SIDDARTH INSTITUTE OF ENGINEERING & TECHNOLOGY:: PUTTUR

(AUTONOMOUS)

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QUESTION BANK (DESCRIPTIVE)

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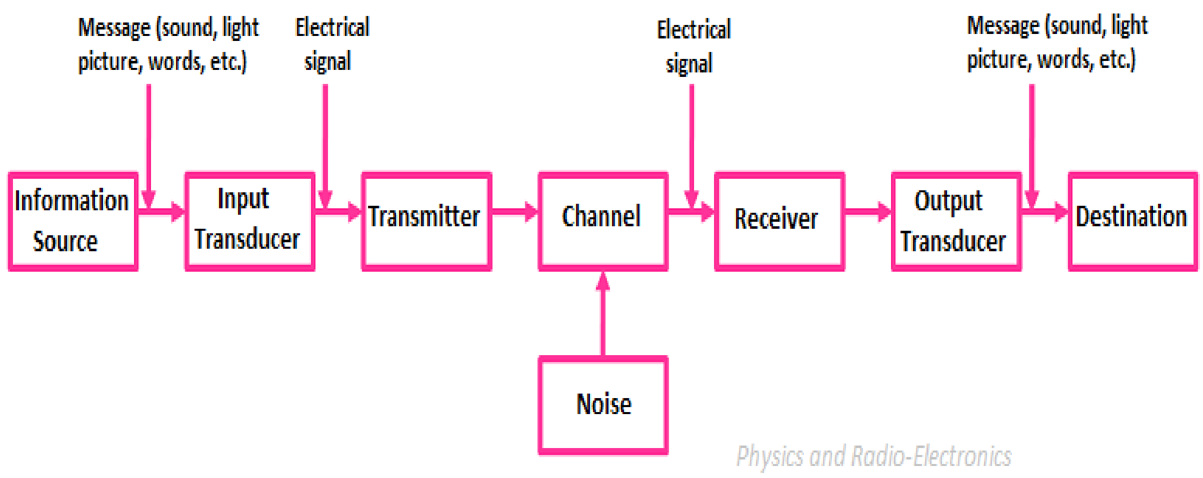
UNIT –I
INTRODUCTION TO COMMUNICATION SYSTEMS

1 a) Define Communication and draw the basic block diagram of communication system.

Communication is the process of establishing a link/ connection between two points (Transmitter, Receiver) for information exchange Via a channel.

(or)

Communication is a process of conveying/ exchanging information.



1 b) Explain the function of each block of communication system.

Information Source:

we know that the communication system serves to communicate a message (or) information source. This message/ information originates is the information source. In general there can be various messages in the form of words, group of words codes, symbols, sound signal etc.

In Short form, we can say that the function of information source is to produce required message which has to be transmitted.

Input Transducer:

A transducer is a device which converts one form of energy or signal into another form of energy or signal. The message form the source may not be an electrical signal.so there is need to convert input message signal into electrical signal (Analog (or) Digital).
Eg:- A microphone converts message signal which is in the form of sound waves into electrical form.

Transmitter:

The main function of the transmitter is to process the electrical signal from different aspects for example in radio board casting the electrical signal obtained from sound signal is processed to restrict its range of audio frequencies (upto 5KHz) in amplitude modulation radio board cast and is often amplified. In wire telephone no real processing is needed. However in long distance radio communication (or) board cast signal amplification is necessary before modulation.

Modulation is the process the main function of the transmitter. In modulation, the message signal is super imposed upon the high frequency carrier signal.

Communication Channel:

The communication channel is a medium through which the signal travels.

or

The communication channel is a wired or wireless medium through which the signal (information) travels from source (transmitter) to destination (receiver).
Communication channels are divided into two categories: wired and wireless. Some examples of wired channels include co-axial cables, fiber optic cables, and twisted pair telephone lines. Examples of wireless channels are air, water, and vacuum.

Noise:

Noise is an unwanted signal that enters the communication system via the communication channel and interferes with the transmitted signal. The noise signal (unwanted signal) degrades the transmitted signal (signal containing information).

Receiver:

The main function of a receiver is to receive signal from the communication channel. This received signal is the distorted version with noise included into it. The original signal can reproduced by a process called Demodulation.

Output Transducer:

A transducer is a device which converts one form of energy or signal into another form of energy or signal**.** An output transducer is used to convert electrical signal into message signal. Eg: - A loud speaker converts electrical signal back to sound signal

Destination:

Destination is the final stage which is used to convert an electrical message signal into its original form by using transducer. In radio board casting the destination is a loud speaker. In destination the actual information will be received.

2 a) Define wired communication and wireless communication.

Wired Communication:

A wired network uses cables to connect devices such as laptop or desktop computers to the internet or another network. A wired network has some dis advantages when compared to a wireless network. The biggest advantage is that your device tethered to a router. The most common wired networks use cables connected at one end to an Ethernet port on the network router and at the other end to a computer or other device.

Wireless Communication:

A wireless network allows devices to stay connected to the network but room untethered to any wires. Access points amplify Wi-Fi signals, so a device can be far from a router but still be connected to the network. When your connection to a Wi-Fi, hot spot at a café, a hotel, an airport lounge or another public place, you are connecting to that business wireless network.

2 b) Compare Analog and Digital communication.

