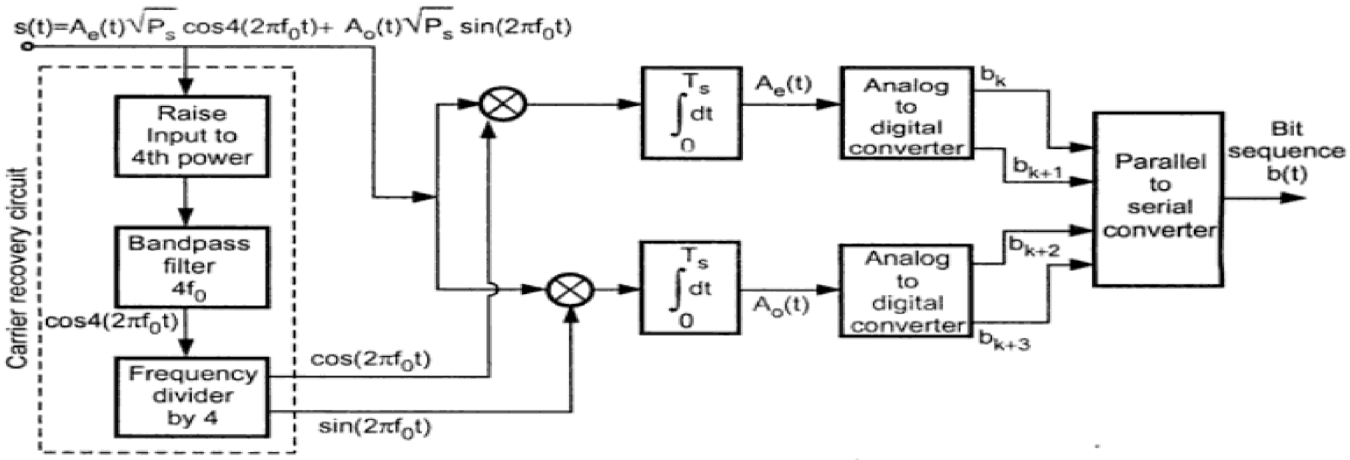


<u>16-QASK TRANSMITTER</u>



1. First the input signal is given to the sertial to parallel converter .
2. Four bits are applied to the digital to analog converter every T_s seconds .
3. Bits b_k and b_{k+1} are applied to top d/a converter.
4. Bits b_{k+2} and b_{k+3} are applied to bottom d/a converter.
5. The top D/A output $A_e(t)$ is multiplied by the carrier $\sqrt{\mathbf{Ps}}\,\mathbf{cos(2\pi f_o t)}$
6. The bottom D/A output $A_o(t)$ is multiplied by the carrier $\sqrt{\mathbf{Ps}}\,\mathbf{sin(2\pi f_o t)}$
7. The adder combines the two signals to get QASK signal .
   $S(t) = A_e(t)\sqrt{\mathbf{Ps}}\,\mathbf{cos(2\pi f_o t)} + A_o(t)\sqrt{\mathbf{Ps}}\,\mathbf{sin(2\pi f_o t)}$
   Where $A_e(t) \rightarrow \mp\sqrt{\mathbf{0.2}}$ **or** $\mp 3\sqrt{\mathbf{0.2}}$

16-QASK receiver:



1. The inputsignals(t) is raised to 4^{th} power.
2. It is then passed through BPF of cut off frequency 4 F_o.
3. The filtered signalis divided by 4 to get the carrier $\cos(2\Pi f_o t)$
4. Quadrature carrier $\sin(2\Pi f_o t)$ is produced by phase shifting of 90.
5. Then both the carriers are multiplied with the QASK signal.

   The QASK signalis $\mathbf{S(t) = A_e(t)}\sqrt{\mathbf{Ps}}\,\mathbf{cos(2\pi f_o t)} + \mathbf{A_o(t)}\sqrt{\mathbf{Ps}}\,\mathbf{sin(2\pi f_o t)}$

6. the 4th power of QASK signalis

   $$\mathbf{S^4(t) = Ps^2\,[Ae(t)\sqrt{Ps}\cos(2\pi f_o t) + A_o(t)\sqrt{Ps}\sin(2\pi f_o t)]^4}$$

7. **The signalis passed through the BPF of 4 Fo. then**

$$\mathbf{S^4(t) = Ps/8\,[Ae^4(t)+ Ao^4(t)\text{-}6\,Ae^2(t)\ Ao^2(t)]\cos4(2\pi f_o t)+ Ps/2[Ae(t)\,Ao(t)Ae^2(t)-Ao2(t)\,\}]\sin4(2\pi f_o t)}$$

8. The average value of second term is zero,hence bandpass filter passes only first term.
9. The integerators integerate the multiplied signalover one symbol period.
10. The output of the integerators at sampling period give $A_e(t)$ and $A_o(t)$.
11. The A/D converter gives the four bits b_k ,b_{k+1} b_{k+2} ,b_{k+3}.
12. The parallel to serial converter generates the bit sequence b(t).

bandwidth:

bandwidth= f_s-(-f_s)

$=2f_s = 2/T_s$

$= 2/NT_b = 2f_b/N$

MSK (MINIMUM SHIFT KEYING):

- ***Minimum-shift keying (MSK) is a type of continuous-phase frequency-shift keying, With a frequency separation of one-half the bit rate (0r) With h = (1/2) the frequency deviation is half the bit rate.***
- This minimum frequency spacing allows the two FSK signals corresponding to symbols 1 and 0 to be coherently orthogonal,
- So That they do not interfere with each other in the process of detection.
- **Cpfsk signal with the deviation ratio as (1/2) is commonly referred to as minimum shift keying.**
- The MSK signalis expressed as $\mathbf{S(t) = S_1\,\phi 1(t) + S2\,\phi 2(t)}$
- Where $\mathbf{\phi_1(t)} = \sqrt{\frac{2}{Tb}}\cos\left(\frac{\pi}{2Tb}t\right)\cos(2\pi f_c t)$ $\quad -T_b \le t \le T_b$

$$\phi_2(t) = \sqrt{\frac{2}{Tb}}\sin\left(\left(\frac{\pi}{2Tb}\right)t\right)\sin(2\pi f_c t) \quad 0 \le t \le T_b$$

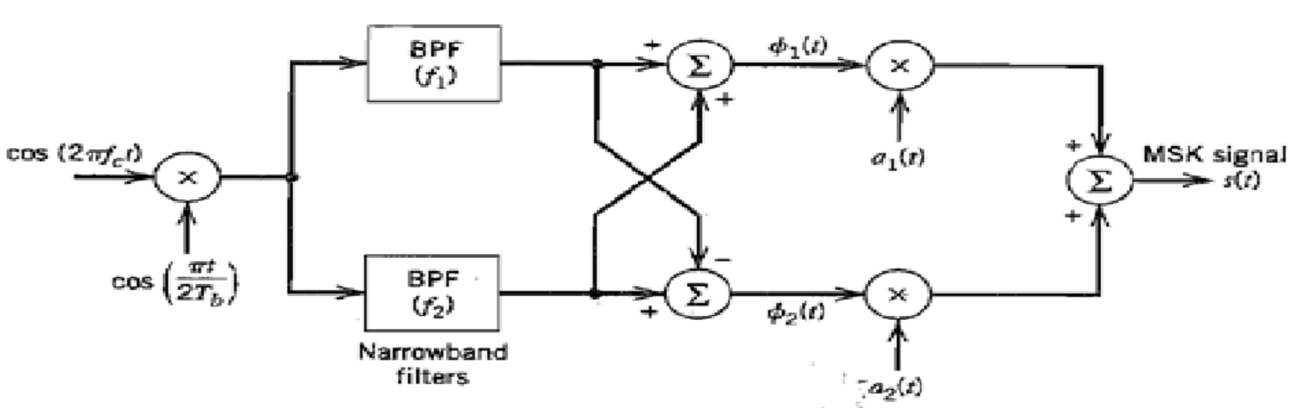
$S_1 \rightarrow \sqrt{Eb}\cos(\theta(0))$	$-T_b \le t \le T_b$
$S_2 \rightarrow -\sqrt{Eb}\sin(\theta(Tb))$	$0 \le t \le 2T_b$

SIGNAL SPACE REPRSENTATION OF MSK:

The MSK can signal may assume one of the four possible states depending on the values of θ(0) and θ(Tb).

Transmitted binary symbol $0 \le t \le T_b$	Phase states		message	
	$\theta(0)$	$\theta(Tb)$	S1	S2
1	0	$+\pi/2$	$+\sqrt{Eb}$	$-\sqrt{Eb}$
0	π	$+\pi/2$	$-\sqrt{Eb}$	$-\sqrt{Eb}$
1	π	$-\pi/2$	$-\sqrt{Eb}$	$+\sqrt{Eb}$
0	0	$-\pi/2$	$+\sqrt{Eb}$	$+\sqrt{Eb}$

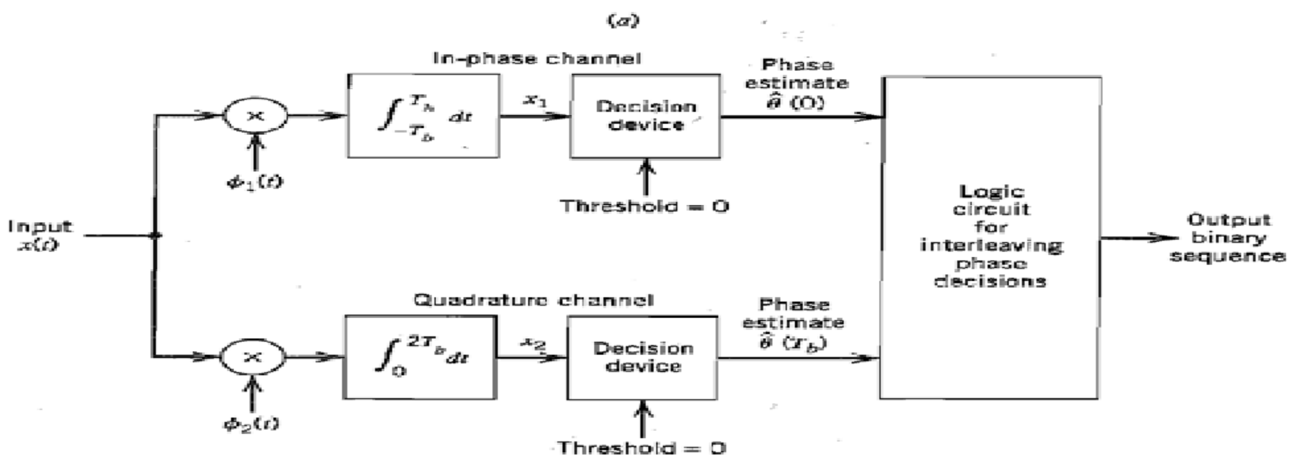
MSK TRANSMITTER:



.

- Two sine i/ps, one of frequency $f_c = n_c/4t_b$ and the other of frequency $1/4t_b$, are applied to a product modulator.
- This produces two phase-coherent sine waves are separated at frequencies f_1 and f_2 .
- The top most BPF allows only the signal f1.
- The bottommost BPF allows only the signal f2.
- Then the resulting filter output s are summed to produce $\emptyset_1(t)$ and $\emptyset_2(t)$.
- Then $\emptyset_1(t)$ and $\emptyset_2(t)$ are multiplied by message signal $m_1(t)$ and $m_2(t)$, with a bit rate of $1/2t_b$.
- This produces MSK waveforms.

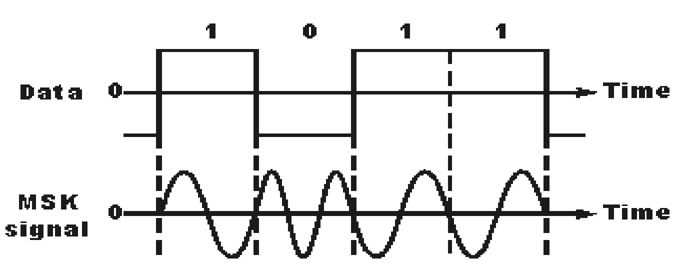
MSK RECEIVER:



- The received signal x$(t)$is correlated with locally generated carrier, $\emptyset_1(t)$ and $\emptyset_2(t)$.
- Then it is integerated over Tb seconds in inphase channel $2t_b$ seconds in quadrature channel .
- Then the correlator outputs, $x_1(t)$and $x_2(t)$, are compared with a threshold of zero volts,
- The estimates of the phase $\theta(0)$ and $\theta(t_b)$ are obtained.
- Then this phase estimates is given to the logic circuit and original binary waves are reconstructed.

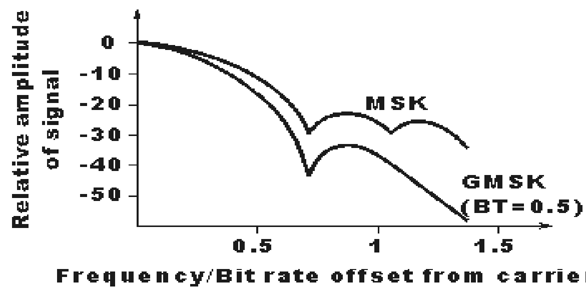
GMSK: (Gaussian filtered Minimum Shift Keying):

- ✖ GMSK, is a form of modulation used in a **variety of digital radio communications systems.**
- ✖ It has advantages **of *being able to carry digital modulation using the spectrum efficiently.***
- ✖ The problem in other techniques is th*at **the sidebands extend outwards from the main carrier and these can cause interference*** to other radio communications systems using nearby channels.
- ✖ A plot of the spectrum of an MSK signal shows *sidebands extending well beyond a bandwidth equal to the data rate.*



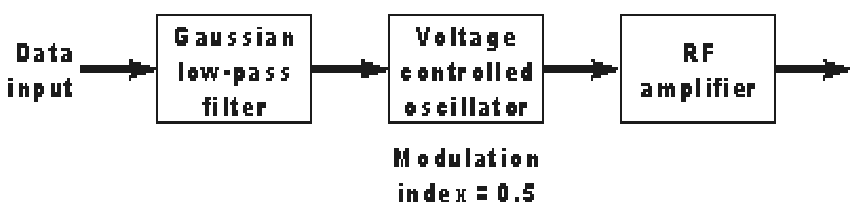
- ✖ This can be reduced by passing the modulating signal through a low pass filter prior to applying it to the carrier.

- The requirements for the filter are that
    - ***it should have a sharp cut-off, narrow bandwidth***
    - ***and its impulse response should show no overshoot.***

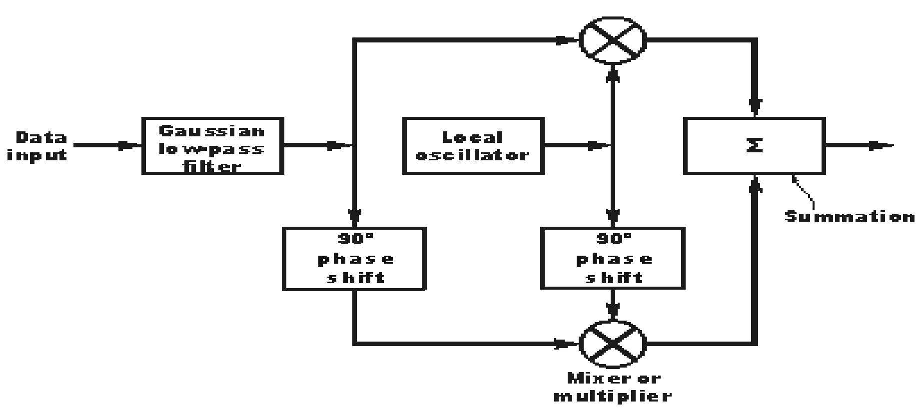


- To achieve the above condition the high frequency components should be removed and excess frequency deviations are avoided .
- This can be achieved by passing the NRZ data to a pulse shaping filter defined by a Gaussian function .In this way the basic MSK signal is converted to GMSK modulation

GMSK MODULATOR



- The msg signalis filtered by **Gaussian filter and applied to the freq. modulator with h=0.5.**
- This method is not suitable because component tolerances drift and cannot be set exactly.



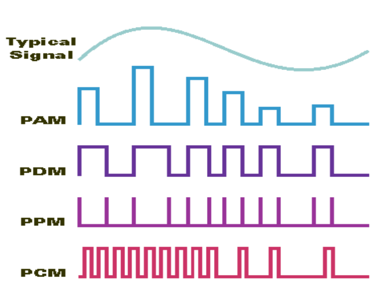
Explanation:

- First the inputdata is Gaussian filtered and is applied directly to the modulator.
- The inputsignalis 90 deg phase shifted and the carrier is 90 deg phase shifted and is applied to the multiplier .
- Then the output s of both the multi[pliers are added by the summer to get GMSK signal.

ADVANTAGES:

- improved spectral efficiency
- is immune to amplitude variations and therefore more resilient to noise.
- it can be amplified by a non-linear amplifier and remain undistorted .

WAVEFORMS:



UNIT III SOURCE CODES, LINE CODES & ERROR CONTROL (Qualitative only)

Primary communication – entropy, properties, BSC, BEC, source coding: Shannon's Fano, Huffman coding: noiseless coding theorem, BW – SNR trade off codes: NRZ, RZ, AMI, HDBP, ABQ, MBnB codes: Efficiency of transmissions, error control codes and applications: convolutions & block codes.

Primary Communication

- The performance of the communication system is measured in the terms of its error probability. An errorless transmission is possible when probability of error at the receiver approaches zero.
- The performance of the system depends upon available signal power, channel noise and bandwidth. Based on these parameters it is possible to establish the condition for error-less transmission. These conditions are referred as Shannon's theorems.
- The information theory is related to the concepts of statistical properties of message (or source), channels, noise interference etc. the information theory is used for mathematical modeling and analysis of the communication systems.
- With the information theory and its modeling for communication systems, following two main points are resolved:
  1. The irreducible complexity below which the signal cannot be compressed.
  2. The transmission rate for reliable communication over a noise channel.

Amount of Information

Let consider the communication system which transmits messages m1, m2, m3,….., with probabilities of p1, p2, p3,…..,

The amount of information through the message $\mathbf{m_k}$ with probability $\mathbf{p_k}$ is given as,

Amount of information $\mathbf{I_k = \log_2 (1/ p_k)}$

Unit of information:

The above equation, $\mathbf{\log_2 (1/ p_k) = [\log_2 (1/ p_k) / \log_{10} 2]}$

Properties of Information:

1. $\mathbf{I(s_k) = 0}$ **for** $\mathbf{p_k = 1}$
   - If we are absolutely certain of the outcome of an event, even before it occurs, then there is no information gained.
2. $\mathbf{I(s_k) \geq 0}$ **for** $\mathbf{0 \leq p_k \leq 1}$
   - The occurrence of an event $S=s_k$ either provides some or no information but never brings about a loss of information.
3. $\mathbf{I(s_k) > I(s_i)}$ **for** $\mathbf{pk < p_i}$
   - That is, the less probable an event is, the more information we gain when it occurs.
4. $\mathbf{I(s_k s_i) = I(s_k) + I(s_i)}$ **,if** $\mathbf{s_k}$ and $\mathbf{s_i}$ are statistically independent.

Information Rate:

The information rate is represented by R and it is given as,

$$\boxed{\text{Information Rate : } R = rH} \quad ...(1.7.1)$$

Here R is information rate.

H is entropy or average information

and r is rate at which messages are generated.

Information rate R is represented in average number of bits of information per second. It is calculated as follows :

$$R = \left(r \text{ in } \frac{\text{messages}}{\text{second}}\right) \times \left(H \text{ in } \frac{\text{Information bits}}{\text{message}}\right)$$

$$= \text{Information bits / second}$$

ENTROPY: (the average information per message is called entropy. It is represented in bits / message.)

- It is a measure of the average information content per source symbol.
- The amount of information, $I(s_k)$ produced by the source depends on the symbol 's_k' emitted by the source at that time.
- $I(s_k)$ is the discrete random variable that takes on the values $I(s_0)$,$I(s_1)$,……….., $I(s_{k-1})$ with probabilities $p_o,p_1,\ldots\ldots\ldots p_{k-1}$ respectively.
- The mean(average) value of $I(s_k)$ over the source alphabet 'I' is given by

$H(I) = E[I(s_k)]$

$=\sum_{k=0}^{k-1} p_k \, I(sk)$

$H(I) =\sum_{k=0}^{k-1} p_k \;\mathbf{log_2} \left(\frac{1}{p_k}\right)$

$$\boxed{\text{Entropy : } \quad H = \sum_{k=1}^{M} p_k \log_2 \left(\frac{1}{p_k}\right)}$$

PROPERTIES OF ENTROPY:

- Consider a discrete memory-less source whose output is modeled as a discrete random variable S which takes on symbol from a finite fixed alphabet I: :{s_0, s_1, ………….. s_{k-1}}
- With probabilities **$P(S = s_k) = P_k$** **k=0,1,2,…………,k-1** **---------------(1)**
- This set of probabilities satisfy the condition

  $\boldsymbol{\sum_{k=0}^{k-1} pk =1}$ **----------- (2)**

The entropy H(I) of such a source is bounded as follows: $0 \leq H(I) \leq$ **$\log_2 k$**

1. H(I) = 0 ,if and only if the probability p_k =1 for some k ,and the remaining probabilities in the set are all zero.this lower bound on entropy corresponds to no uncertainty.
2. H(I) = **$\log_2 k$** **,** if and only if p_k=1/K for all k(ie all the symbols in the alphabet are equiprobable).this upper bound on entropy corresponds to maximum uncertainty.

Problems :

Ex:1

Consider a discrete memoryless source with source alphabet I ={so,s1,s2} with probabilities

p(so)= p0= ¼ ; p(s1)=p1= ¼ and p(s2)=p2 =1/2.

Calculate entropy of th e source.

Solution:

$$H(I) = \sum_{k=0}^{k-1} pk\ \mathbf{log2}\ \left(\frac{\mathbf{1}}{\boldsymbol{pk}}\right)$$

$$= \sum_{k=0}^{2} pk\ \mathbf{log2}\ \left(\frac{\mathbf{1}}{\boldsymbol{pk}}\right)$$

$$= p_o \log_2 (1/p0) + p_1 \log_2(1/p1) + p_2 \log_2(1/p2)$$

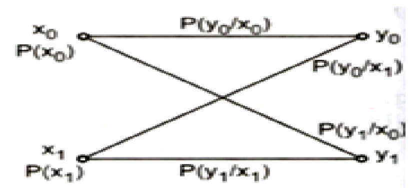
$$H = \sum_{k=0}^{k-1} P_K \left[\log_{10} (1/P_K) / \log 2 \right]$$

$$= (¼) \log_2(4) + (¼)\log_2(4) + (1/2)\log_2(2)$$

H(*I*) = 3/2 bits.

Binary Communication Channel

Consider the case of the discrete channel where there are only two symbols transmitted. Fig. 3.5.1 shows the diagram of a binary communication channel. We can write the equations for probabilities of Y 0 and Y 1 as,



$$P(y_o) = P(y_o / x_o)\ P(x_o) + P(y_o / x_1) P(x_1) \text{ ---------(1)}$$

and $$P(y_1) = P(y_1 / x_1)\ P(x_1) + P(y_1 / x_o) P(x_o) \text{ ---------(2)}$$

Above equations can be written in the matrix form as,

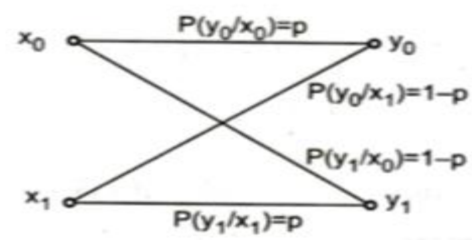
Fig. 1: Binary Communication Channel

$$\begin{bmatrix} P(y_0) \\ P(y_1) \end{bmatrix} = \begin{bmatrix} P(x_0) & P(x_1) \end{bmatrix} \begin{bmatrix} P(y_0 / x_0) & P(y_1 / x_0) \\ P(y_0 / x_1) & P(y_1 / x_1) \end{bmatrix} \text{ -------------------------- (3)}$$

Note that the 2x 2 matrix in above equation is a probability transition matrix.

1. **Binary Symmetric Channel (BSC):**

The binary communication channel of Fig. 2 is said to be symmetric if $P(y_o / x_o) = P(y_1 / x_1) = p$. Such channel is shown in Fig. 2.

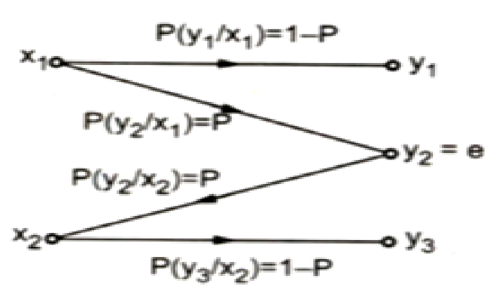


For the above channel, we can write eqn.3 as,

Fig. 2: Binary Symmetric Channel

$$\begin{bmatrix} P(y_0) \\ P(y_1) \end{bmatrix} = \begin{bmatrix} P(x_0) & P(x_1) \end{bmatrix} \begin{bmatrix} p & 1-p \\ 1-p & p \end{bmatrix} \text{ ---------------------- (4)}$$

2. **Binary Erasure Channel (BEC)**



- The channel diagram of binary erasure channel is show n in Fig.3.Note that there are two input symbols and three output symbols.
- The middle symbol (y_2) represents error (e).
- This error is detected and erased. Hence this channel is also called Binary Erasure Channel (BEC).
- From above figure we can write the channel matrix for BEC as follows :

Fig. 3: Binary Erasure Channel

$$P(Y/X) = \begin{bmatrix} P(y_1 / x_1) & P(y_2 / x_1) & P(y_3 / x_1) \\ P(y_1 / x_2) & P(y_2 / x_2) & P(y_3 / x_2) \end{bmatrix} = \begin{bmatrix} 1-p & p & 0 \\ 0 & p & 1-p \end{bmatrix} \text{ --------------------- (5)}$$

SOURCE CODING THEOREM

- Source encoding is the process in which the data generated by the discrete source are efficiently represented by the source encoder.
- Efficient source encoders can be designed which use the statistical properties of the source.
- For example: ***the messages which occur frequently are assigned short codeword.***
  the messages which occur rarely are assigned long codeword.
- The efficient source encoder should satisfy the following requirements.
  - The codeword generated by the encoder should be binary in nature.
  - The source code is uniquely decodable, so that the original sequence can be reconstructed perfectly from the encoded binary sequence.
- Let there be 'K' number of messages emitted by the source.
- The probability of the k^{th} message is p_k and the number of bits assigned to this message is I_k.
- **Then the average number of bits ($\bar{L}$) in the codeword of the message are given as ,**

$$\bar{L} = \sum_{k=0}^{K-1} p_k l_k$$

- The coding **efficiency** of the source encoder is defined as

$$\eta = \frac{L_{\min}}{\bar{L}}$$

L_{min}→denote the minimum possible value of ($\bar{L}$)

With $\bar{L} \geq L_{\min}$, we clearly have $\eta \leq 1$. The source encoder is said to be *efficient* when η approaches unity.

Source coding theorem statement:

SHANNON'S FIRST THEOREM:

Given a discrete memoryless source of entropy $H(\mathscr{S})$, the average code-word length $\overline{L}$ for any distortionless source encoding scheme is bounded as

$$\overline{L} \geq H(\mathscr{S})$$

$H(\mathscr{S})$ → Entropy of the source symbol

If Lmin = $H(\mathscr{S})$, then efficiency $\eta = \frac{H(\mathscr{S})}{\overline{L}}$

CODE REDUNDANCY:

- **It is the measure of redundancy of bits in the encoded message sequence.**
- **Redundancy (γ) = 1- code efficiency**
- **γ = 1-η**
- Redundancy should be as low as possible.

CODE VARIANCE:

- **Variance is the measure of variability in code word lengths.**
- Variance of the code is given as,

$\sigma^2 = \sum_{k=0}^{k-1} \boldsymbol{Pk(Ik - \overline{L}\ )^2}$ $\sigma^2 = \sum_{k=0}^{M} p_k \left[n_k - \overline{N}\right]^2$

σ^2 →variance of the code

(K-1) or M: is the number of symbols;

P_k is the probability of bits assigned to k^{th} symbol.

$\overline{\boldsymbol{L}}$ **or** $\overline{\boldsymbol{N}}$ **→ Average codeword length.**

<u>**SHANNON-FANNON CODING:**</u>

- If the probabilities of occurrence of all the messages are not equally likely, then average information (i.e. entropy) is reduced.
- This in turn reduces the information rate (R).
- This problem is solved by coding the messages with different number of bits.

- As the probability of the message is increased, less number of bits is used to encode it.
- Shannon fanon algorithm is used to encode the messages depending upon their probabilities.
- This algorithm assigns: * less number of bits for frequently occurring message and
  * more number of bits for rarely occurring messages.

CODING STEPS:

1. The messages are given as x1 to x5 (In example problem) the probabilities of the occurrence of messages are arranged in 2nd column.
2. Divide the second column into two partitions such that the sum of the probabilities of upper partition is equal to the lower partition.
3. Then the messages in the upper partition are assigned '0'and the messages in the lower partition are assigned '1'.
4. Then the partitions are further divided into new partitions and the values are assigned following the same rule.
5. This partitioning is continued till there is only one message in the partition.
6. The codeword is obtained by reading the bits of a particular message row wise through all columns.

Example 1:

A discrete memoryless source has five symbols x_1, x_2, x_3, x_4, x_5, with probabilities 0.4,0.19,0.16,0.15 and 0.15 respectively attached to every symbol. Construct Shannon-fano code for the source and calculate code efficiency.

Solution:

message	**Probability of message (p_k)**	**I**	**II**	**III**	**Codeword for message**	**Number of bits per message (l_k)**
x_1	0.4	0			0	1
x_2	0.19	1	0	0	100	3
x_3	0.16	1	0	1	101	3
x_4	0.15	1	1	0	110	3
x_5	0.15	1	1	1	111	3

Entropy:

$$H(I) = \sum_{k=0}^{k-1} pk \; \mathbf{log2} \left(\frac{\mathbf{1}}{\mathbf{pk}}\right)$$

$$H = \sum_{k=0}^{k-1} P_K \left[\log_{10} (1/P_K) / \log 2 \right]$$

$H = 0.4\log_2 (1/0.4) + 0.19\log_2 (1/0.19) + 0.16\log_2 (1/0.16) + 0.15\log_2 (1/0.15) + 0.15\log_2 (1/0.15)$

H = 2.2281 bits/message

Code efficiency:

$$\eta = \frac{H(\mathcal{S})}{\bar{L}}$$

L_{min} the average number of bits per messge is given by

$$\bar{L} = \sum_{k=0}^{K-1} p_k l_k \quad \text{(or)} \quad \bar{N} = \sum_{k=0}^{4} p_k n_k$$

$\bar{L}$ =0.4(1) +0.19(3)+0.16(3)+0.15(3)+0.15(3)

$\bar{L}$ **= 2.35**

Then code efficiency

$$\eta = \frac{H(\mathcal{S})}{\bar{L}}$$

η = (2.2281/2.35)=0.948

HUFFMAN CODING:

- Huffman coding assigns different number of binary digits to the messages according to their probability of occurrence.
- This coding makes average number of binary digits per message nearly equal to entropy.

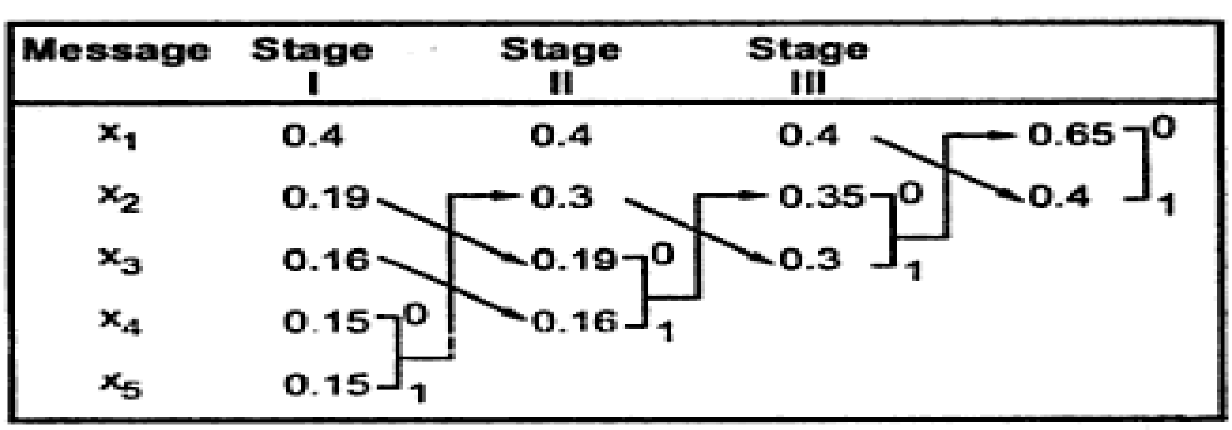
Coding steps:

1. The source symbols are listed in order of decreasing probabilities.
2. The two source symbols of lowest probabilities are assigned a 0 and a 1.
3. These source symbols are combined into a new source symbol with probability equal to the sum of the two original probabilities.
4. The probability of the new symbol is placed in the list in accordance with its value.
5. The procedure is repeated until we are left with a final list of source symbols of only two for which a 0 and a 1 are assigned.
6. Then the codeword are found by tracing the sequence of 0's and 1's assigned to that symbol as well as its successor.

Example 2:

A discrete memoryless source has five symbols x_1, x_2, x_3, x_4, x_5, with probabilities 0.4, 0.19, 0.16, 0.15 and 0.15 respectively attached to every symbol. Construct Huffman code for the source and calculate code efficiency.

Solution:



Message	Probability (p_k)	Codeword b_0 b_1 b_2	Number of digits n_k or (l_k)
x_1	0.4	1	1
x_2	0.19	000	3
x_3	0.16	001	3
x_4	0.15	010	3
x_5	0.15	011	3

- **Entropy:**

$H(I) = \sum_{k=0}^{k-1} pk \ \mathbf{log_2} \left(\frac{\mathbf{1}}{\mathbf{pk}}\right)$

$H = \sum_{k=0}^{k-1} P_K [\log_{10}(1/P_K)/\log 2]$

H= $0.4\log_2(1/0.4) + 0.19\log_2(1/0.19) + 0.16\log_2(1/0.16) + 0.15\log_2(1/0.15) + 0.15\log_2(1/0.15)$

H = 2.2281 bits/message

- **Code efficiency:**

$$\eta = \frac{H(\mathscr{S})}{\bar{L}}$$

- The average number of bits per message is given as $\bar{L}$

$$\bar{L} = \sum_{k=0}^{K-1} p_k l_k$$

$\bar{L}$ = **0.4(1)+0.19(3)+0.16(3)+0.15(3)+0.15(3)**

$\bar{L}$ **= 2.35**

- **Hence, code efficiency η = 2.2281 / 2.35 = 0.948**

Example 3:

A DMS have five symbols $s_0, s_1, \ldots s_4$, characterized by probability distribution as 0.4, 0.2, 0.1, 0.2 and 0.1 respectively. Evaluate two distinct variable length Huffman codes for the source to illustrate nonuniqueness of Huffman technique. Calculate the variance of the ensemble as defined by,

$$\sigma^2 = \sum_{k=0}^{M-1} p_k \left[l_k - l_{avg}\right]^2$$

where p_k and l_k are probability and length of codeword respectively for symbol s_k and l_{avg} is average length. Conclude on result.

Solution:

(I) Placing combined symbol as high as possible :

(i) To obtain codeword :

Table 1.10.12 lists the coding. The combined symbol is placed as high as possible.

Symbol	Probability Stage - I	Stage - II	Stage - III	Stage - IV
s_0	0.4	0.4	0.4	0.6 0
s_1	0.2	0.2	0.4 0	0.4 1
s_3	0.2	0.2 0	0.2 1	
s_2	0.1 0	0.2 1		
s_4	0.1 1			

As per the above table, the codes are listed below :

Symbol	Probability p_k (P_k)	Codeword $b_0\ b_1\ b_2$	No. of bits per symbol n_k or (l_k)
s_0	0.4	0 0	2
s_1	0.2	1 0	2
s_2	0.1	0 1 0	3
s_3	0.2	1 1	2
s_4	0.1	0 1 1	3

(ii) To obtain average codeword length :

Average codeword length can be calculated as,

$$\bar{N} = \sum_{k=0}^{4} p_k n_k$$

$$= 0.4\ (2) + 0.2\ (2) + 0.1\ (3) + 0.2\ (2) + 0.1\ (3)$$

$$= 2.2$$

(iii) To obtain variance of code :

Variance can be calculated as,

$$\sigma^2 = \sum_{k=0}^{4} p_k \left[n_k - \bar{N}\right]^2$$

$$= 0.4[2-2.2]^2 + 0.2[2-2.2]^2 + 0.1[3-2.2]^2$$

$$+ 0.2[2-2.2]^2 + 0.1[3-2.2]^2$$

$$= 0.16$$

(II) Placing combined symbol as low as possible :

(i) To obtain codewords :

Table 1.10.14 shows the listing of huffman coding algorithm. The combined symbol is placed as low as possible.

Symbol	Probability Stage - I	Stage - II	Stage - III	Stage - IV
s_0	0.4	0.4	0.4	0.6 0
s_1	0.2	0.2	0.4 0	0.4 1
s_3	0.2	0.2 0	0.2 1	
s_2	0.1 0	0.2 1		
s_4	0.1 1			

As per the above table, the codes are listed below :

Symbol	Probability p_k	Codeword $b_0\, b_1\, b_2$	No. of bits per symbol n_k
s_0	0.4	1	1
s_1	0.2	0 1	2
s_2	0.1	0 0 1 0	4
s_3	0.2	0 0 0	3
s_4	0.1	0 0 1 1	4

(ii) To obtain average codeword length :

Average codeword length is given as,

$$\bar{N} = \sum_{k=0}^{4} p_k\, n_k = 0.4\ (1) + 0.2\ (2) + 0.1\ (4) + 0.2\ (3) + 0.1\ (4)$$

$$= 2.2$$

(iii) To obtain variance of the code :

Variance can be calculated as,

$$\sigma^2 = \sum_{k=0}^{4} p_k \left[n_k - \bar{N}\right]^2$$

$$= 0.4[1-2.2]^2 + 0.2[2-2.2]^2 + 0.1[4-2.2]^2 + 0.2[3-2.2]^2 + 0.1[4-2.2]^2$$

$$= 1.36$$

Results :

Sr. No.	Method	Average length	Variance
1	As high as possible.	2.2	0.16
2	As low as possible.	2.2	1.36

<u>Example 4:</u>

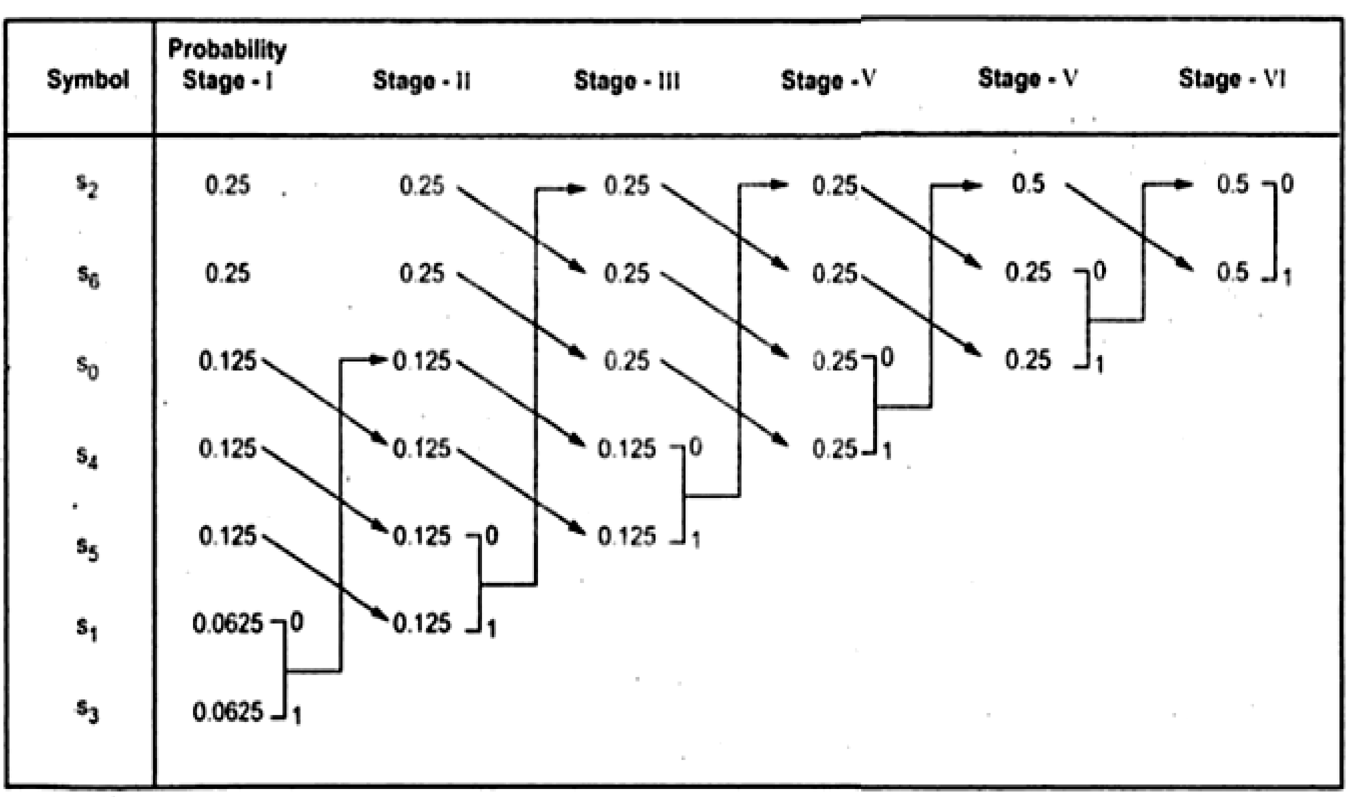
A DMS has following alphabet with probability of occurrence as shown below :

Symbol	s_0	s_1	s_2	s_3	s_4	s_5	s_6
Probability	0.125	0.0625	0.25	0.0625	0.125	0.125	0.25

Generate the Huffman code with minimum code variance. Determine the code variance and code efficiency. Comment on code efficiency.

Solution : (i) To obtain codewords :

<u>Based on the above table, the codes are traced and generated as follows :</u>



Symbol	Probability	Digits obtained by tracing	Codeword	No. of bits per symbol n_k
s_0	0.125	1 0 0	0 0 1	3
s_1	0.0625	0 0 0 0	0 0 0 0	4
s_2	0.25	0 1	1 0	2
s_3	0.0625	1 0 0 0	0 0 0 1	4
s_4	0.125	0 1 0	0 1 0	3
s_5	0.125	1 1 0	0 1 1	3
s_6	0.25	1 1	1 1	2

(ii) To obtain average codeword length :

Average codeword length is given as,

$$\overline{N} = \sum_{k=0}^{6} p_k n_k$$

$$= 0.125\,(3) + 0.0625\,(4) + 0.25\,(2) + 0.0625\,(4) + 0.125\,(3) + 0.125\,(3) + 0.25\,(2)$$

$$= 2.625 \text{ bits/symbol.}$$

(iii) To obtain entropy of the source :

Entropy is given as,

$$H = \sum_{k=0}^{6} p_k \log_2\left(\frac{1}{p_k}\right)$$

$$= 0.125\log_2\left(\frac{1}{0.125}\right) + 0.0625\log_2\left(\frac{1}{0.0625}\right) + 0.25\log_2\left(\frac{1}{0.25}\right) + 0.0625\log_2\left(\frac{1}{0.0625}\right) + 0.125\log_2\left(\frac{1}{0.125}\right) + 0.125\log_2\left(\frac{1}{0.125}\right) + 0.25\log_2\left(\frac{1}{0.25}\right)$$

$$= 2.625 \text{ bits/symbol.}$$

(iv) To obtain code efficiency :

Code efficiency is given as,

$$\eta = \frac{H}{\overline{N}}$$

$$= \frac{2.625}{2.625} = 1 \text{ or } 100\ \%$$

Here efficiency of the code is 100% since the combined probabilities are equal in stage IV and stage VI.

(v) To obtain variance :

$$\sigma^2 = \sum_{k=0}^{6} p_k \left(n_k - \overline{N}\right)^2$$

$$= 0.125(3-2.625)^2 + 0.0625(4-2.625)^2 + 0.25(2-2.625)^2$$

$$+0.0625(4-2.625)^2 + 0.125(3-2.625)^2$$

$$+0.125(3-2.625)^2 + 0.25(2-2.625)^2$$

$$= 0.4843$$

NOISELESS CODING THEOREM:

- The source coding theorem is also called as noiseless coding theorem, noiseless in the sense that it establishes the condition for error –free encoding to be possible.
- A problem related to source coding is that of data compression where we use a “rate” R bits per source symbol that is less than the source entropy H(I).
- For such rate we cannot find error-free coding.

CHANNEL CAPACITY THEOREM (OR) SHANNON’S SECOND THEOREM:

Shannon’s theorem says that it is possible to transmit information with an arbitrarily small probability of error provided that the information rate R is less than or equal to rate C called the channel capacity.

Theorem:

Given a source of M equally likely messages, with M>>1, which is generating information at a rate R. Given a channel with channel capacity C .then if

$R \leq C$

There exists a coding technique such that the output of the source may be transmitted over the channel with a probability of error in the received message which may be made arbitrarily small.

SHANNON HARTLEY THEOREM FOR GAUSSIAN CHANNEL:

The channel capacity of a white bandlimited gaussian channel is,

$$\boxed{C = B\log_2\left(1+\frac{S}{N}\right) \text{ bits / } sec.}$$

Here B is the channel bandwidth,

S is the signal power,

and N is the total noise power within the channel bandwidth.

We know that signal power is given as,

$$\text{Power } P = \int_{-B}^{B} \text{Power spectral density}$$

Here B is bandwidth. And power spectral density of white noise is $\frac{N_0}{2}$. Hence noise power N becomes,

$$\text{Noise power } N = \int_{-B}^{B} \frac{N_0}{2}\, df$$

$$\therefore \quad N = N_0 B$$

BANDWIDTH AND SIGNAL TO NOISE RATIO TRADE OFF:

1. The capacity of the Gaussian channel is given as ,

$$\boxed{C = B \log_2 \left(1 + \frac{S}{N}\right) \text{ bits / sec.}}$$

2. The channel capacity depends on two factors,
   - Bandwidth **(B)** and
   - Signal to noise ratio $\frac{S}{N}$.
3. **Noiseless channel has infinite capacity.**
   - If there is no noise in the channel then **N=0.** Hence, $\frac{S}{N} = \infty$.
   - The capacity of the channel will be $C = B \log_2(1+\infty) = \infty$
4. **Infinite bandwidth channel has limited capacity**
   - If bandwidth ‘B’ is infinite, the channel capacity is limited.
   - If bandwidth increases ,noise power increases since $N=N_0B$.
   - Due to the increase in noise power the signal to noise ratio decreases.
   - As $B \rightarrow \infty$,capacity approaches an upper limit.
   - The channel capacity is given by

     $C_\infty = lt_{B \rightarrow \infty} C = 1.44 \, S/N_0$

RATE/BANDWIDTH AND SIGNAL TO NOISE RATIO, E_B/N_O TRADE OFF (Implication of Information Capacity Theorem):

- let the system is transmitting at the rate R_b equal to the channel capacity C .
- then average transmitted signal power will be

$S= E_b C$ --------------- (1)

Where,

E_b → transmitted energy per bit .
C → average signal power.

- The capacity of the continuous channel is given by

$$\boxed{C = B \log_2\left(1+\frac{S}{N}\right) \text{ bits / sec.}}$$ ------------------------(2)

Sub, $S = E_b C$ and $N=N_oB$ in eqn (2)

$$C = B \log_2\left(1+\frac{E_b C}{N_0 B}\right)$$

i.e. $$\frac{C}{B} = \log_2\left(1+\frac{E_b}{N_0}\frac{C}{B}\right)$$

$$\therefore \quad \frac{C}{B} = \frac{\log_e\left(1+\frac{E_b}{N_0}\frac{C}{B}\right)}{\log_e 2}$$

$$\therefore \quad \frac{C}{B}\log_e 2 = \log_e\left(1+\frac{E_b}{N_0}\frac{C}{B}\right)$$

$$\frac{E_b}{N_0} = \frac{2^{\left(\frac{C}{B}\right)}-1}{(C/B)}$$ ---------------(3)

E_b/N_0 → energy per bit to noise spectral density ratio.

C/B → bandwidth efficiency.

If C= Rb then eqn (3) becomes

$$\frac{E_b}{N_0} = \frac{2^{(R_b/B)}-1}{(R_b/B)}$$ ---------- (4)

Rb/B → rate bandwidth

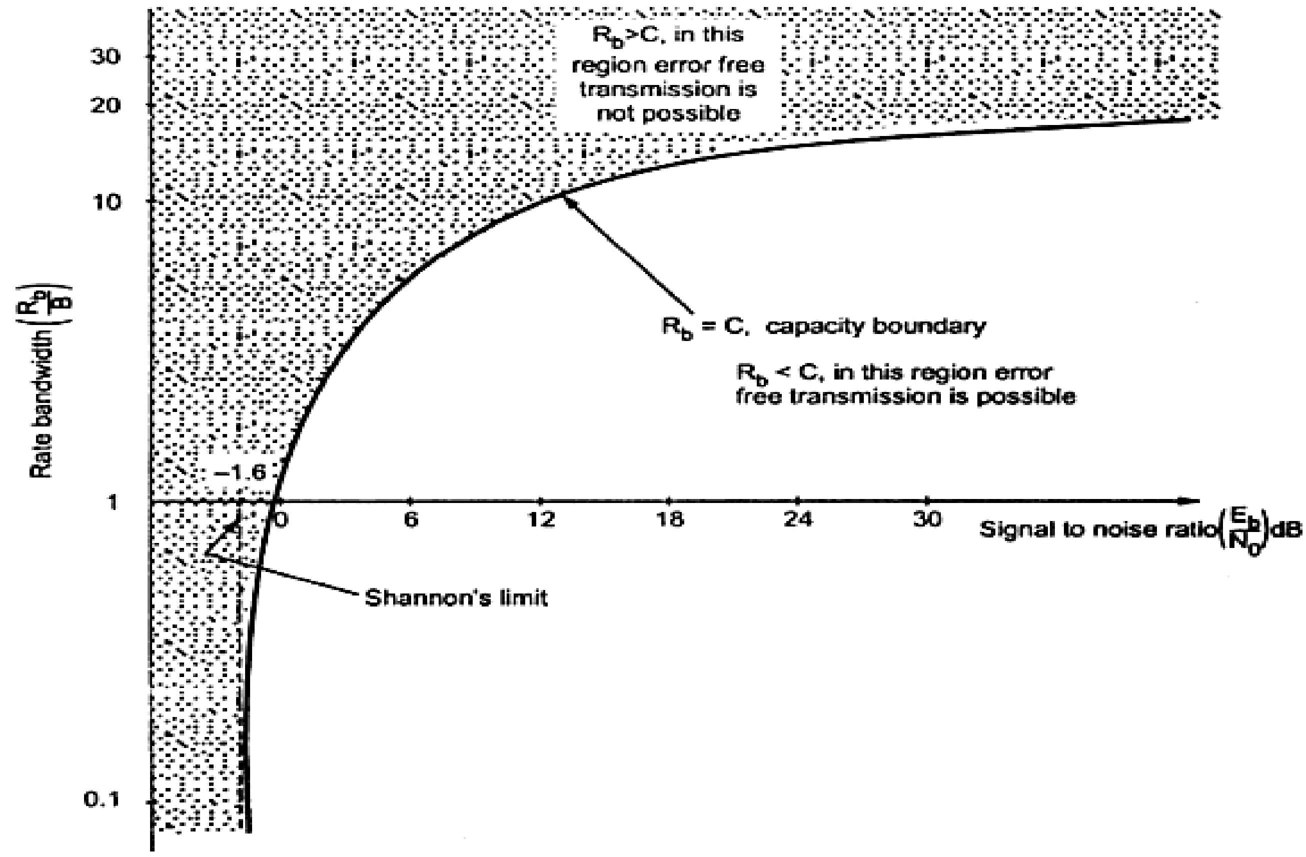


Fig. 4: Rate/bandwidth versus SNR

The following observations can be made from the above figure

i). for infinite bandwidth

$$\mathbf{C_\infty = 1.44\ S/No}$$

Here, S = E_b C and N=NoB then the above eqn.(1) becomes,

$$\mathbf{C_\infty = 1.44 \left(\frac{Eb\ C}{NoB}\right)}$$

$$\left(\frac{Eb}{No}\right)=\left(\frac{1}{1.44}\right) = 0.693$$

$$\left(\frac{Eb}{No}\right)dB = 10 \log \left(\frac{Eb}{No}\right)$$

$$= 10 \log (0.693) = -1.6\ dB$$

This value of $\left(\frac{Eb}{No}\right)$ is called shannon's limit.

And the capacity at shannon's limit

$$\mathbf{C_\infty = 1.44 \left(\frac{Eb\ C}{NoB}\right)}$$

ii). The plot has two regions

- ✓ For the region R_b<C error free transmission is possible
- ✓ For the region R_b>C error free transmission is not possible
- ✓ The curve R_b=C is called capacity boundary.

iii). the probability of error for an ideal system is given as

$$P_e = \begin{cases} 1 & for \quad R_b \geq C \\ 0 & for \quad R_b < C \end{cases}$$

UNIT - IV
MULTIPLE ACCESS TECHNIQUES

SS&MA techniques: FDMA, TDMA, CDMA, SDMA application in wire and wireless communication: Advantages (merits)

Spread Spectrum Modulation:

- Spread spectrum modulation is a special type of modulation that offers secure communication.
- It prevents the unauthorized receivers to receive or to detect the transmitted message.
- It also resists external interference.
- Spread spectrum modulation solves two types of problem
  - ✓ The unintentional interference caused when other user transmitting the message through the same channel.
  - ✓ The interference created by the hostile transmitter to jam the transmission.

Definition

- Spread spectrum is a means of transmission in which the data sequence occupies a Bandwidth in excess of the minimum Bandwidth required to send it.
- The Spectrum spreading is accomplished before transmission through the use of a code that is independent of the data sequence. The same code is used in the receiver to de-spread the received signal so that the original data sequence may be recovered.

Applications

- Secure communication
- Interference rejection
- Multipath rejection
- multiple access

Classification of Spread Spectrum:

- Direct Sequence Spread Spectrum (DSSS)
- Frequency Hop Spread Spectrum (FHSS)
- Time Hoping Spread Spectrum (TH- SS)
- Hybrid Spread Spectrum (HSS)

Direct Sequence Spread Spectrum (DSSS)

- First incoming data sequence is modulated using a wide band PN code. This code transforms a narrow band data sequence into a noiselike wide band signal.
- [A pseudo-noise (PN) sequence is a periodic binary sequence with a noiselike waveform that is usually generated by means of a feedback shift-register.]
- Then this signal undergoes second modulation using PSK.

Frequency Hop Spread Spectrum (FHSS)

Spectrum of a data modulated carrier is widened by changing the carrier frequency in a pseudo random manner.

<u>Direct-Sequence Spread Spectrum with Coherent BPSK:</u>

- First incoming data sequence is modulated using a wide band PN code. This code transforms a narrow band data sequence into a noise like wide band signal.
- Then this signal undergoes second modulation using PSK.

<u>Direct-Sequence BPSK Transmitter:</u>

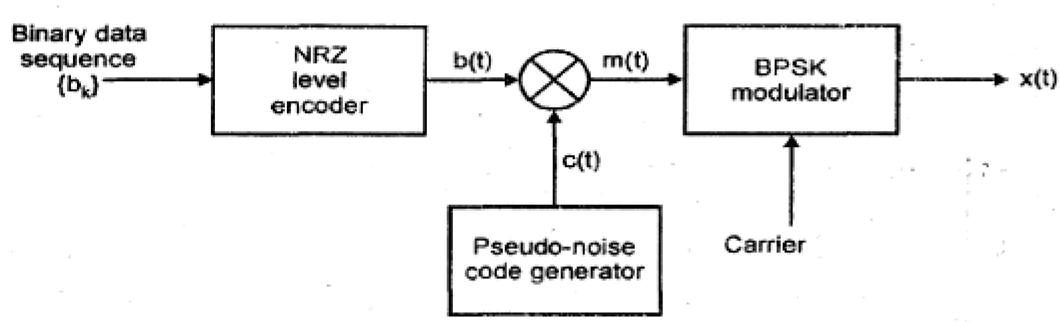


Fig. 1: Direct-Sequence Spread Spectrum with Coherent BPSK Transmitter

- First, the binary data sequence b_k is given to Non-Return to zero level encoder.
- This encoder converts b_k into polar NRZ waveform b(t), which is followed by 2 stages of modulation.
- The first stage consists of a product modulator (or) multiplier.
- The second stage consists of BPSK modulator.
- The Pseudo-noise sequence generator generates the PN sequence.
- The product modulator multiplies the two signals b(t) and c(t). The output of multiplier is direct-sequence spread signal m(t).
- This output signal m(t) is given as modulating signal to Binary PSK Transmitter.
- The direct sequence BPSK (DS-BPSK) is generated at the Output x(t).
- The phase modulation $\theta(t)$ of x(t) has one of two values 0 and π, depending on the polarities of the message signal b(t) and PN signal c(t) at time t in accordance with the truth table.

<u>Direct-Sequence BPSK waveforms</u>

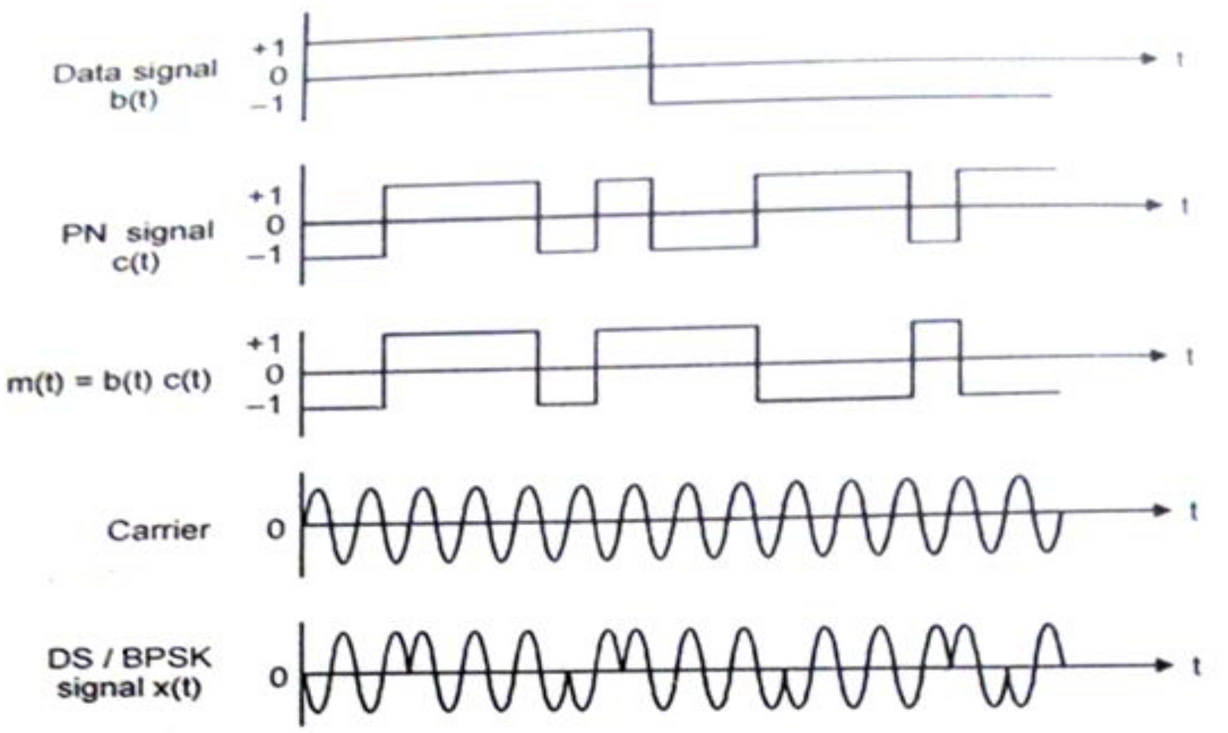


Fig. 2: Direct-Sequence BPSK waveforms

Truth Table for phase modulations:

		Polarity of data sequence b(t) at time (t)	
		+	-
Polarity of PN sequence c(t) at time t	+	**0**	**π**
	-	**π**	**0**

When, m (t)→ +ve; phase shift is 0°.
m (t) → -ve; phase shift is180°.

Direct-Sequence BPSK Receiver:

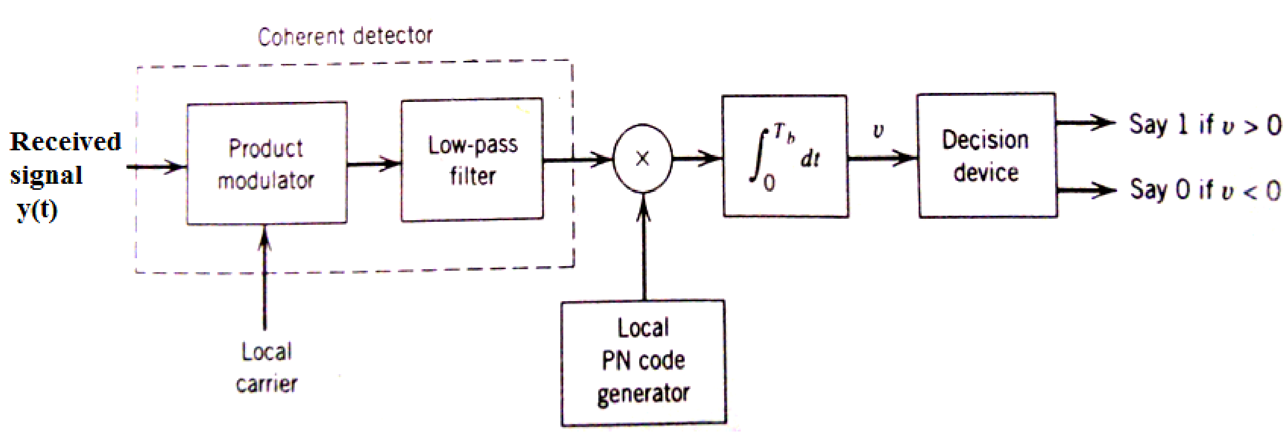


Fig. 3: Direct-Sequence Spread Spectrum with Coherent BPSK Transmitter

- Receiver consists of 2 stages of Demodulation.
- In the first stage, the received signal y(t) and a locally generated carrier are applied to a product modulator followed by a low-pass filter whose bandwidth is equal to that of the original message signal m(t).
- This stage of demodulation process reverses the phase shift keying applied to the transmitted signal.
- The second stage of demodulation performs spectrum de-spreading by multiplying the low-pass filter output by a locally generated replica of the PN code c(t).
- The Integrator integrates the product of detected message signal and pseudo noise signal over one bit interval $0 \le t \le T_b$.
- Finally decision is made by the receiver:
    - ➔ If **v >zero**, binary **symbol 1** was sent in the interval $0 \le t \le T_b$.
    - ➔ If **v <zero**, binary **symbol 0** was sent.
    - ➔ If **v =0**, the receiver makes a random guess in favor of 1 or 0.

<u>Advantages of DSSS:</u>

- Interference by multipath rejection is minimized.
- Performance of DSSS in presence of noise is better compared with other systems.

<u>Disadvantages:</u>

- Since Acquisition time is too large DSSS systems are very slow.
- Channel Bandwidth required is high.

<u>Frequency-Hop Spread Spectrum:</u>

- The type of spread spectrum in which the carrier hops randomly from one frequency to another is called frequency –hop spread spectrum.
- Frequency hopping means to transmit the data bits in different frequency slots.
- The total bandwidth of the output signal is equal to the sum of all these frequency slots or hops.
- Since the carrier changes from one frequency to another it is difficult for the jammer to cover the entire bandwidth to trap the signal.
- The modulation format used is M-ary frequency shift keying(MFSK)
- Based on the rate at which hops occurs it is classified into two types.
    1. Slow-frequency hopping, 2. Fast -frequency hopping.

<u>Slow frequency-Hopping:</u>

- Several symbols are transmitted on each frequency hop.

- Slow-frequency hopping, in which the symbol rate R_s of the MFSK signal is an integer multiple of the hop rate R_h.

Fast Frequency-Hopping:

- The carrier frequency changes several times during the transmission of one symbol.
- Fast -frequency hopping, in which the hop rate R_h is an integer multiple of the MFSK symbol rate R_s.

Slow Frequency-Hopping:

Frequency-hopping MFSK Transmitter:

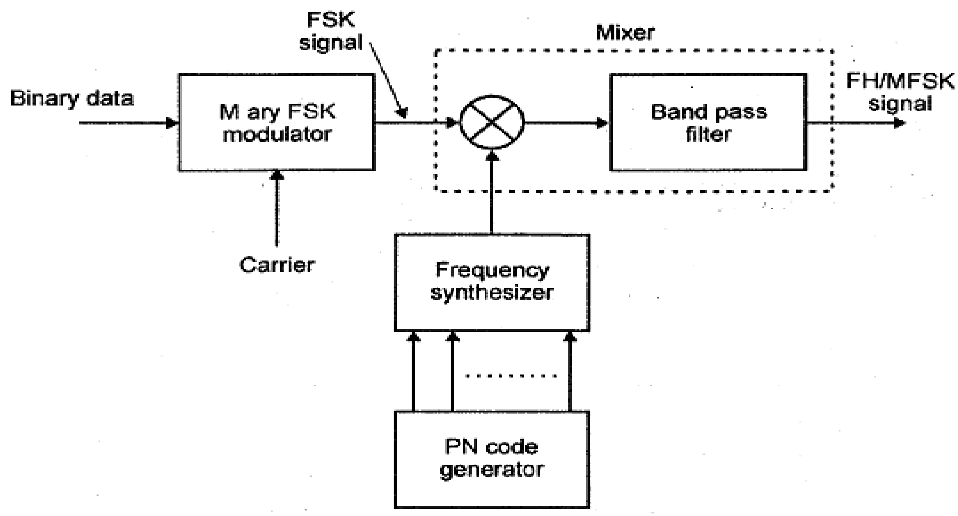


Fig. 4: Frequency-Hop spread M-ary frequency-shift keying Transmitter

- The Binary data is applied to the M-ary FSK Modulator.
- The output of FSK modulator is then applied to the mixer.
- A particular Frequency from the frequency synthesizer is also applied to the mixer.
- The mixer, which consists of a multiplier followed by a Band pass filter.
- The filter is designed to select the sum of frequency component results from the multiplication process of the transmitted signal.
- The output of mixer is FH/MFSK signal with large Bandwidth.
- For k-bits of PN sequence the carrier hops over 2^k distinct values

Frequency-Hop MFSK Receiver:

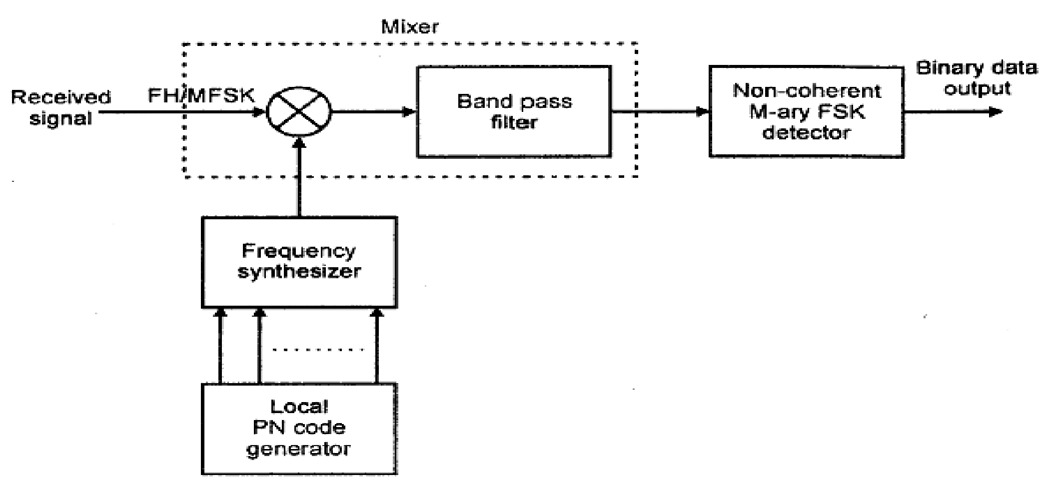


Fig. 5: Frequency-Hop spread M-ary frequency-shift keying Receiver

- The frequency hopping (FH) is first removed by mixing the received signal with the o/p of a frequency synthesizer.
- The mixer o/p is then band pass filtered and it is subsequently processed by a non-coherent MFSK Detector.
- The M-ary Detector is constructed using a M-non coherent matched filters, each of which is matched to any one of the MFSK tones .
- The original symbol is obtained by selecting the largest filter o/p.

- An individual FH-MFSK tone of shortest duration is referred as a "chip".
- The chip rate is defined as

$$R_c = \max(R_h, R_s)$$

Where, R_h = hop rate

R_s = Symbol rate

- In slow FH-MFSK, multiple symbols are transmitted per hop. Each symbol is a chip.

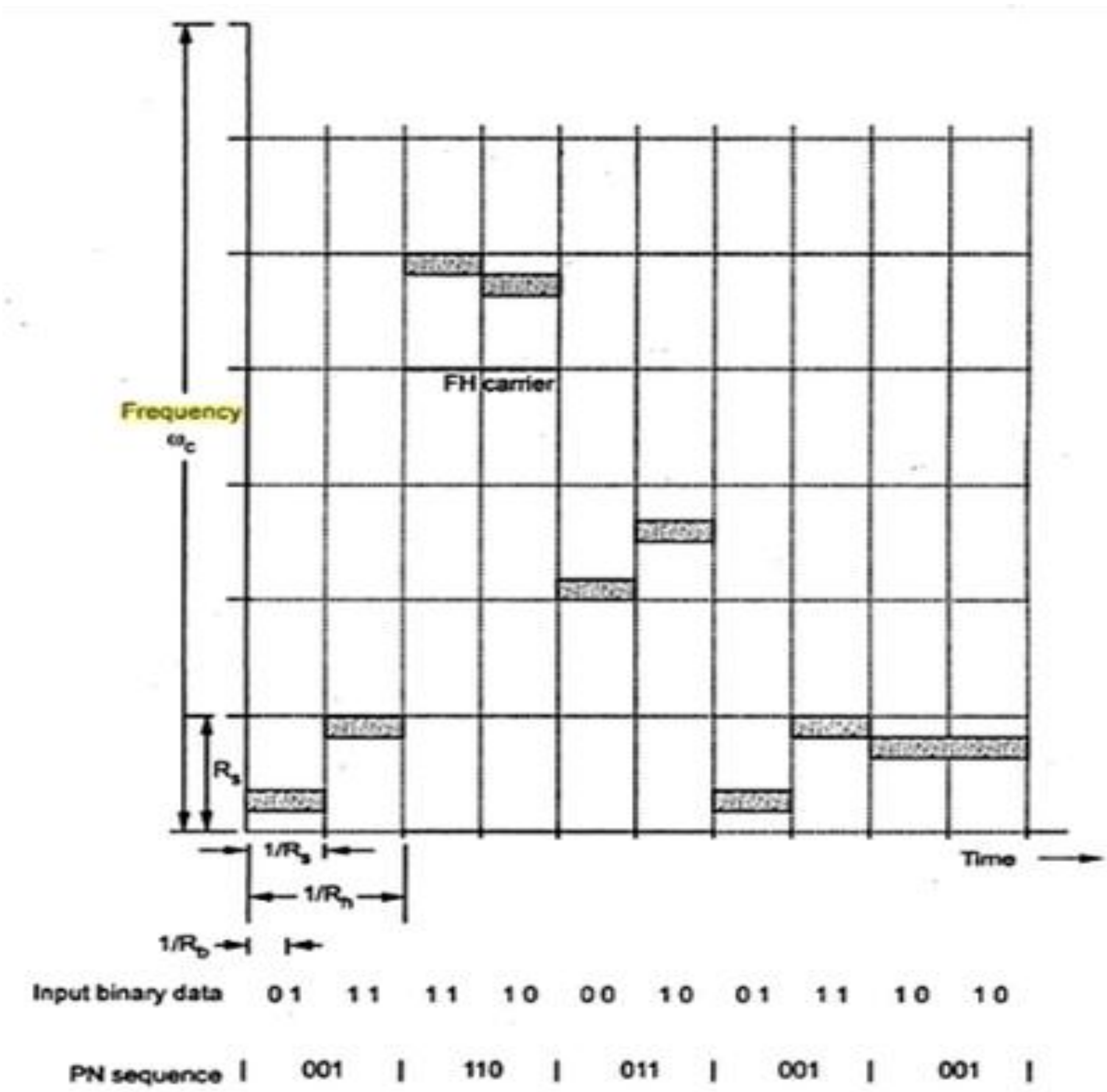


Fig. 5.1: Slow Frequency Hopping

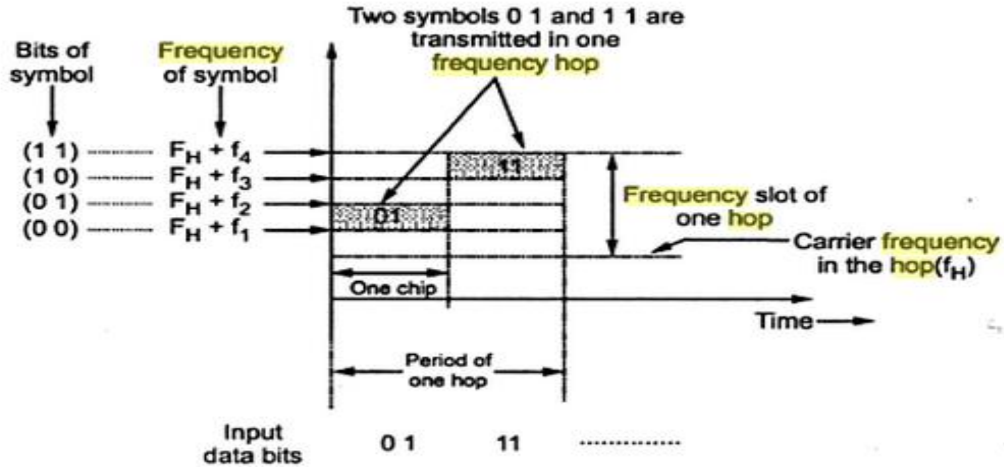


Fig. 5.2: illustration of slow hopping

<u>Fast Frequency-Hopping:</u>

- When several frequencies hops take place to transmit one symbol it is called fast frequency-hopping.
- Hop rate (Rh) is higher than the symbol rate (Rs).
- Advantage of fast hopping is that before the jammer tries to complete the reception of one symbol, the carrier frequency is changed.

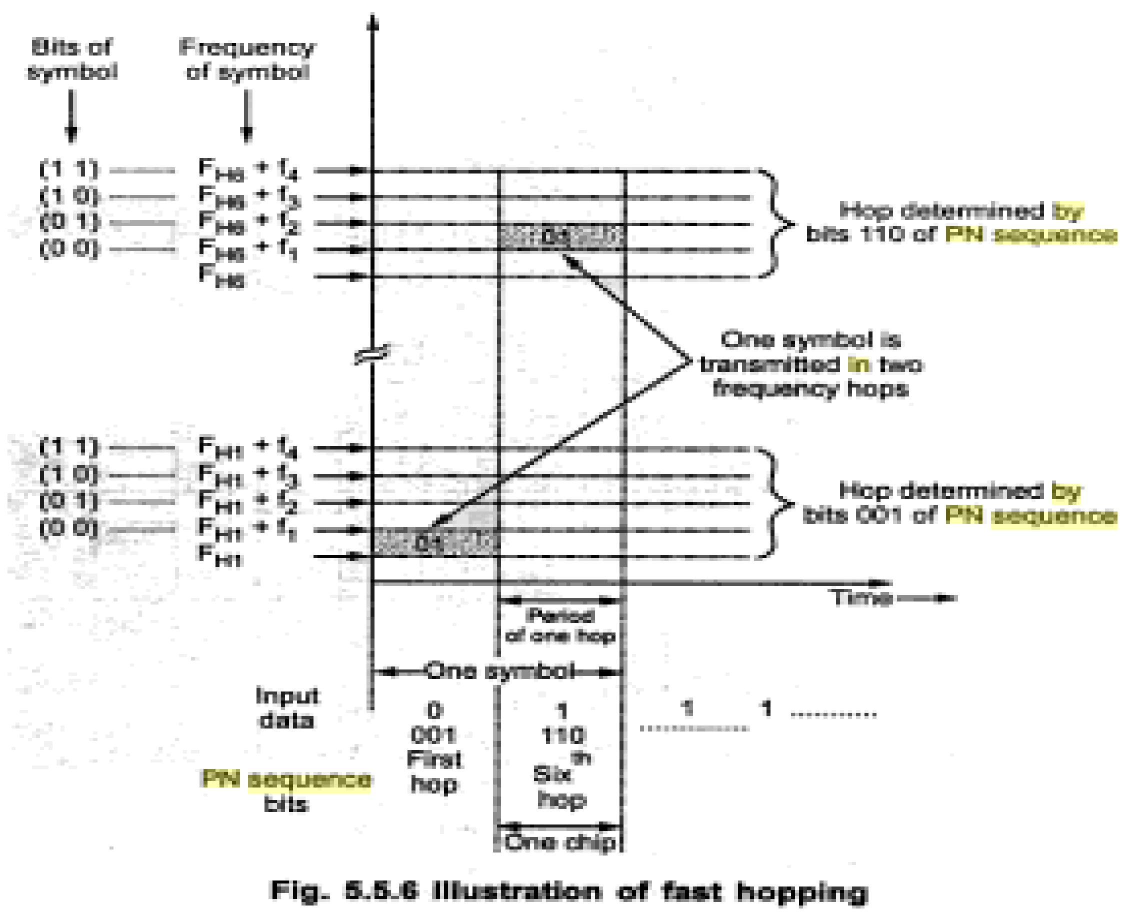
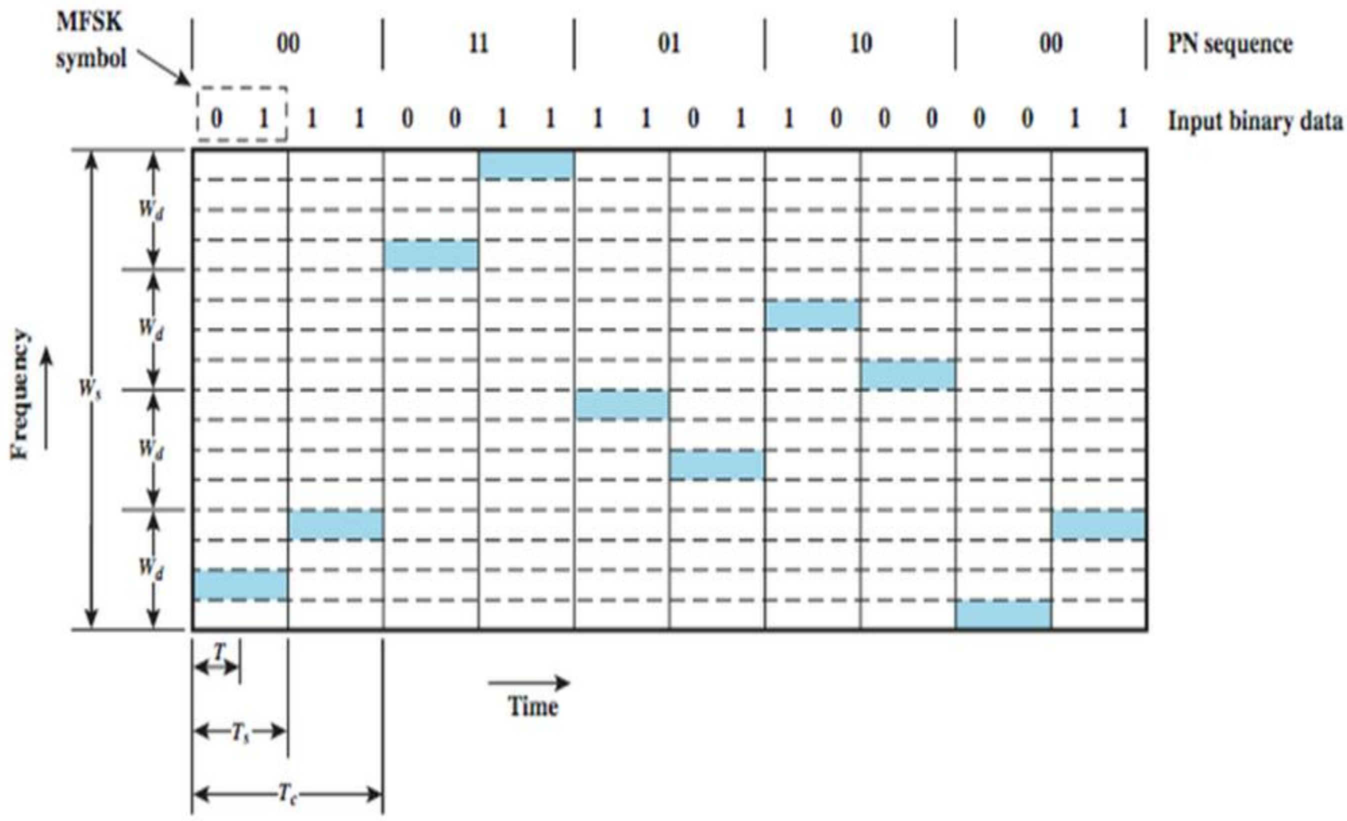


Fig. 5.3: illustration of fast hopping



- W_d - is channel bandwidth
- $W_s = 4W_d$ (total bandwidth)

Fig. 5.4: illustration of slow hopping

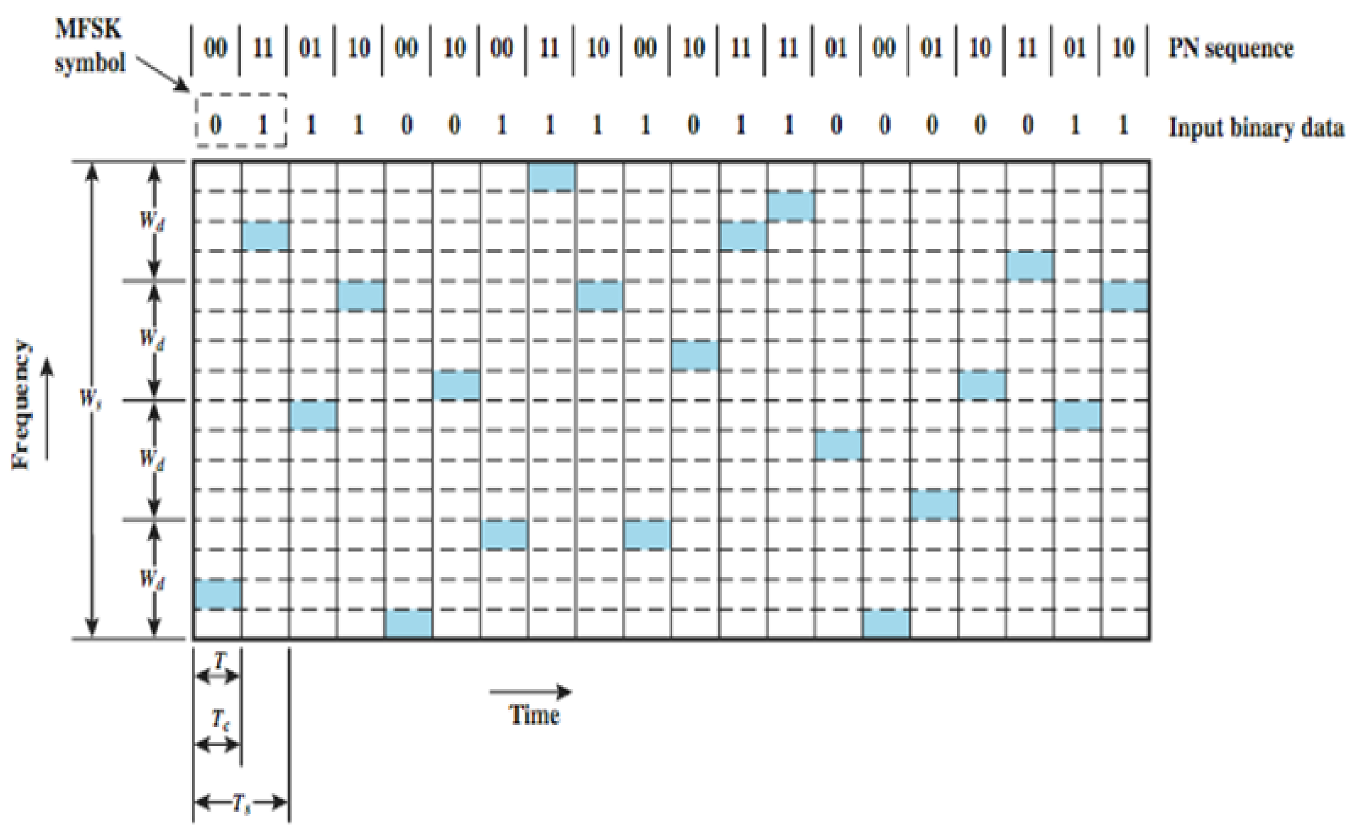


Fig. 5.4: illustration of fast hopping

<u>Chip rate for fast hopping</u>

- Since the hop rate is higher
- the chip rate is equal to hop rate

Chip rate, $R_c = R_h$.

1. The first two bits 01 of the input data form one symbol.

   Two hops are used to transmit one symbol.

2. The frequency of the FSK signal for symbol 01 is f_2.
3. This symbol is transmitted in first hop (fh_1+f_2) and in other hop (fh_6+f_2).One chip is equal to one hop.

Multiple Access Techniques:

- Multiple access schemes are used to allow many mobile users to share simultaneously a finite amount of radio spectrum.
- The sharing of spectrum is required to achieve high capacity by simultaneously allocating the available bandwidth (or the available amount of channels) to multiple users.
- In wireless communications systems, it is often desirable to allow the subscriber to send simultaneously information to the base station while receiving information from the base station, this effect is called duplexing.
- For example, in conventional telephone systems, it is possible to talk and listen simultaneously, and this effect, called duplexing.
- Duplexing may be done using frequency or time domain techniques.
- Frequency Division Duplexing (FDD) provides two distinct bands of frequencies for every user.
    1. The forward band provides traffic from base station to mobile.
    2. The reverse band provides traffic from mobile to base station.
- Time Division Multiplexing (TDD) uses time instead of frequency to provide both a forward and reverse link. If the time split between the forward and reverse time slot is small, then the transmission and reception of data appears simultaneous to the user.
- The multiple access techniques used in wireless communication are

i. Frequency Division Multiple Access (**FDMA)**

ii. Time Division Multiple Access **(TDMA)**

iii. Spread Spectrum Multiple Access **(SSMA)** : an example is Code division multiple access **(CDMA)**

iv. Space division multiple access **(SDMA)**

FREQUENCY DIVISION MULTIPLE ACCESS(FDMA);

- FDMA is a technique in which each individual user is assigned a unique frequency slot or channel.
- These channels are assigned on demand to users who request service.
- During the period of call, no other users can share the same channel.

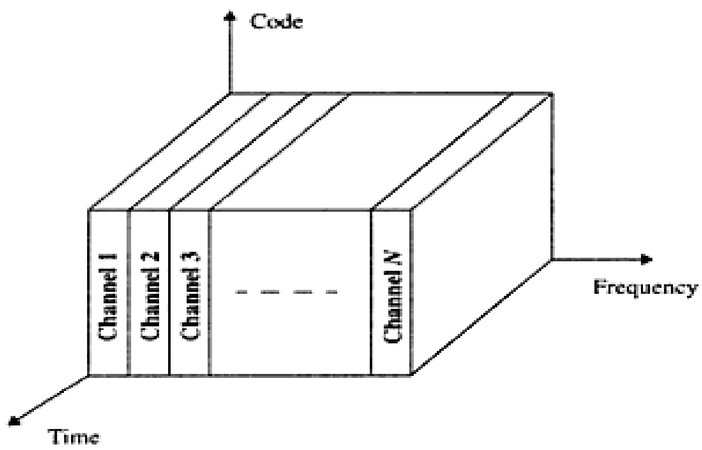


Fig. 6: FDMA scheme (where different channels are assigned different frequencies)

Features of FDMA:

1. The FDMA channel carries only one phone circuit at a time.
2. If an FDMA channel is not in use, it remains idle and it cannot be used or shared by other users. It is essentially a wasted resource.
3. After the assignment of voice channel, the base station and the mobile transmit simultaneously and continuously.
4. The bandwidths of FDMA channel are relatively narrow (30 KHz) as each channel supports only one phone circuit per carrier. Thus FDMA is implemented in narrowband systems.
5. The symbol time of FDMA signal is large compared to the average delay spread. This implies that amount of intersymbol interference is low and thus no equalization is needed in FDMA system
6. The complexity of FDMA systems is lower.
7. Fewer bits are needed for framing and synchronization in FDMA.
8. FDMA system have higher cell site system costs as compared to TDMA .
9. The FDMA mobile unit uses duplexer since both the transmitter and receiver operates at the same time. this results in an increase in the cost of FDMA subscriber and mobile units.
10. FDMA requires tight RF filtering to remove adjacent channel interference.
11. The number of channels in a FDMA system is given by

$$\mathbf{N=(B_t - 2B_{guard}) / B_c}$$

Where,

B_t →total spectrum allocation, B_{guard}→Guard Band, B_C→ Channel Bandwidth

Advantages:

1. The users can transmit continuously without any interruption.
2. The channel bandwidth is utilized more efficiently.
3. No synchronization is required.

Disadvantages:

1. Extra guard bands are required to avoid interchannel interference.
2. There is the possibility of intermodulation distortion at the transponder.
3. Power efficiency is reduced.

Non-linear effects in FDMA:

1. In a FDMA system, many channels share the same antenna at the base station.
2. The power amplifiers are non-linear. These non-linearities cause signal spreading and generate intermediate frequencies.(IF).
3. Undesired IF frequencies interfere with other channels causes adjacent channel interference.

Applications:

1. FDMA is used for wire-line channels for voice and data transmission.
2. Telephone communication.

Time Division Multiple Access (TDMA):

- In time division multiple accesses, the entire channel bandwidth is divided into number of individual time slots.
- Each user is allotted a particular time slot.
- The user can transmit and receive the message within the time.
- The other user has to wait for their particular time slot.
- TDMA transmits information in the form of a frame structure.

Frame Structure:

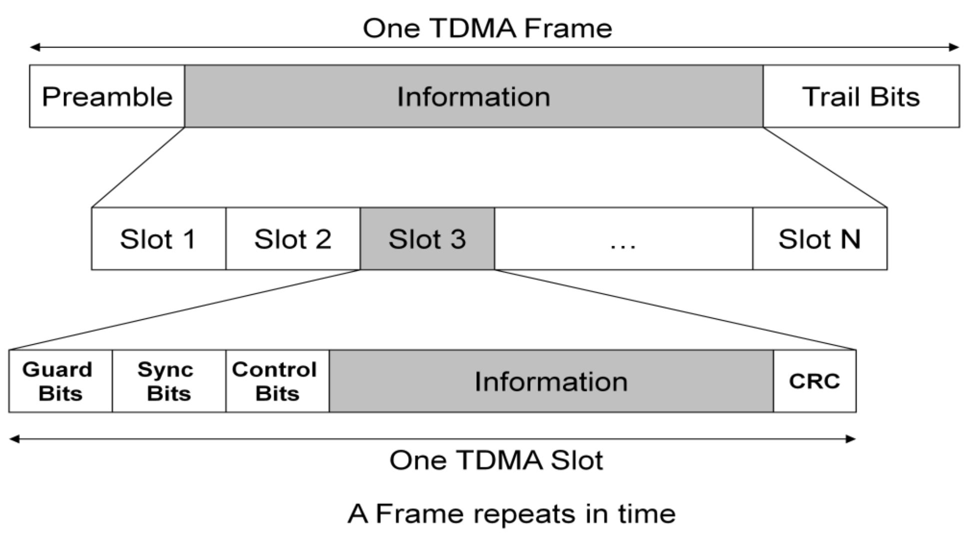


Fig.7: TDMA Frame Structure

1. Each frame is made of a preamble, an information message and tail bits.
2. In TDMA/TDD ,half of the time slots in the frame information message would be used for the forward link channels and half would be used for reverse link channels.
3. In TDMA/FDD ,similar frame structure is used for forward and reverse link channels but the carrier frequencies are differed.
4. In a TDMA frame ,the pre-amble contains the address and synchronization information that both the base station and the subscribers use to identify each other.
5. Guard times are used to allow synchronization of the receivers between different slots and frames.

Features of TDMA:

1. In TDMA multiple users shares a single carrier frequency on a time basis. The number of time slots depends on modulation technique and channel bandwidth.
2. Data transmission in TDMA system is not continuous, but occurs in bursts.
   This results in low battery consumption.
3. Because of the discontinuous data transmission, the handoff process is much simpler.
4. TDMA uses different time slots for transmission and reception , thus duplexes are not required.
5. Transmission rates are generally very high as compared to FDMA.Therefore Adaptive Equalization Is Necessary In TDMA System.
6. In TDMA guard time should be minimized.
7. High synchronization is required because of a burst transmission.
   The number of channels in a TDMA system is given by

$$\mathbf{N= m(B_{total} - 2}\ B_{guard})\ /\ \mathbf{B_c}$$

where,

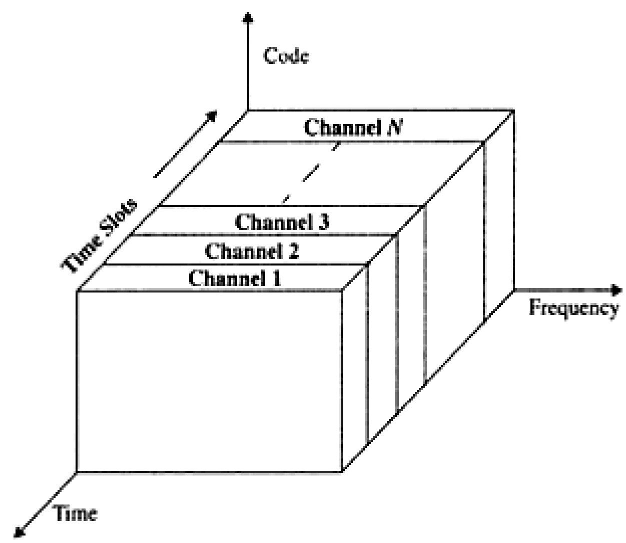
m→ maximum number of TDMA users

B_t →

$Bt \rightarrow total\ spectrum\ allocation;$

Bguard→ guard band ;Bc→channel bandwidth;

Fig. 8: TDMA scheme (where each channel occupies a cyclically repeating time slot)



8. The efficiency of the TDMA system is a measure of the percentage of transmitting data that contains information compared to overhead.
9. The number of overhead bits /frame is

$$\mathbf{b_{oh}= N_r b_r\ +\ N_t b_p + N_t b_g + N_r b_g}$$

N_r→ no of reference burst per frame

N_t→ no of traffic burst per frame.

b_r→ number of overhead bits per reference burst.

b_p→ number of overhead bits per preamble in each slot.

T_r→frame duration; R= channel bit rate.

- **Total no. of bits per frame = $b_T = T_f R$**

  %η = (1- boh/bt)* 100

- No of TDMA channels is given by **N= m(Btot-2Bguard)/Bc**

Advantages:

1. The user gets full bandwidth of the channel in a particular time slot.
2. For bursty signals like voice or speech TDMA gives maximum utilization is possible.
3. Most suitable technique for digital transmission.

Disadvantages:

1. Extra guard times are necessary
2. Synchronization is necessary.
3. It is not much suitable for continuous signals.

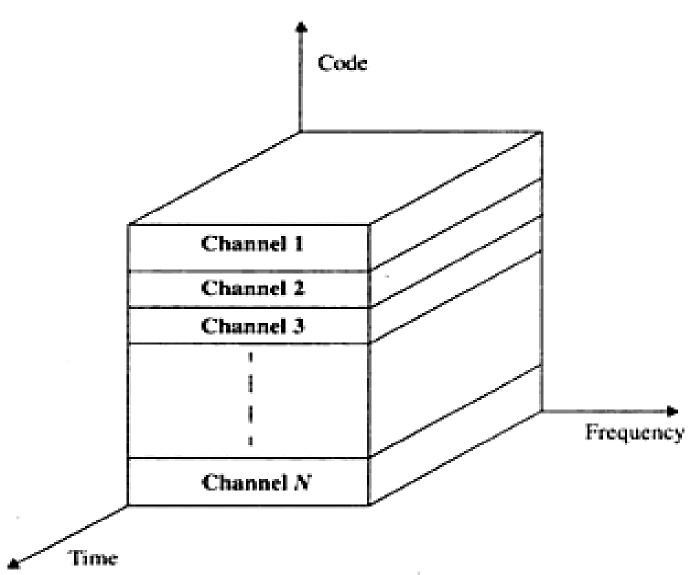
Applications:

1. It is used for voice and data transmission.
2. Bursty signals can be transmitted using TDMA.

Code Division Multiple Access (CDMA):

- ✓ **CDMA is a technique in which each individual user is assigned a unique codeword.**
- ✓ In CDMA systems the narrow band message signal is multiplied by a very large bandwidth signal called the spreading signal.
- ✓ All users in CDMA system use the same carrier frequency and transmit simultaneously.
- ✓ Each user has a code word that is orthogonal to all other code words.
- ✓ The receiver performs a time correlation operation to detect only the specific codeword. All other codeword appears as noise due to decoration.

Fig.9: CDMA scheme



Features of CDMA:

1. Many users of a CDMA system share the same frequency. Either TDD/FDD is used.
2. CDMA has a soft capacity limit. Increasing the number of users raises the noise problem. hence the system performance is degraded.
3. Multipath fading is reduced because the signal is spread over a large spectrum.
4. Channel data rates are very high.
   - Therefore the symbol duration is very short and usually much less than the channel delay spread.
   - A RAKE receiver can be used to improve the reception by collecting the time delayed versions of the required signals.
5. Since CDMA uses co-channel cells, it can use macroscopic spatial diversity to provide soft handoff.
6. Self jamming is a problem in CDMA.

- Self jamming arises from the fact that the spreading sequences of different users are not exactly orthogonal, hence in despreading the receiver responds to the other users.

7. The near-far problem exists in CDMA.
    - It occurs when many mobile users share the same channel.
    - The strongest received mobile signal will capture the demodulator at a base station. Therefore the weak signals are rejected and the noise will be increased.

CDMA System Block Diagram:

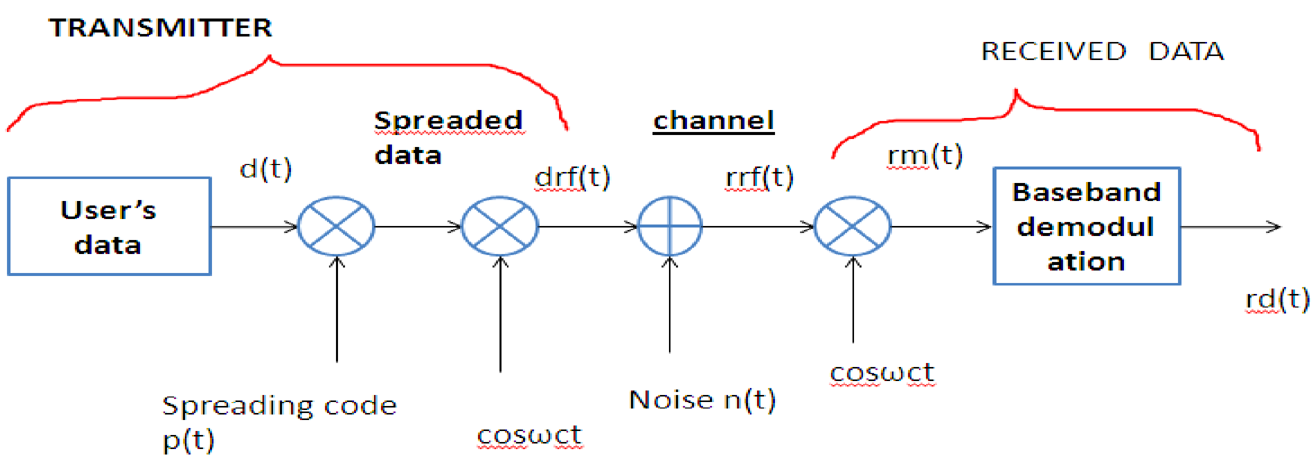


Fig. 10: CDMA system block diagram

1. The unmodulated user's data can be either a serial binary bit or multilevel data stream.
2. This data stream is then modulated by a code sequence p(t) to increase the bandwidth of the signal.
3. This spread spectrum signal $d_m(t)$is finally modulated with the carrier $\cos\omega_c t$ and transmitted through the channel.
4. During the transmission the signal gets corrupted by the addition of multipath interference ,random noise n(t).
5. The received signal is then demodulated to obtain the baseband signal.
6. The final baseband demodulation de-spreads and recovers the original data.

CDMA Receiver:

RAKE Receiver:

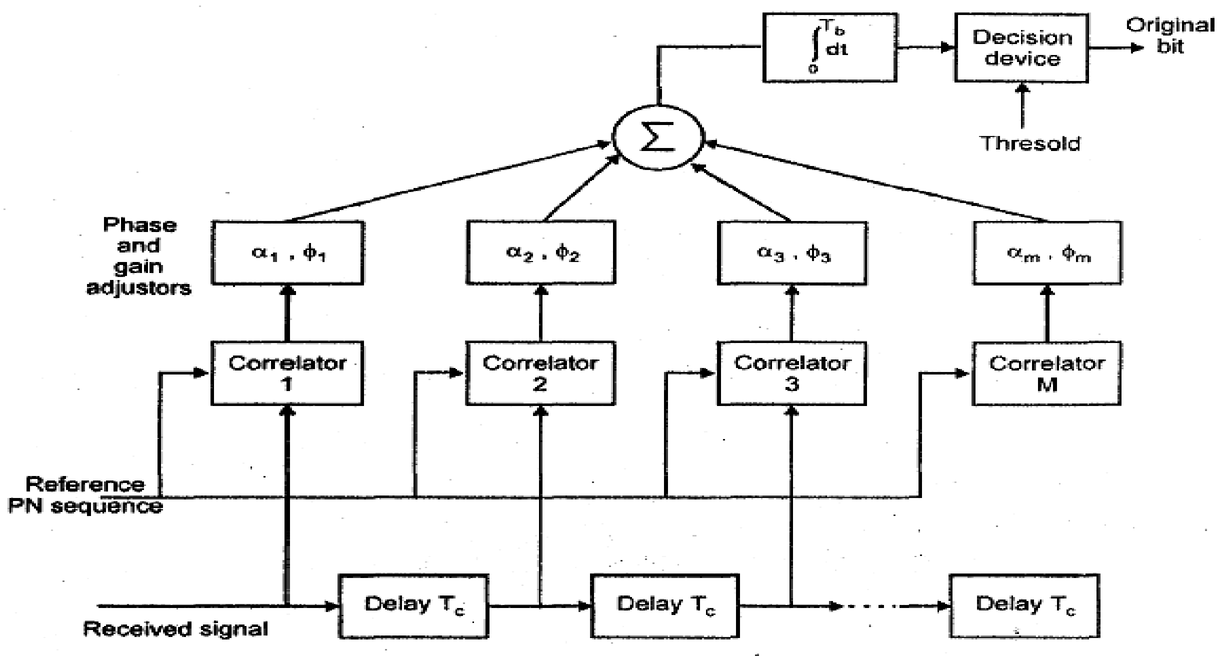


Fig. 11: Rack Receiver

1. Due to multipath propagation the original signals are delayed.
2. This time delayed versions of original signals are combined by CDMA receivers to improve the SNR.
3. The RAKE receivers minimizes the effect of multipath by using a correlation method to detect the echo signals individually and then adding them algebraically.
4. The receiver consists of a number of correlator connected in parallel.
5. Each correlator has two inputs
   i) A delayed version of the received signal.
   ii) A similar PN sequence used as the spreading code to generate the spread spectrum modulated signal at the transmitter
6. The bandwidth of the PN sequence should be sufficiently high to identify the significant echoes in the received signal.
7. the functions of the phase and gain adjusters are

   i). An appropriate delay is introduced in each correlator output so that phase angles of the correlator outputs are in agreement with each other.

   ii). If the correlator responds to strongest path in the multipath environment are accelerated and the correlators responds to non-significant path are suppressed.

8. These phase adjusted correlator outputs are combined by linear summer.
9. The linear summer output is **y(t)**=$\sum_{k=1}^{M} \alpha k\, xk(t)$

   Xk(t)→ phase compensated output of kth correlator.

   M= number of correlator in the receiver.

10. Then original data is estimated from y(t) using a decision device.

Advantages:

1. Maximum utilization of the channel takes place.
2. Synchronization is not necessary.

Disadvantages:

1. Chance of data collision because of overlapping.
2. Protocols are necessary to avoid collision.

Space Division Multiple access(SDMA);

1. Space Division Multiple access controls the radiated energy for each user in the space.
2. SDMA serves different users by using spot beam antennas.
3. These areas may be served by the same frequency or different frequencies.
4. However for limited co-channel interference it is required that the cells are sufficiently separated. This limits the number of cells in a region can be divided into and hence limits the frequency re-use factor.
5. Frequency re-use concept is used to increase the capacity of a channel.
6. In a practical cellular environment it is impossible to have just one transmitter fall within the receiver beam width.
7. Therefore we can use other multiple access techniques along with SDMA.
8. When different areas are covered by the antenna beam, frequency can be re-used, in which case TDMA or CDMA is employed, for different frequencies FDMA can be used.

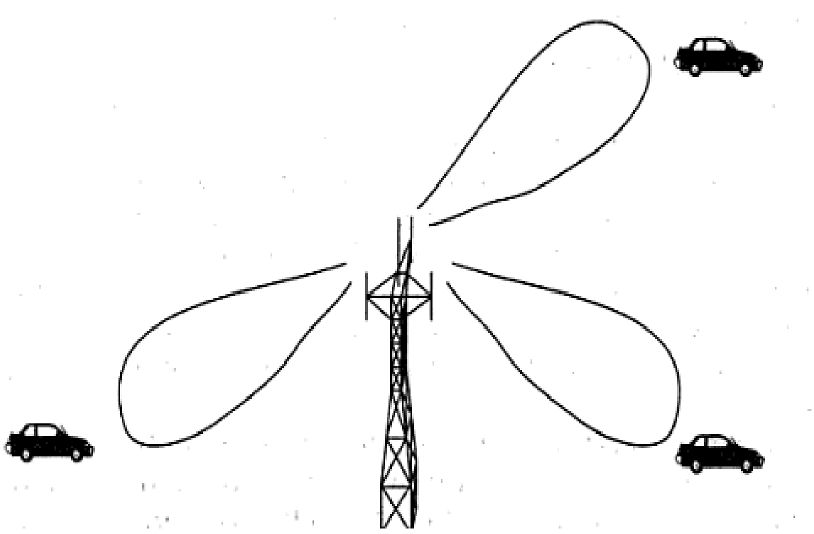


Fig. 11: A spatially filtered base station antenna serving different users by using spot beams.

Advantages:

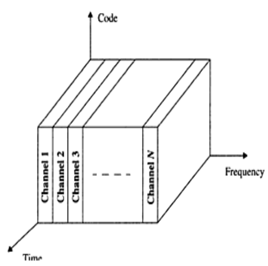
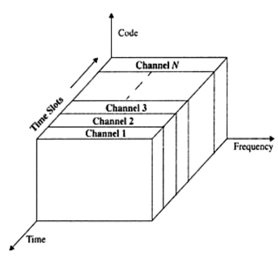
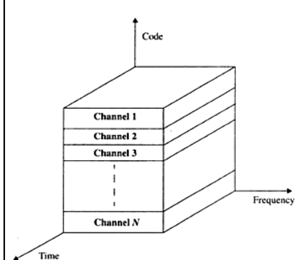
1. Very simpler technique.
2. Increases capacity per km²

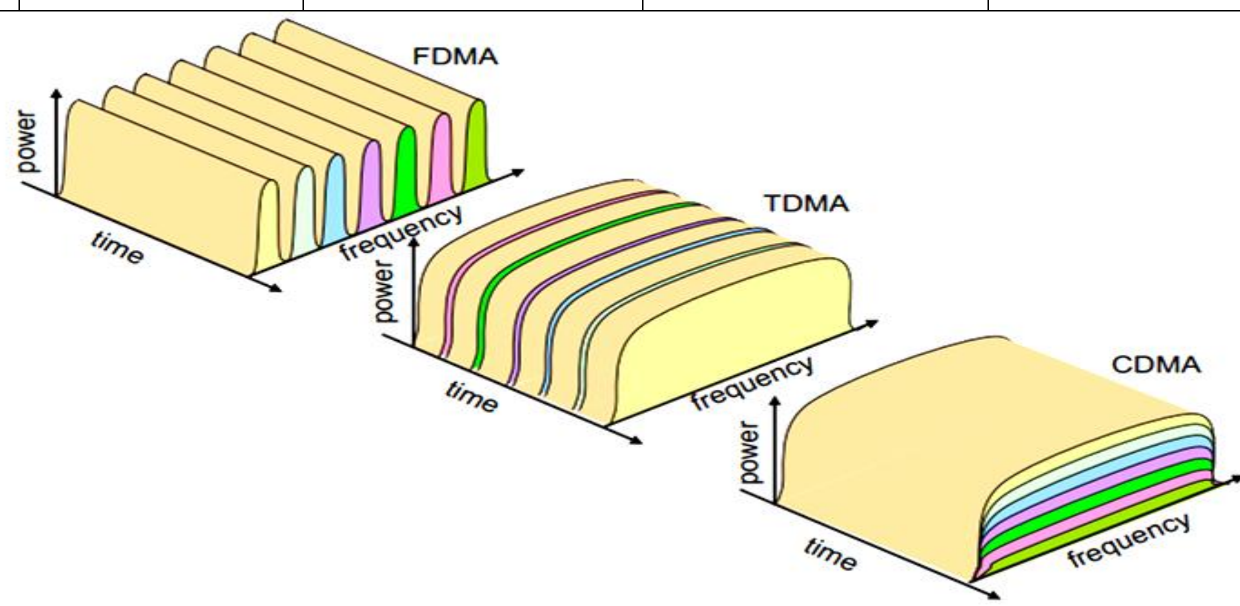
Disadvantages:

1. Inflexible, since antennas are typically fixed.
2. It can be used only with TDMA CDMA ,or FDMA

Comparison of CDMA, FDMA and TDMA:

S.No.	Parameter	FDMA	TDMA	CDMA
1.	Technique	Each channel is	Each channel is	Each channel is

		allocated the individual frequency slots	allocated the individual time slots	allocated the individual code words
2.	Synchronization	No synchronization is required	Time synchronization is required	No synchronization is required
3.	Power efficiency	Power efficiency is reduced	Full power efficiency is possible	Full power efficiency is possible
4.	Channel bandwidth	Channel bandwidth is subdivided into channels	Entire channel bandwidth is used by the user	the Entire channel bandwidth is used by the user
5.	Code word requirement	No code word	No code word	code word required
6.	Guard time and bands	***Guard bands*** are required	***Guard times*** are required	***Guard time and bands*** are required
7.	Interference effects	More	More	Less
8.	structure			



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Department of Electronics and Communication Engineering

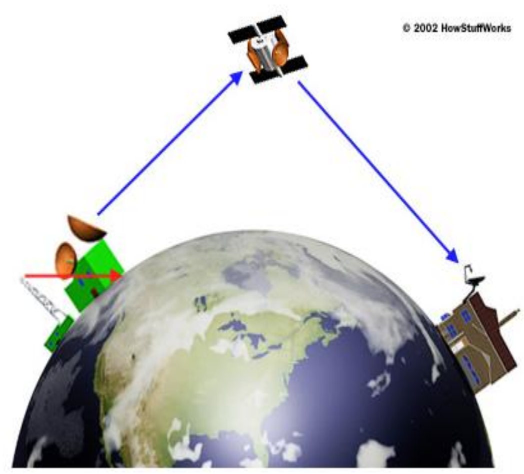
COMMUNICATION ENGINEERING

UNIT V SATELLITE, OPTICAL FIBER – POWERLINE, SCADA 9

Orbits : types of satellites : frequency used link establishment, MA techniques used in satellite communication, earth station; aperture actuators used in satellite – Intelsat and Insat: fibers – types: sources, detectors used, digital filters, optical link: power line carrier communications: SCADA

INTRODUCTION:

1. A satellite is a celestial body that orbits around a planet.
2. Artificial satellites can be launched into orbit for a variety of purposes.
3. A communication satellite is a microwave repeater in the sky that consists of the following components:Transmitter,Receiver,Amplifier,Regenerator
   ,
   Filter,Onboardcomputer,Multiplexer,
   De-multiplexer,AntennaWaveguide.
4. A satellite radio repeater is called a transponder.
5. A satellite system consists of
   - ✓ one or more satellite space vehicles
   - ✓ a ground-based station to control the operation of the system
   - ✓ a user network of earth stations that provides the interface facilities for the transmission and reception of communication traffic through the satellite system.



6. Transmissions to and from satellites are categorized as either bus or pay load.
7. The bus includes control mechanisms that support the pay load operation.
8. The pay load is the actual user information conveyed through the system.

How satellite works?

1. A Earth Station sends message in GHz range. **(Uplink)**
2. Satellite Receive and retransmit signals back. **(Downlink)**
3. Other Earth Stations receive message in useful strength area. **(Footprint)**

ORBITS:

- ✓ **The satellite can be rotated around the earth through various paths.These paths are called orbits of the satellite.**
- ✓ When the satellite is movimg in the orbits ,it stays in position because the centripetal force on the satellite balances the gravitational attractive force of the earth.
- ✓ This balance depends on the distance from the earth ,the speed of the satellite,earth's radius and gravitational force of the earth.
- ✓ The orbit of the satellite is selected based on the coverage area, transmission path loss ,delay time, and the period for which satellite is visible from the point on the earth.

TERMS USED TO DESCRIBE ORBIT:

Apogee: The point in an orbit that is located ***farthest from earth.***

Perigee: The point in an orbit that is located ***closest to the earth.***

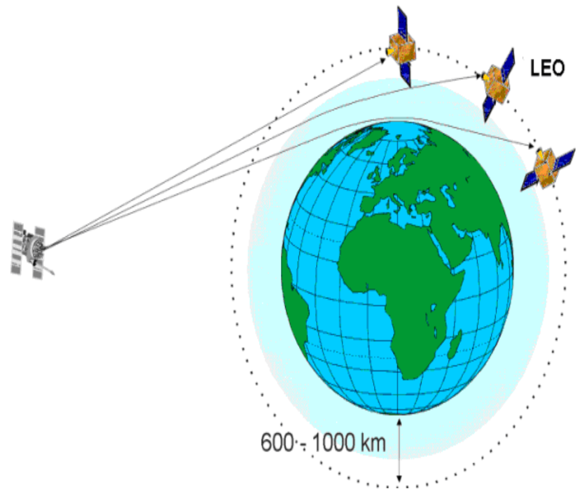
Major axis: The line joining the perigee and apogee through the center of earth.otherwise called line of apsides.

Minor axis : The line perpendicular to the mojor axis and halfway between the perigee and apogee.

Semiminor axis: Half the distance of the minor axis is called the semiminor axis.

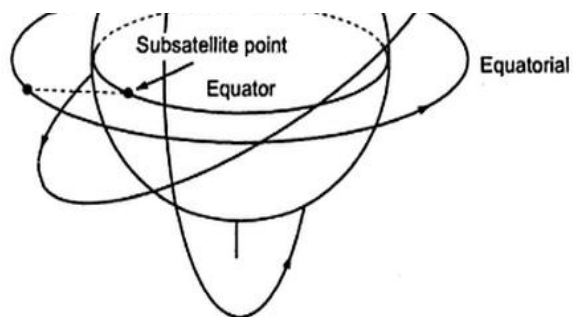
TYPES OF ORBIT:

Inclined orbits are virtually all orbits that does not travel directly above the equator or directly over the north and south poles.



- ✓ The angle of inclination vary between o deg and 180 deg.
- ✓ It provides coverage to polar regions.
- ✓ To provide coverage to high latitudes,inclined orbits are generally elliptical.
- ✓ The satellite remains visible for a longer period of time to the high latitude regions if the apogee is placed above the high latitude region.

A polar orbit is a orbit in which the satellite rotates in a path over the north and south poles in an orbit perpendicular to the equatorial plane.



- ✓ The angle of inclination is nearly 90 deg .
- ✓ 100% of the earth surface is covered with a single satellite in polar orbit.
- ✓ Satellites in polar orbit rotate around the earth in longitudinal orbit while earth is rotating on its axis in a latitudinal rotation.

An Equatorial orbit is when the satellite rotates in a orbit directly above the equator ,usually in a circular path.

- ✓ The angle of inclination is 0 deg .
- ✓ There are no ascending or descending nodesand hence no line of nodes.
- ✓ All geosynchronous satellites are in equatorial orbits.

Geostationary orbit:

- ✓ ***If the satellite is moving along with the earth(ie*** angular velocity os satellite is equal to the earth's angular velocity) ***the satellite appears to be stationary to the point on earth***. Such satellite paths are called geostationary orbit.
- ✓ the circular equatorial orbit is exactly in the plane of equator on the earth.
- ✓ All the points on this orbit are at equal distance from earth's surface.
- ✓ The rotational period of earth around its axis is 23 hours and 56 minutes.

Geo synchronous orbit:

- ✓ When the inclination of orbit is not zero or eccentricity is not zero then it is called geo synchronous orbit.
- ✓ The period of geosynchronous orbit is equal to period of revolution of earth itself.

Low Earth Orbit(LEO): Height of LEO satellites is about 1500 km.

- ✓ Altitude (375-1000 miles)
- ✓ Revolution time: 90 min - 3 hours.
- ✓ **Advantages:**

- Reduces transmission delay
- Eliminates need for bulky receiving equipment.

✓ **Disadvantages:**

- Smaller coverage area.
- Shorter life span (5-8 yrs.) than GEOs (10 yrs).

✓ **Subdivisions:** Little, Big, and Mega (Super) LEOs.

Medium Earth Orbit(MEO) :Height of MEO satellites lies between1500 km. to 36000.

- ✓ Circular orbit at an altitude in the range of 5000 to 12,000 km
- ✓ Orbit period of 6 hours
- ✓ Diameter of coverage is 10,000 to 15,000 km
- ✓ Round trip signal propagation delay less than 50 ms
- ✓ Maximum satellite visible time is a few hours

Sun synchronous orbit:

- ✓ Plane of this orbit maintains a constant aspect angle with the direction of sun.
- ✓ Earth resources satellites are launched in sunsynchronous orbit.

Molniya orbit: It is the special case off highly elliptical orbit.

- ✓ It provides coverage to northern regions.
- ✓ It has eccentricity of 0.74.

TYPES OF SATELLITES:

1. Geosynchronous satellite OR Geostationary satellite.
2. Non synchronous satellites.

GEOSYNCHRONOUS SATELLITE (OR) GEOSTATIONARY SATELLITE:

1. These satellites rotates above the equator with the same angular velocity as earth.
2. The angle of inclination is 0 deg.
3. Since these satellites appears to remain in a fixed location ,no special antenna tracking system is necessary.
4. A single high altitude geosynchronous satellite can provide reliable communications to approximately 40% of the earth's surface.
5. Geosynchronous orbits are circular,therefore the speed of rotation is constant throughout the orbit.
6. There is only one geosynchronous orbit;however it is occupied by a large number of satellites.
7. The unbalanced forces causes geosynchronous satellites to drift slowly away from their assigned locations.
8. Ground controllers must periodically adjust satellite positions to counteract these forces.

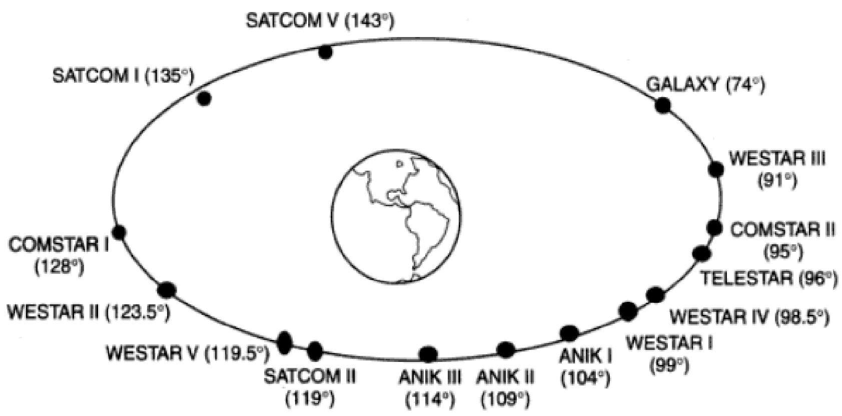


FIGURE 25-8 Satellites in geosynchronous earth orbits

ADVANTAGES OF GEOSYNCHRONOUS SATELLITE:

1. Geosynchronous satellites remains stationary in respect to earth station.Therefore tracking equipment is not required at the earth stations.
2. These satellites are available to all earth stations within their shadow 100%of the time.
3. There is no need to switch from one geosynchronous satellite to another as they orbit overhead.cosequently ,there are no transmission breaks due to switching times.
4. The effects of Doppler shifts are negligible.

DISADVANTAGES OF GEOSYNCHRONOUS SATELLITE:

1. It requires sophisticated and heavy propulsion devices on board to keep them in a fixed orbit.
2. High altitude geosynchronous satellites introduce much longer propagation delays.
3. The round trip propagation delay between two earth stations through a satellite is between 500ms and 600ms.
4. It requires high power transmitter and more sensitive receivers because of the longer distances and greater pah losses.
5. High precision spacemanship is required to place a geosynchronous satellite into orbit and to keep it there.

NON SYNCHRONOUS SATELLITES:

1. Nonsynchronous satellites rotate around earth in an elliptical or circular pattern.
2. In circular orbit,the speed or rotation is constant.
3. However in elliptical orbit the speed depends on the height the satellite is above earth.
4. The speed of the satellite is greater when it is close to earth than when it is farther away.

SATELLITE SYSTEM LINK MODELS:

It consists of three basic sections:

1. an uplink,
2. a satellite transponder and
3. a downlink.

Uplink model:

1. The primary component within the uplink section of a satellite system is the ***earth station transmitter.***
2. A typical earth system consists of an IF modulator, an IF to RF microwave up converter, a high power amplifier (HPA) and output band pass filters.

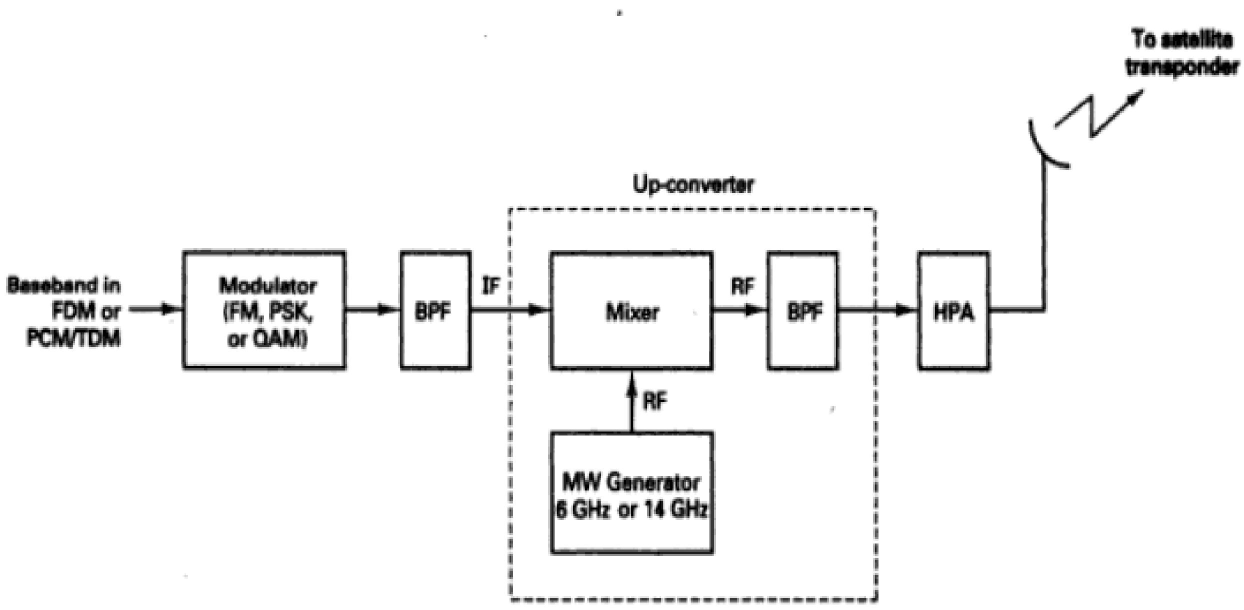
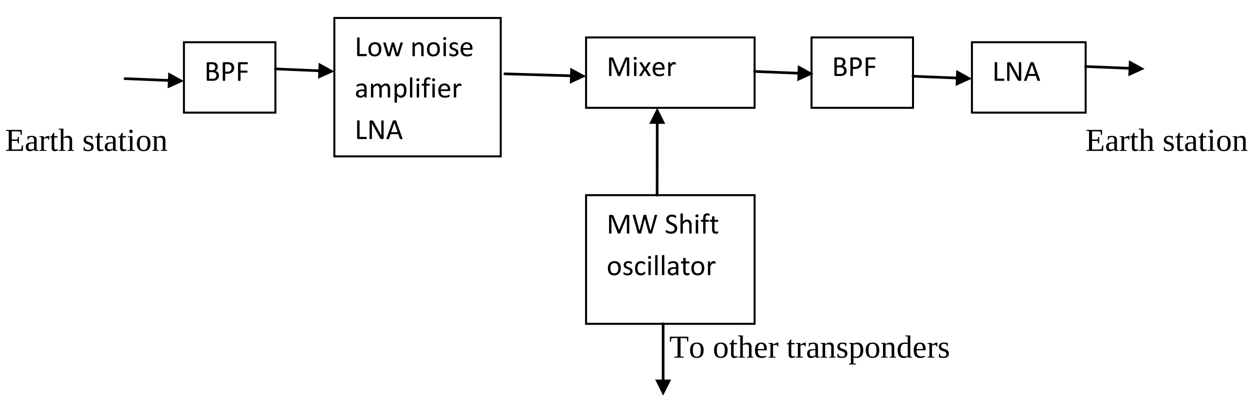


FIGURE 25-19 Satellite uplink model

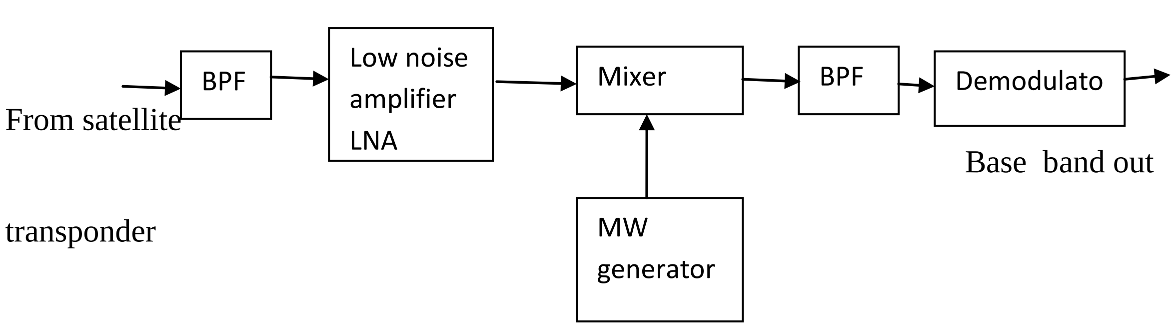
3. The IF modulator converts the input base band signals to either a FM, a PSK or a QAM modulated intermediate frequency.
4. The up converter converts the IF to an appropriate RF carrier frequency.
5. The HPA provides adequate gain and output power to propagate the signal to the satellite transponder.
6. Klystrons and travelling –wave tubes are widely used HPAs.

Transponder:



1. A typical satellite transponder consists of an input band limiting device (BPF), an input low noise amplifier (LNA), a frequency translator, a low level power amplifier, and an output band pass filter.
2. This transponder is an RF to RF repeater.
3. Other transponder configurations are IF and base band repeaters similar to those used in microwave repeaters.
4. The input BPF limits the total noise applied to the input of the LNA.
5. The output of the LNA is fed to a frequency translator, which converts the high band uplink frequency to the low band downlink frequency.
6. The low level power amplifier, which is commonly a traveling wave tube, amplifies the RF signal for transmission through the downlink to earth station receivers.
7. Each RF satellite channel requires a separate transponder.

Downlink model:



1. An earth station receiver includes an input BPF, an LNA and an RF to IF down converter.
2. The BPF limits the input noise power to the LNA.
3. The LNA is a highly sensitive, low noise device.
4. The RF to IF down converter is a mixer/band pass filer combination that converts the received RF signal to an IF frequency.

Cross links:

1. Occasionally, there is an application where it is necessary to communication between satellites.
2. This is done by using satellite cross links or inter satellite links (ISLs).
3. Disadvantage of using an ISL is that both the transmitter and the receiver are space bound.

FREQUENCY USED IN SATELLITES:

Presently **C band and X-bands** are widely being used by satellites.

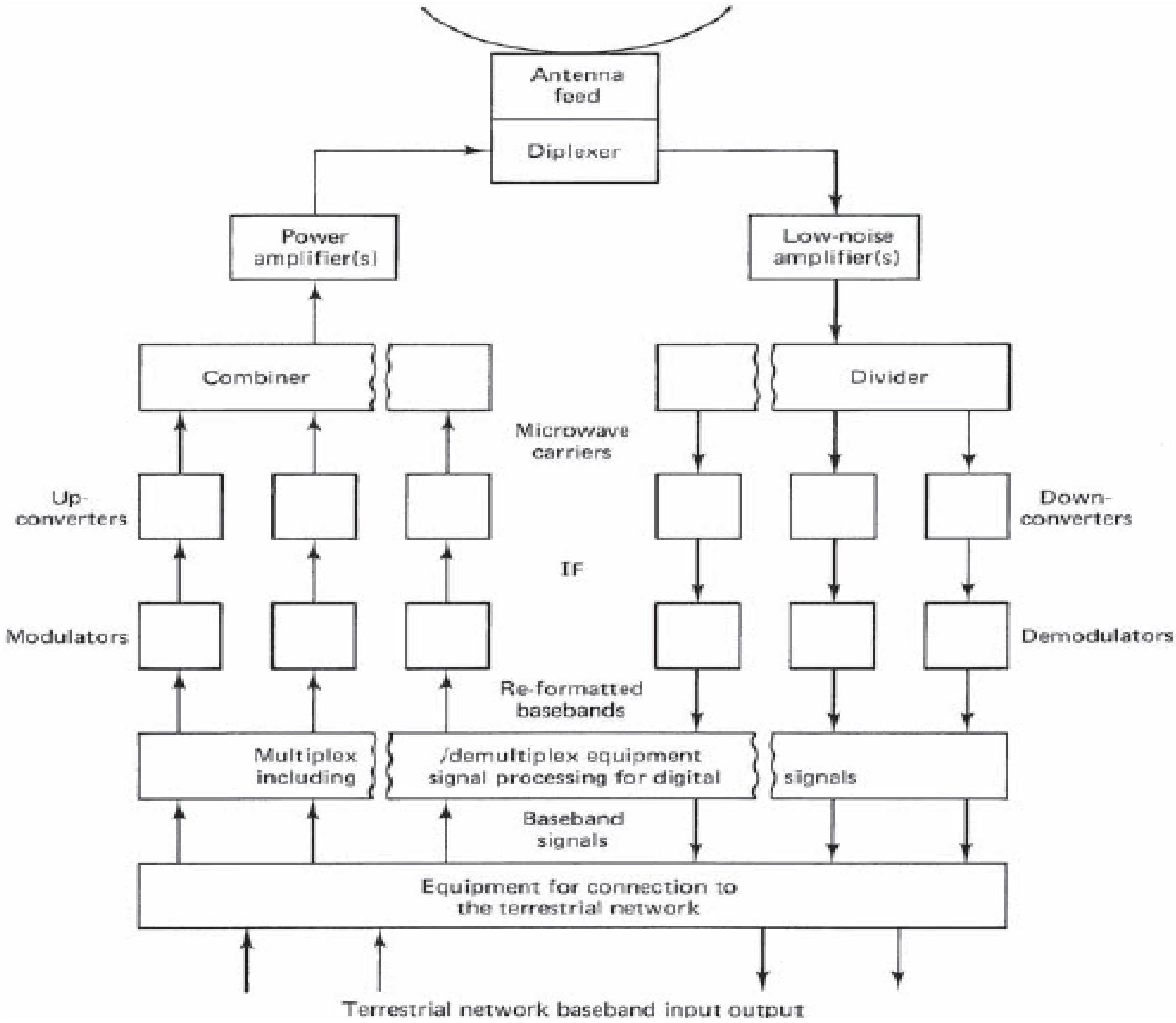
C band→ uplink frequency (5.9 -6.4 GHz)

downlink frequency(3.7 to 4.2GHz)

X band → uplink frequency (7.9-8.4GHz)

downlink frequency(7.25-7.75GHz)

Earth station:



1. The first block shows the interconnection equipment required between satellite station and the terrestrial network.
2. The multiplexing equipment combines the signals without mutual interference.
3. The multiplexed signal is modulated onto a carrier wave at an intermediate frequency usually 70 MHz.parallel IF stages are required for each microwave carrier to be transmitted.
4. After amplification at the 70MHz IF,the modulated signal is then upconverted to the required carrier frequency.
5. The upconverted carriers are combined and the resulting wideband signal is amplified.
6. The wideband power signal is fed tio the antenna through the diplexer,which allows the antenna to handle transmit and receive signals simultaneously.
7. The station antenna functions in both,the transmit and receive modes ,but at different frequencies.
8. In the receiver branch the incoming wideband signal is amplified in an Low Noise Amplifier.
9. Then it is passed to a divider network which seperates out the individual microwave carriers.
10. Then each carrier is downconverted to an IF band and passed onto the multiplex block where the
11. multiplexed signals are reformatted as required by the terrestrial network.

MULTIPLE ACCESS TECHNIQUES USED IN SATELLITE COMMUNICATION:

Satellite multiple accessing implies that more than one user has access to one or more radio channels (transponders)within a satellite communication channel.

The three most commonly used multiple accessing arrangements are

1. Frequency division multiple access.
2. Time-division multiple access
3. Code division multiple access.

Frequency division multiple access:

1. FDMA is a technique in which the a given RF bandwidth is divided into smaller frequency bands called subdivisions.
2. Each subdivisions has its own IF carrier frequency.
3. A control mechanism is used to ensure that two or more earth stations do not transmit in the same subdivision at the same time.
4. If each subdivision carries only one voice band channel then it is known as single-channel per carrier.
5. When several voice band channels are frequency multiplexed together to form a composite signal,then it is called as multiple channel per carrier.
6. With FDMA each earth station may transmit simultaneously within a same spectrum but on different voice band channels.

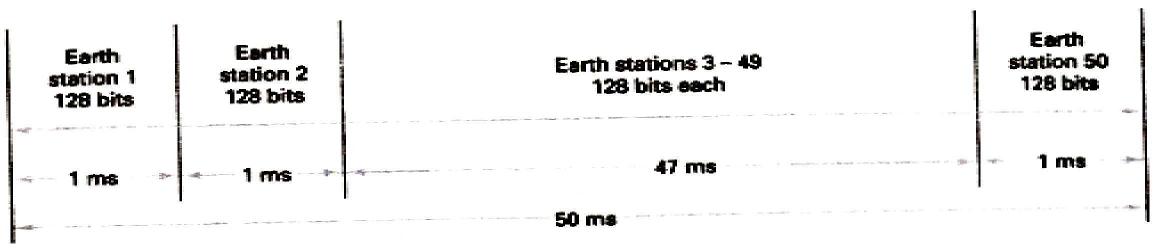


Figure: FDMA, SPADE common signaling channel (CSC)

SPADE demand access system:

1. The satellite capacity of all the stations in FDMA network is pooled and it is used on SCPC(single carrier per channel) basis.
2. When the call is to be made,a pair of frequencies is assigned to the duplex circuit.This is called SCPC demand access method.
3. This system allows circuits to be selected by an terminal on demand.they are assigned from the satellite pool as required.

Advantage:

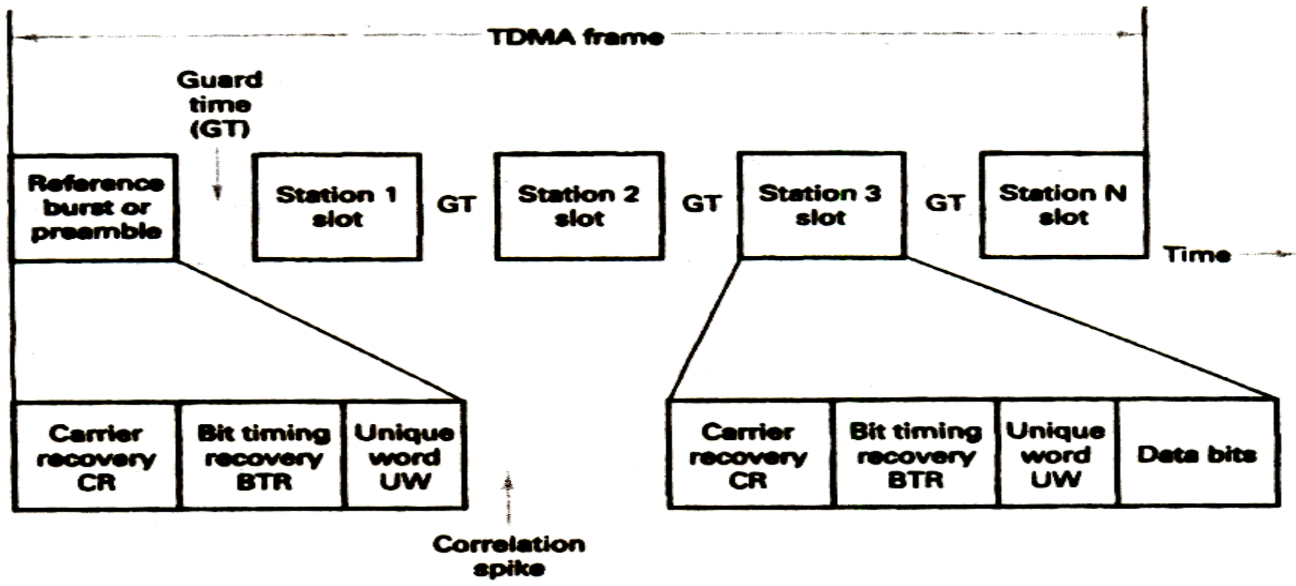
Power required for the transmission at satellite as well as earth station is dependent on the number of channels being transmitted.

Disadvantage:

Carriers from multiple earth station may be present in a satellite transponder at the same time .this results in cross-modualtion distortion.

Time-division multiple access:

1. TDMA is the most efficient method of transmitting digitally modulated carriers.
2. TDMA is a method of time-division multiplexing digitally modulated carriers of earth station within a satellite network through a satellite transponder.
3. With TDMA each earth station transmits a short burst of modulated carrier during a precise time slot within a TDMA frame.
4. Each station burst is synchronized so that it arrives at the satellite transponder at a different time.
5. Consequently only one earth station carrier is present in the transponder at any given time, thus avoiding the collision with another earth's station.
6. Each earth station receives the bursts from all other earth stations and must select from them the traffic destined only for itself.
7. Transmissions from all earth stations are synchronized.
8. TDMA frame consists of reference burst contains a carrier recovery sequence (CRS),bit timing recovery(BTR), and unique word (UW).



- ✓ ***a carrier recovery sequence (CRS***) is used by the receiving station to recover the frequency and phase of the carrier for demodulation.
- ✓ **bit timing recovery(BTR),** is used to recover the clock pulses.
- ✓ **unique word (UW):** it is transmitted at the end of each reference burst.The UW ia typically a string of 20 successive binary 1's terminated with a binary 0.
- ✓ Each earth station demodulates and integerates the UW sequence.
- ✓ The UW sequence is used to establish a precise time reference that each of the earth station uses to synchronize the transmission of its burst.

9. So each station waits a different length of time before it begins transmitting.

ADVANTAGES:

1. TDMA only the carrier from one earth station is present in the satellite transponder at a time.,thus reducing intermodulation distortion.
2. Each earth station is capable of transmitting and receiving multiple carrier frequencies.
3. TDMA is much better suited for the transmission of digital information than FDMA.

DISADVANTAGES:

1. Synchronization is needed.
2. Bit and frame timing must be maintained.

CODE DIVISION MULTIPLE ACCESS:

1. Each earth station may transmit at any time and can use all the bandwidth allocated to a particular satellite system .
2. With CDMA all earth satations within the system may transmit on the same frequency at the same time.
3. Transmissions are separated through envelope encryption /decryption technique.
4. Each earth station's transmissions are encoded with a unique binary word called a chip code.
5. To receive a particular earth sattion's transmission ,a receive station must know the chip code for that station.

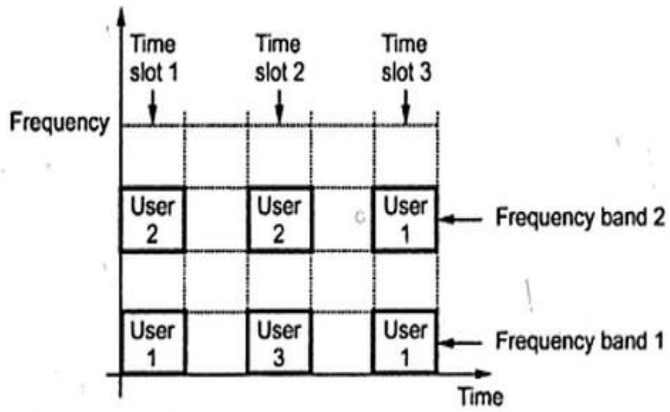


Fig. 4.9.7 CDMA transmission

Advantages:

- ✓ The entire bandwidth of the satellite channel may be used for each transmission from every earth station.

Disadvantage:

- ✓ Time synchronization is required.

INTELSAT:

1. **INTELSAT** stands for International Telecommunications Satellite Organization.
2. **INTELSAT** is an intergovernmental consortium owning and managing a constellation of communications satellites providing international broadcast services.
3. Intelsat's first satellite, the Intelsat I (nicknamed *Early Bird*), was placed in geostationary orbit above the Atlantic Ocean by a Delta D rocket.
4. For international traffic, INTELSAT covers three main regions—**the *Atlantic Ocean Region* (AOR), the *Indian Ocean Region* (IOR), and the *Pacific Ocean Region* (POR).**
5. For the ocean regions the satellites are positioned in geostationary orbit above the particular ocean, where they provide a transoceanic telecommunications route.
6. These satellites provide a much wider range of services than those available previously, including such services as Internet, DTH TV, tele-medicine, tele-education, and interactive videoand multimedia.
7. In addition to providing transoceanic routes, the INTELSAT satellites are also used for domestic services within any given country and regional services between countries.
8. Two such services are Vista for telephone and Intelnet for data exchange.

INSAT:

1. **INSAT** stands for the ***Indian National Satellite System.***
2. The ***Indian National Satellite System*** is a series of multipurpose Geo-stationary satellites launched by ISRO to satisfy the telecommunications, broadcasting, meteorology, and search and rescue operations.
3. Commissioned in 1983, INSAT is the largest domestic communication system in the Asia Pacific Region.
4. The system provides services to telecommunications, television broadcasting, weather forecasting, disaster warning and Search and Rescue operations
5. INSAT satellites provide transponders in various bands (C, S, Extended C and K_u) to serve the television and communication needs of India.
6. Some of the satellites also have the Very High Resolution Radiometer (VHRR), CCD cameras for metrological imaging.

7. The satellites also incorporate transponder(s) for receiving distress alert signals for search and rescue missions in the South Asian and Indian Ocean Region, as ISRO is a member of the Cospas-Sarsat programme.
8. Some of the INSATs also carry instruments for meteorological observation and data relay for providing meteorological services. KALPANA-1 is an exclusive meteorological satellite.
9. INSAT space segment consists of 24 satellites out of which 10 are in service (INSAT-3A, INSAT-4B, INSAT-3C, INSAT-3E, KALPANA-1, INSAT-4A, INSAT-4CR,GSAT-8, GSAT-12 and GSAT-10.

OPTICAL FIBRE COMMUNICATION:

- ✓ An optical fibre communication is the one in which the information is transmitted in the form of light.
- ✓ The light is guided through an optical fibre by total internal reflection principle.

ADVANTAGES OF OPTICAL FIBRE COMMUNICATION:

1. Basically light waves are also electromagnertic waves with very high frequencies in the range of 3000000GHz.Hence ***very large bandwidths are possible in fibre optic communication***.
2. This increases the number of channels and frequency bands.Thus ***the information carrying capacity of optical fibres is much higher than conventional microwave radio systems.***
3. The optical fibres are made up of silica glass or silicon dioxide(sand).hence ***optical fibres are very low cost and cheap compared to metallic wires.***
4. Since it does not involve any electric voltage or current there is no electromagnetic interference and no cross talks.
5. Transmission losses are very minimum.
6. Optical fibres are more flexible,and offers light weight.

Disadvantages:

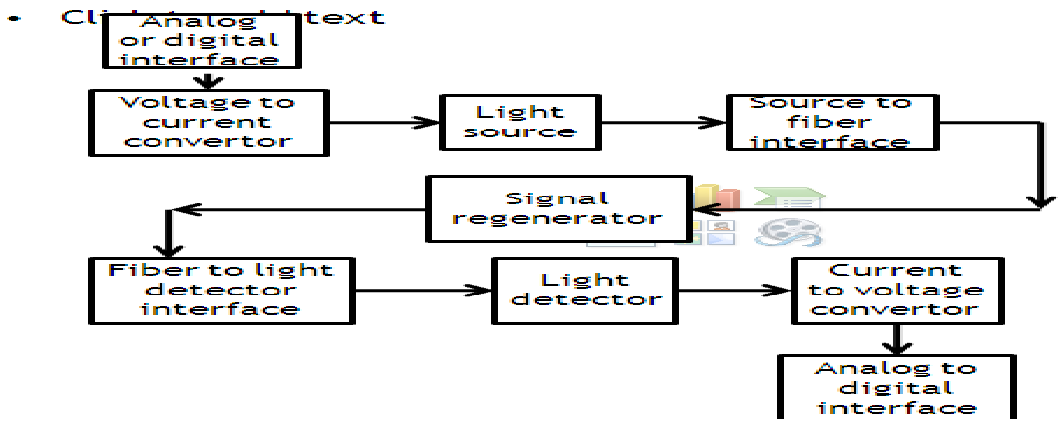
1. Installation cost and Interfacing cost is higher.
2. Strength: Lower tensile strength
3. more expensive to repair/maintain.
4. Specialized and sophisticated tools are needed

BLOCK DIAGRAM OF AN OPTICAL FIBER COMMUNICATION SYSTEM:

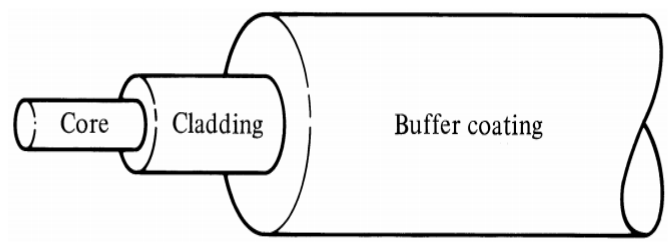
Three primary building blocks are the transmitter, the receiver, and the optical fiber cable.

- ✓ Transmitter is comprised of a voltage to current converter, a light source and a source to fiber interface (light coupler).
- ✓ The fiber guide is the transmission medium, which is either an ultra pure glass or a plastic cable.
- ✓ It may be necessary to add one or more regenerators to the transmission medium, depending on the distance between the transmitter and receiver.
- ✓ The regenerator performs light amplification. But in reality the signal is not actually amplified, it is reconstructed.
- ✓ The receiver includes a fiber to interface, a photo detector and a current to voltage converter.

In the transmitter, the light source can be modulated by a digital or an analog signal.



- ✓ The voltage to current converter serves as an electrical interface between the input circuitry and the light source.
- ✓ The light source is either an infrared light emitting diode (LED) or an injected laser diode (ILD).
- ✓ The amount of light emitted by either an LED or ILD is proportional to the amount of drive current.
- ✓ Thus, the voltage to current converter converts an input signal voltage to a current that is used to drive the light source.
- ✓ The light outputted by the light source is directly proportional to the magnitude of the input voltage.
- ✓ The source to fiber coupler is a mechanical interface.



- ✓ Its function is to couple light emitted by the light source into the optical fiber cable.
- ✓ The optical fiber consists of a glass or plastic fiber core surrounded by a cladding and then encapsulated in a protective jacket.
- ✓ The fiber to light detector coupling device is also a mechanical coupler.
- ✓ Its function is to couple as much light as possible from the fiber cable into the light detector.
- ✓ The light detector is generally a PIN (p-type intrinsic n-type) diode, an APD (avalanche photodiode), or a phototransistor.
- ✓ All three of these devices convert light energy to current.
- ✓ A current to voltage converter is required to produce an output voltage proportional to the original source information.
- ✓ The current to voltage converter transforms changes in detector current to changes in voltage.
- ✓ The analog or digital interfaces are electrical interfaces that match impedances and signal levels between the information source and destination to the input and output circuitry of the optical system.

OPTICAL FIBRE:

1. Optical fibre consists of two concentric layers ***the light carrying core and the cladding*** and outer buffer coating.
2. The cladding acts as a refractive index medium and allows the light to be transmitted through the core and the other end with very little distortion or attenuation.
3. When light is introduced into the fibre the cladding refracts or reflects the light in a zigzag pattern through out the entire length of the core.
4. This process is possible because the angle of incidence and angle of reflection are equal.

TYPES OF OPTICAL FIBRE:

Optical fiber configuration:

Light can be propagated down an optical fiber cable using either reflection or refraction.

How the light propagates depends on the mode of propagation and the index profile of the fiber.

1. **Mode of propagation:**
   - Mode simply means ***path*** in fiber optics terminology.
   - If there is ***only one path*** for light rays to travel in a cable, it is called ***single mode.***
   - If there is ***more than one path***, it is called ***multimode.***
   - The number of modes possible for a multi mode fiber cable depends on the frequency (wavelength) of the light signal, the refractive indexes of the core and cladding and the core diameter.
   - Mathematically, the number of modes possible for a given cable can be approximated by the following formula:

   $$N \approx ((\Pi d/\lambda) SQ(n_1^2 - n_2^2))^2$$

   Where N = number of propagating modes

   D= core diameter (meters)

   λ = wavelength (meters)

   n_1 = refractive index of core

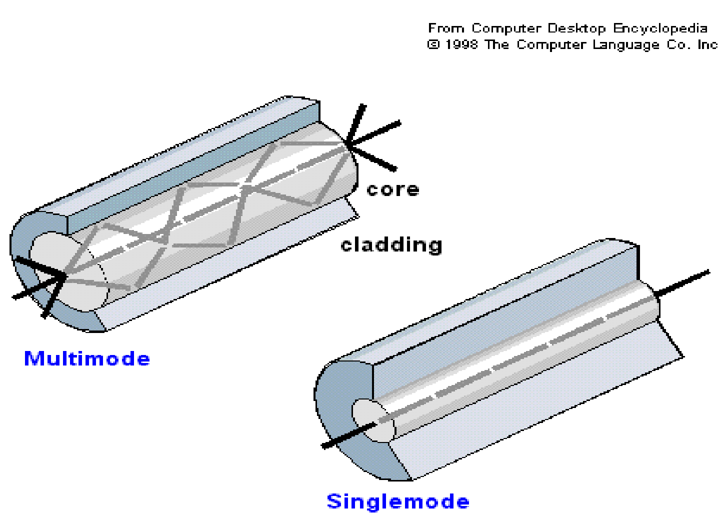
   n_2 = refractive index of cladding

2. **Index profile:**
   - The index profile of an optical fiber is a ***graphical representation of the magnitude of the refractive index across the fiber.***
   - The refractive index is plotted on the horizontal axis and the radial distance from the core axis is plotted on the vertical axis.

There are two basic types of index profiles. They are

1. Step index
2. Graded index

1. ***Step index:***
   - It has a ***central core with a uniform refractive index.***
   - An ***outside cladding that also has a uniform refractive index surrounds the core***; however the refractive index of the cladding is less than that of the central core.



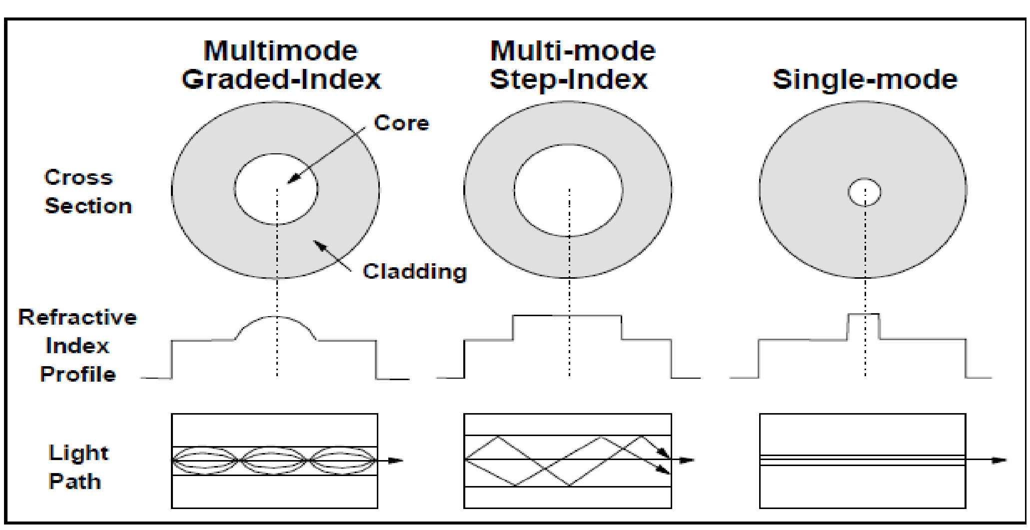
2. ***Graded index:***
   - There is ***no cladding and the refractive index of the core is non uniform.***
   - It is the highest in the center of the core and decreases gradually with distance towards the outer edge.
   - The index profile shows a core density is maximum in the center and decreases symmetrically with distance from the center.

<u>CLASSIFICATION OF OPTICAL FIBRE:</u>

The three practical types of optical fiber configurations are

1. Single mode step index
2. Multi mode step index
3. Multi mode graded index

1. **Single mode step index:**
   - A ***single mode step index fiber has a central core*** of smaller diameter.
   - The diameter is sufficiently small so there is essentially only one path that light may propagates down the cable.
   - In the simplest form of single mode step index fiber, the outside cladding is air.
   - The refractive index of the glass core (n_1) is approximately 1.5 and the refractive index of the air cladding (n_2) is 1.
   - A single mode step index fiber has a wide external acceptance angle, which makes it relatively easy to couple light into the cable from an external source.
2. **Multi mode step index:**
   - Multi mode step index fibers are similar to the single mode step index fibers except the ***center core is much larger with the multimode configuration.***
   - This type of fiber has a large light to fiber aperture and consequently, allows more external light to enter the cable.
   - The light rays that strike the core/cladding interface at an angle greater than the critical angle are propagated down the core in a zigzag fashion, continuously reflecting off the interface boundary.
   - The light rays that strike the core/cladding interface at an angle less than the critical angle ray enter the cladding and are lost.
   - As a result, all light rays don't follow the same path and consequently, don't take the same amount of time to travel the length of the cable.
3. **Multi mode graded index:**
   - These are characterized by a ***central core with a non uniform refractive index.***
   - Thus cable density is its maximum at the center and decreases gradually toward the outer edge.
   - Light ***rays propagate down this type of fibers through refraction rather than reflection.***
   - As a light rays propagates diagonally across the core toward the center, it is continually intersecting a less dense to more dense interface.
   - Light rays enter the fiber at many different angles. As the light rays propagate down the fiber, the rays travelling in the outer most area of the fiber travel a greater distance than the rays traveling near the center.
   - Because, the refractive index decreases with distance from the center and the velocity is inversely proportional to refractive index, the light rays travelling farthest from the center propagate at a higher velocity.



OPTICAL FIBER COMPARISON:

1. **Single mode step index:**

Advantages:

1. Minimum dispersion: All rays propagating down the fiber take approximately the same path; thus take approximately the same length of time to travel down the cable.
2. Because of high accuracy in reproducing transmitted pulses at the receive end, wider bandwidths and higher information transmission rates are possible.

Disadvantages:

1. Because the central core is very small, it is difficult to couple light into and out of this type of fiber.
2. Again, because of the small central core, a highly directive light source, such as a laser is required to couple light into a single mode step index fiber.
3. These are expensive and difficult to manufacture.

2. Multi mode step index:

Advantages:

1. These are relatively inexpensive and simple to manufacture.
2. It is easier to couple light into and out of multi mode step index fibers because they have a large source to fiber aperture.

Disadvantages:

1. Light rays may take different paths down the fiber, which results in large differences in propagation times.
2. This type of fiber has a tendency to spread out. A pulse of light propagating down a multi mode step index fiber is distorted more than with the other types of fibers.
3. The bandwidths and rate of information transfer rates possible with this type of cable are less than that possible with the other types of fiber cables.

3. **Multi mode graded index:**

There are no outstanding advantages and disadvantages of this type of fiber.

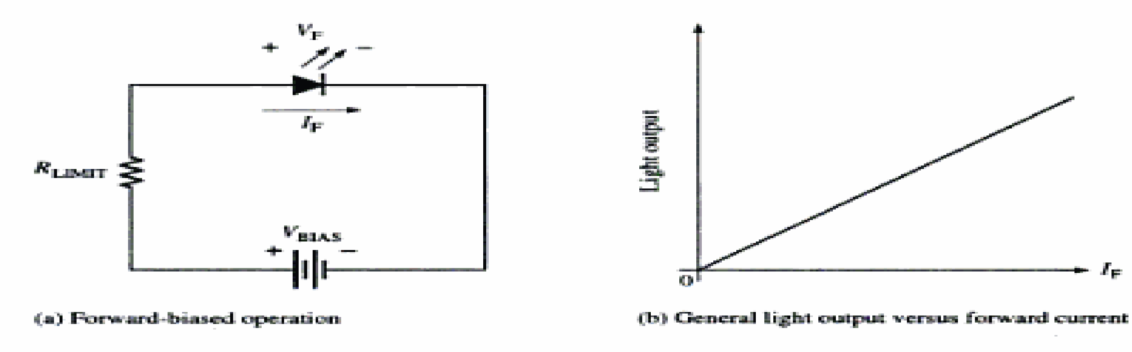
1. These are easier to couple light into and out of than single mode step index fibers but more difficult than multi mode step index fibers.
2. Distortion due to multiple propagation paths is greater than in single mode step index fibers but less than in multi mode step index fibers.
3. This multimode graded index fiber is considered an intermediate fiber compared to the other fiber types.

TYPES OF OPTICAL SOURCES:

1. Light Emitting Diode(**LED**).
2. Laser Diode

Light Emitting Diode(LED).

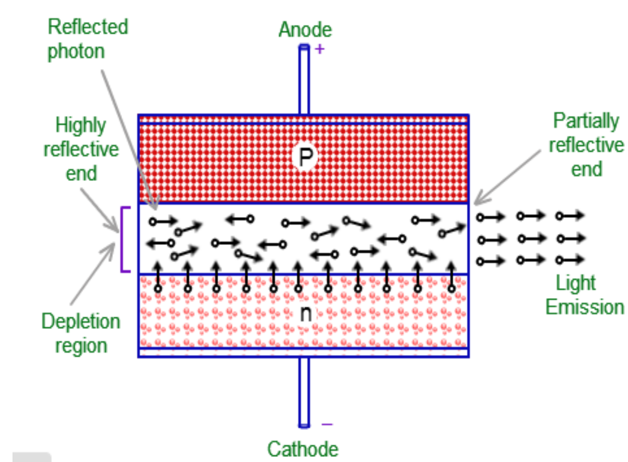
1. LED is a PN junction device which emits light when forward biased ,by a phenomenon called electroluminescence.
2. LED is made up of special type of semiconducting material such as gallium phospide(Gap) or gallium arsenide phosphide(GaAsP).



(a) Forward-biased operation

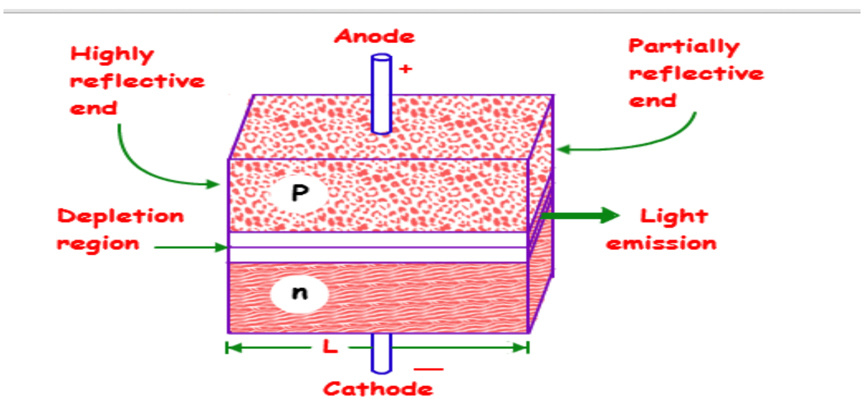
(b) General light output versus forward current

3. The PN junction of LED is formed by diffusing a thin layer of p-type material into the surface of n-type substrate.
4. When the diode is forward biasaed electrons and holes moves towards the junction and recombination takes place.
5. As aresult of recombination the electrons lying in the conduction band of N-region fall into the holes lying in the valence band of a p-region .
6. The difference of energy between the conduction band and the valence band is emitted in the form of light energy.
7. The brightness of the emitted light is directly proportional to the forward bias current.



Laser diode:

1. LASER Stands For Light Amplification By Stimulated Emissions Of Radiation.
2. Lasers are used to convert the electrical signal to light energy.
3. In direct bandgap semiconducting materials where high recombination velocities exist,optical gain can be achieved by creating population inversion of carriers through high level current injectionand by forming a resonant cavity.
4. This cavity is produced by cleaving the material along the faces perpendicular to the junction plane.



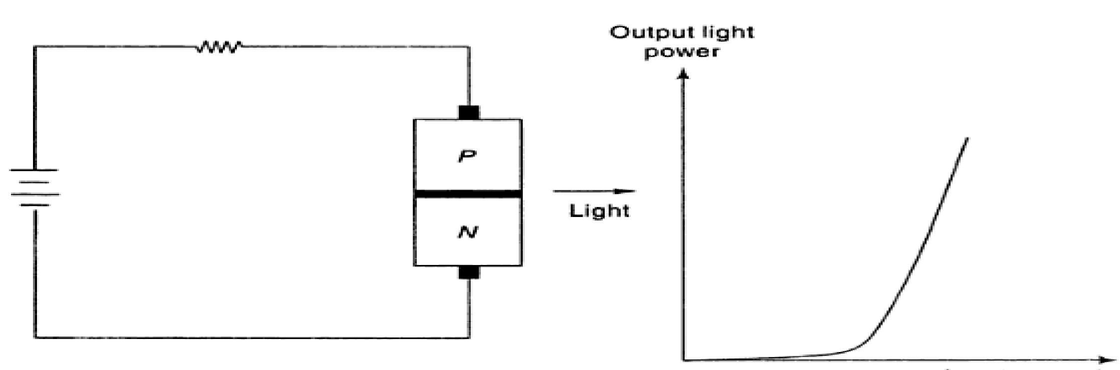


Fig. 5.34 Structure and characteristics of laser diode

5. In this diode the opposite ends of the junction are polished to get a mirror like surfaces.

6. When free electrons combine with holes, the emitted photons reflected back and forth between the mirror surfaces.
7. The region between the mirrored ends acts like a cavity that filters the light and purifies its colour.
8. As the photons bounce back and forth they induce an avalanche effect that causes all newly created photons to be emitted with the same phase.
9. One of the mirror surface is semi transparent.
10. From this surface a fine thread like beam of photons emerge out.
11. All the photons of laser light have same frequency and phase and hence coherent.
12. From the power output vs drive current characteristics it has been observed that it contains well defined threshold.

13.Below this threshold the device exhibits low levels of spontaneous emission

14. At the limiting current density stimulated emission occurs and emitted radiation increases linearly with drive current.

Optical fibres detectors :

1. PIN photo diode.
2. Avalanche photo diode.

PIN photo diode:

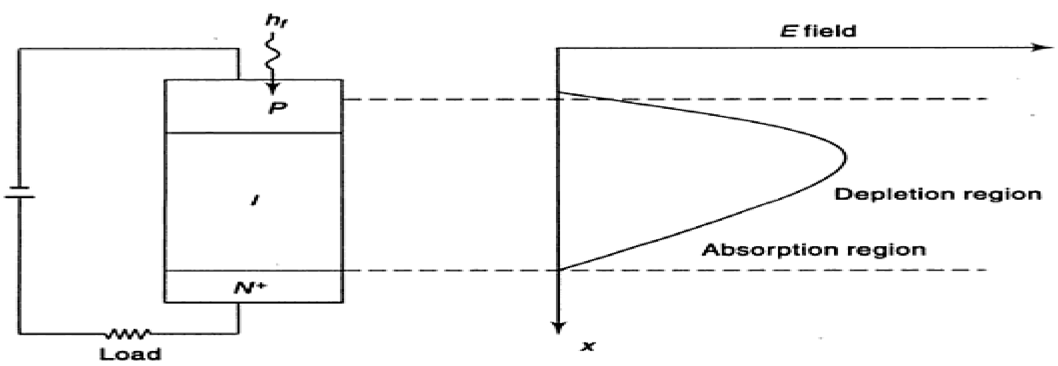


Fig. 5.32 Structure of PIN photodiode

1. It is used for the detection of light at the receiving end in the optical communication.
2. It is a three region reverse biased junction diode.
3. A layer of intrinsic silicon is sandwiched between heavily doped P and N type semiconducting materials.
4. The depletion layer extends upto the intrinsic layer where most of the absorption of light photons takes place.
5. The width of the intrinsic layer is large compared to the width of the other two layers.
6. This ensures large absorption of light photons in the depletion region.
7. Light photons induced on the PIN photo diode are absorbed in the absorption region which leads to the generation of electron hole pairs.
8. These charge carriers present in the depletion region drift under the influence of existing electric field that is set up due to the applied reverse bias.
9. The reverse current owing in the external circuit increases with the level of illumination.

Avalanche photo diode:

1. APD is used in the optical communication for the detection of light.
2. It converts the input light signal into electrical energy.
3. APD consists of a reverse biased PN junction.
4. The depletion region in the reverse biased PN junction is formed by positively charged donor atoms in the N-type semiconductor and negatively charged acceptor atoms in the p-type semiconductor.

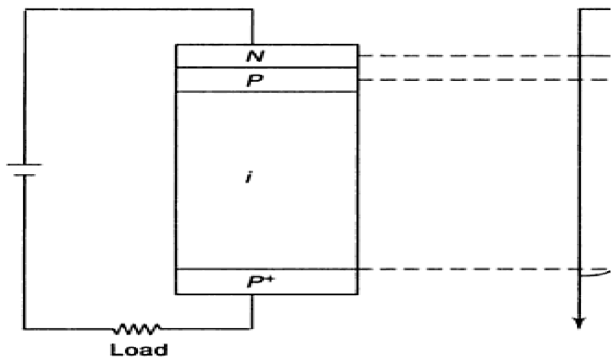
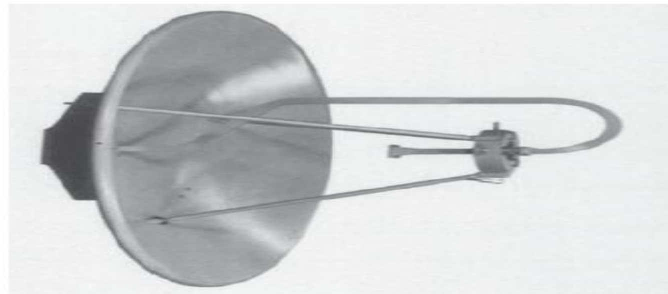


Fig. 5.33 Structure of



5. The electric field in the depletion region is very high where most of the photons are absorped and primary charge carriers are generated.
6. These charge carriers acquire sufficient energy from the electric field to excite new electron-hole pairs by a process known as impact ionisation.
7. Impact ionization takes place when reverse bias voltage is of the order of 100v-400v.
8. These new charge carriers in turn induces more electron hole pairs.
9. Electrons hole pairs generated by this process are difted under the influence of electric field,and diffuse outside the depletion region and finally collected at the detector terminals.
10. This leads to a flow of current in the external circuit and is proportional to the intensity of light incident on APD.
11. Quantum efficiency close to 100% can be obtained.

<u>Aperture actuators used in satellites:</u>

1. The open end of a waveguide is an example of a simple aperture antenna.
2. It is capable of radiating energy being carried by the guide, and it can receive energy from a wave impinging on it.
3. In satellite communications,the most commonly encountered aperture antennas are horn and reflector
   antennas.
4. The horn antenna is an example of an aperture antenna that provides a smooth transition from a waveguide to a larger aperture that couples more effectively into space.
5. Horn antennas are used directly as radiators aboard satellites to illuminate comparatively large areas of the earth.
6. They are also widely used as primary feeds for reflector type antennas both in transmitting and receiving modes.
7. The three most commonly used types of horns are illustrated in Fig. 6.10.

Figure 6.10 Horn antennas: (*a*) smooth-walled conical, (*b*) corrugated, and (*c*) pyramidal.

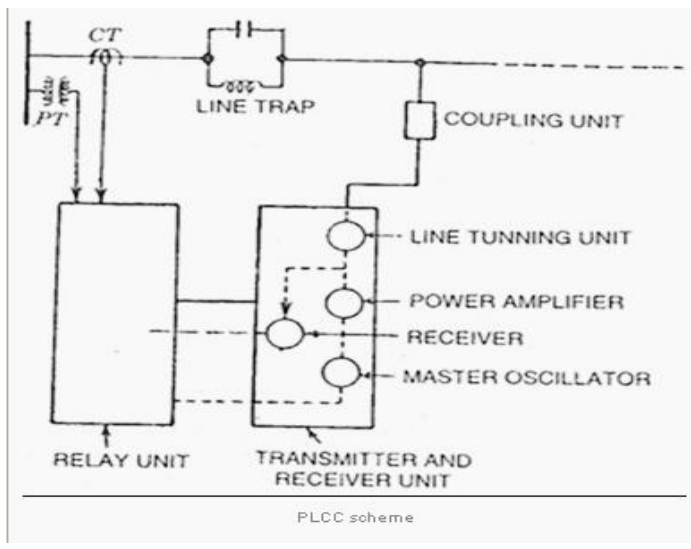
8.Parabolic reflectors are widely used in satellite communications systems to enhance the gain of antennas.

- ✓ The reflector provides a focusing mechanism which concentrates the energy in a given direction.
- ✓ The commonly used form of parabolic reflector has a circular aperture, as shown in Fig. 6.15.
- ✓ This is the type seen in many home installations for the reception of TV signals.
- ✓ The circular aperture configuration is referred to as a *paraboloidal reflector*

<u>Power line carrier communication:</u>

1. Power Line Communication (PLC) is a communication technology that enables sending data over existing power cables.
2. **Power line communication (PLCC)** are used for point to point communication over high voltage power lines.
3. **PLCC is used to transmit and receive speech and data signals by using high frequency carrier .**
4. For large power system power line carrier communication is used for data transmission as well as protection of transmission lines. Carrier current has a frequency range of ***30 to 200 kHz in USA*** and ***80 to 500 kHz in UK***.

- ***Each end of transmission line is provided with identical PLCC equipment consisting of equipment:***
  - Transmitters and Receivers
  - Line Tuners
  - Line Traps
  - Power amplif ier
  - Coupling capacitors.
- Distance protection relay in relay panel at one end of the transmission line gets the input f rom CT and CVT in line.
- The output of relay goes to modem of PLCC.
- The output of PLCC goes to coupling capacitor and then to transmission line and travels to another end where it is received through coupling capacitor and inputted to relay and control panel at that end.

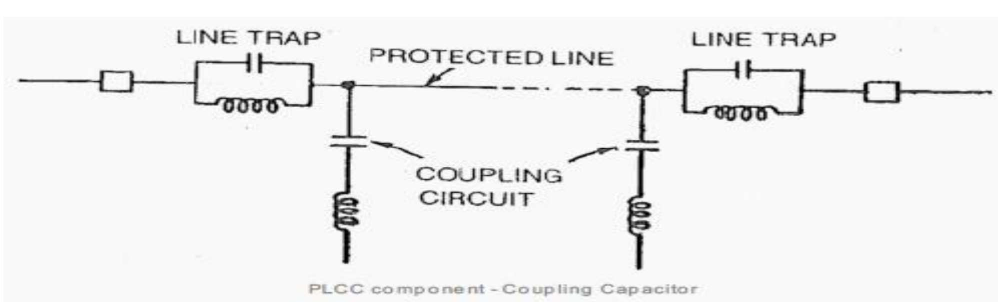


PLCC scheme

Main Components of PLCC:

Coupling Capacitor

1. Coupling capacitor connects the carrier equipment to the transmission line.
2. The coupling capacitor's capacitance is selected such that it offers low impedance to carrier f requency ***(1/ωC)*** but high impedance to power f requency (50Hz).
3. Thus coupling capacitor allows carrier f requency signal to enter the carrier equipment.
4. To decrease the impedance f urther and make the circuit purely resistive, low impedance is connected in series with coupling capacitor to form resonance at carrier f requency.



PLCC component - Coupling Capacitor

Line trap Unit

1. The line trap circuit is a parallel tuned circuit comprising of inductance (L) and capacitance (C). which is tuned to carrier frequency.
2. It has low impedance (less than 0.1) f or power f requency (50 Hz) and high impedance to carrier frequency.

3.This unit prevents the high f requency carrier signal f rom entering the neighboring lin

4.The line trap unit is used to isolate the carrier frequency from the station and directs it towards the line terminal.

5. Hence a line trap unit/Wave trap is inserted between busbar and connection of coupling capacitor to the line.

Transmitter and receiver:

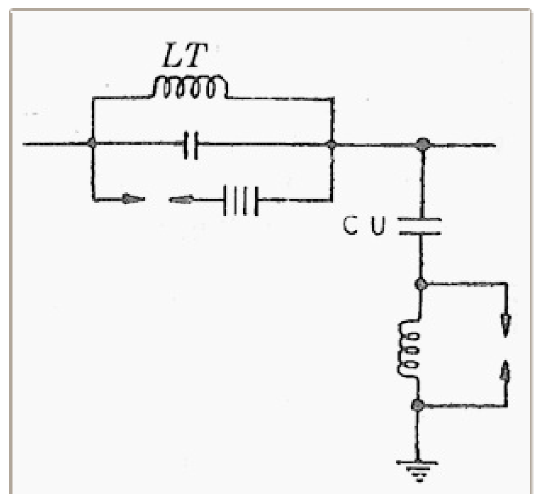
1. The carrier transmitters and receivers are usually mounted in a rack or cabinet in the control house, and the line tuner is out in the switchyard.
2. This means there is a large distance between the equipment and the tuner, and the connection between the two is made using a coaxial cable.

Hybrids and Filters:

1. The purpose of the hybrid circuits is to enable the connection of two or more transmitters together on one coaxial cable without causing intermodulation distortion .
2. Hybrids may also be required between transmitters and receivers, depending on the application.
3. The hybrid circuits causes large losses in the carrier path and must be used appropriately.
4. High/low-pass and band-pass networks may also be used, in some applications, to isolate carrier equipment f rom each other.

Line Tuners

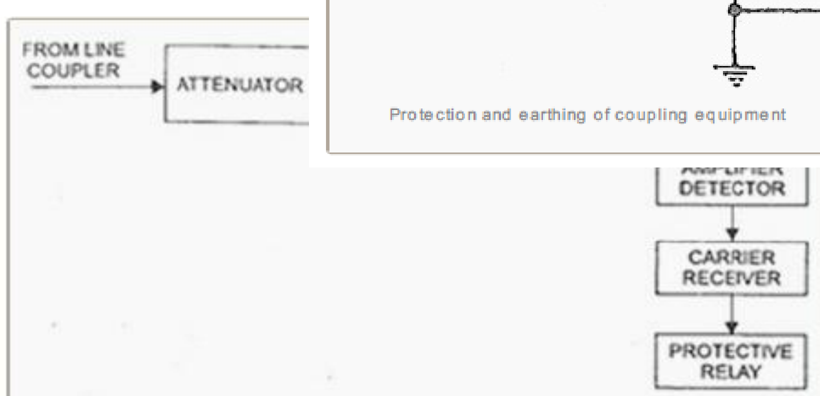
1. The purpose of the line tuner in conjunction with the coupling capacitor is
   - ➔ to provide low impedance path f or the carrier energy to the transmission line and
   - ➔ a high impedance path to the power f requency energy.
2. The line tuner/coupling capacitor combination provides a low impedance path to the power line by f orming a series resonant circuit tuned to the carrier f requency.



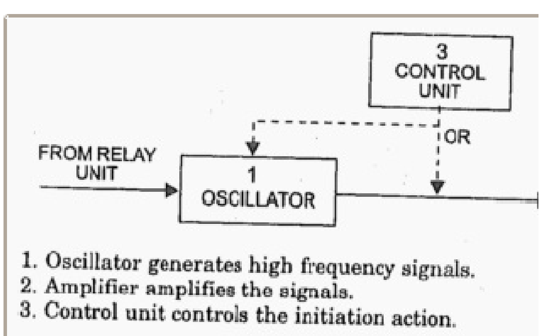
Protection and earthing of coupling equipment

Master Oscillator and Amplifiers

1. ***High frequency carrier signal is generated by a crystal oscillator.***
2. The output voltage of a oscillator is held constant by voltage stabilizer.
3. The output of oscillator is f ed to amplif ier so that loses in transmission can be compensated.



PLCC component - Transmitters and receivers



PLCC component - Master oscillator anc

Protection and earthing of coupling equipment:

1. Over voltage can be caused due to lightning, switching and sudden loss of load etc.
2. ***They produce stress on coupling equipment and line trap units.***
3. Non linear resistor in series with protective gap is connected across the line trap unit and inductor of coupling unit.
4. The gap is adjusted to spark at a set value of over voltage.
5. Coupling unit and PLCC equipment are earthed through a ***separate and dedicated system***, so that
6. ground potential rise of station earthing system does not affect the ref erence voltage level/Power supply common ground of the PLCC equipment.

Applications of PLCC

a. **Transmission & Distribution Network:** PLCC was first adopted in the electrical transmission and distribution system to transmit information at a fast rate.

b. **Home control and Automation:** PLCC technology is used in home control and automation. This technology can reduce the resources as well as efforts for activities like power management, energy conservation, etc.

c. **Entertainment:** PLCC is used to distribute the multimedia content throughout the home.

d. **Telecommunication:** Data transmission for different types of communications like telephonic communication, audio, video communication can be made with the use of PLCC technology.

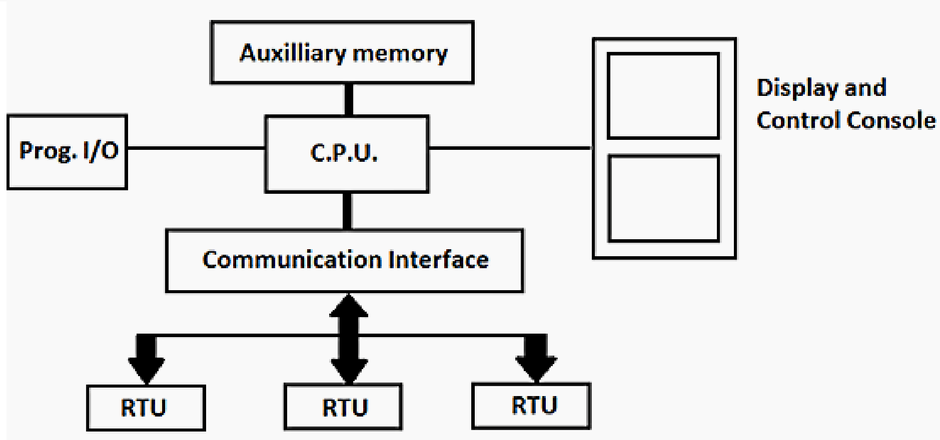
e. **Security Systems:** In monitoring houses or businesses through surveillance cameras, PLCC technology is far useful.

f. **Automatic Meter Reading** – Automatic Meter reading applications use the PLCC technology to send the data from home meters to Host Central Station.

SCADA:

1. ***SCADA stands for supervisory control and data acquisition.***
2. ***SCADA*** monitors and controls multiple sites and the system that are located at remote areas .
3. It is an industrial control system where a computer system monitors and controls a process in industries such as telecommunications, water and waste control, energy, oil and gas refining and transportation.

COMPONENTS OF SCADA:



A simple SCADA system with single computer

1. Human Machine Interface (HMI)

It is an interface which presents ***process data to a human operator***, and through this, the human operator monitors and controls the process.

2. Supervisory (computer) system

It gathers data on the process and sending commands (*or control*) to the process.

3. Remote Terminal Units (RTUs)

It connect to sensors in the process, converting sensor signals to digital data and sending digital data to the supervisory system.

4. Programmable Logic Controller (PLCs)

It is used as field devices because they are more economical, versatile, flexible, and configurable than special-purpose RTUs.

5. Communication infrastructure

It provides connectivity to the supervisory system to the Remote Terminal Units.

- The SCADA system may allow operators to change the set points for the flow, and enable alarm conditions, such as loss of flow and high temperature, to be displayed and recorded.
- The feedback control loop passes through the RTU or PLC, while the SCADA system monitors the overall performance of the loop.
- The Indian National Satellite (INSAT) system which are placed in Geo-stationary orbits is one of the largest domestic communication satellite systems in Asia-Pacific region. Established in 1983 with commissioning of INSAT-1B, it initiated a major revolution in India's communications sector and sustained the same later. INSAT space segment consists of 24 satellites out of which 10 are in service (INSAT-3A, INSAT-4B, INSAT-3C, INSAT-3E, KALPANA-1, INSAT-4A, INSAT-4CR,GSAT-8, GSAT-12 and GSAT-10)

  The system with a total of 168 transponders in the C, Extended C and Ku-bands provides services to telecommunications, television broadcasting, weather forecasting, disaster warning and Search and Rescue operations.

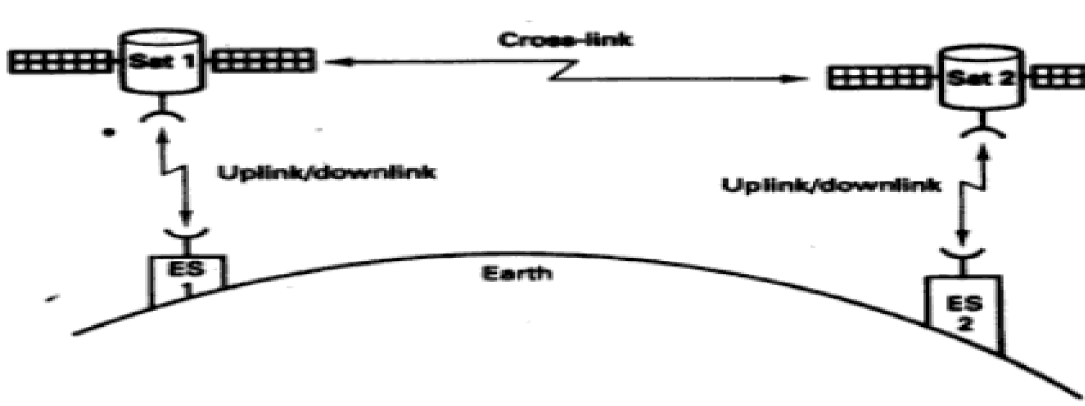


FIGURE 25-22 Intersatellite link

A **very small aperture terminal** (**VSAT**), is a two-way satellite ground station or a stabilized maritime Vsat antenna with a dish antenna that is smaller than 3 meters. The majority of VSAT antennas range from 75 cm to 1.2 m. Data rates typically range from 56 kbit/s up to 4 Mbit/s. VSATs access satellite(s) in geosynchronous orbit to relay data from small remote earth stations (terminals) to other terminals (in mesh topology) or master earth station "hubs" (in star topology).

VSATs are most commonly used to transmit narrowband data (point of sale transactions such as credit card, polling or RFID data; or SCADA), or broadband data (for the provision ofsatellite Internet access to remote locations, VoIP or video). VSATs are also used for transportable, on-the-move (utilising phased array antennas) or mobile maritimecommunications.