Analog Communication	**Digital Communication**
1. If the message signal is analog.	1. If the message signal is digital then it is known as digital Modulation.
2. Get affected by noise and distortion.	2. Immune to Noise and Distortion.
3. Low Bandwidth requirement.	3. High Bandwidth requirement.
4. Hardware is complicated and less flexible than digital system.	4. Hardware is flexible and less complicated than Analog system.
5. High Power is required.	5. Low Power is required.
6. Error Probability is High.	6. Error Probability is Low.
7. It consists of continuous values	7. It consists of discrete values.
8. Analog signal can be represented by sine wave	8. Digital signal can be represented by square wave.

3 a) Define modulation. Classify different types of modulation.

Modulation:

Modulation is defined as changing the characteristics of a carrier signal in accordance with the instantaneous values of the another signal called message signal/ modulating signal. Signals containing information are referred as modulating signals. This information bearing signals is also called base band signal. The high frequency signal is known as carrier signal. The signal resulting from the process of modulation is called modulated signal.

Types of Modulation:

There are three types of modulation

1. Amplitude Modulation
2. Frequency Modulation
3. Phase Modulation

Amplitude Modulation:

Amplitude Modulation is defined as the modulation in which the amplitude of the carrier wave is varied in accordance with the instantaneous amplitude of the modulating signal, keeping its carrier frequency and phase constant.

Frequency Modulation:

Frequency Modulation is defined as the modulation in which the frequency of the carrier wave is varied in accordance with the instantaneous amplitude of the modulating signal, keeping its carrier amplitude and phase constant.

Phase Modulation:

Phase Modulation is defined as the modulation in which the Phase of the carrier wave is varied in accordance with the instantaneous amplitude of the modulating signal, keeping its carrier amplitude and phase constant.

3 b) Explain the need for Modulation.

Need of Modulation:

1. **Reduces the height of Antenna:**

Height of antenna is a function of wave length λ. The minimum height of antenna is given by λ/4.

i.e. Height of the Antenna= λ/4=c/4f

where λ=c/f

c=3*10^8, velocity of light.

f= Transmitting Frequency

Eg: - (i) f=15KHz

Height of the Antenna= λ/4=c/4f= 3*10^8/4*15*10^3= 5000 meters.

Eg: - (i) f=1 MHz

Height of the Antenna= λ/4=c/4f= 3*10^8/4*1*10^6= 7 meters.

For the above two examples it is clear that as the transmitting frequency is increased, height of the antenna is decreased.

2. **Avoid missing of signals:**

All audio (message) signals ranges from 20Hz to 20 KHz. The transmission of message signals from various sources causes that missing of signals and then it is difficult to separate these signals at the receiver end.

3. **Increases the range of communication:**

Low frequency signals have poor radiation and they get highly attenuated. Therefore baseband signals cannot be transmitted directly over long distance. Modulation increases the frequency of the signal and this they can be transmitted over long distance.

4. **Allows adjustments in the Bandwidth:**

Bandwidth of a modulated signal may be mode smaller or larger.

5. **Improves quality of reception:**

Modulation techniques like frequency modulation, pulse code modulation reduces the effect of noise to great extent. Reduction of noise improves quality of reception.

4 a) Define Amplitude Modulation. Derive expression for AM wave.

Amplitude Modulation:

Amplitude Modulation is defined as the modulation in which the amplitude of the carrier wave is varied in accordance with the instantaneous amplitude of the modulating signal, keeping its carrier frequency and phase constant.

Expression for Amplitude Modulation wave:

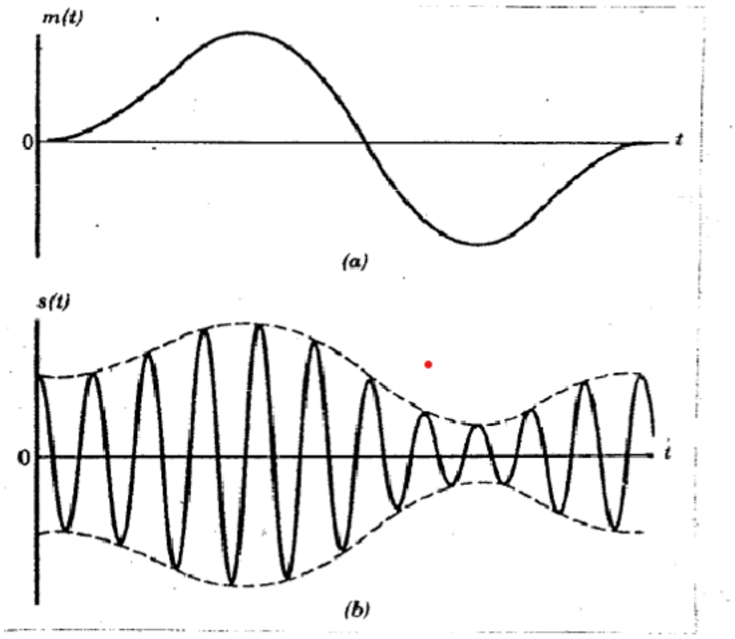


Fig (a) Message Signal, Fig (b) AM wave S(t)

* The Instantaneous value of modulating Signal is given by

$$\boxed{m(t) = A_m \cos(2\pi f_m t)} \longrightarrow (1)$$

Where, $A_m \rightarrow$ maximum amplitude of the modulating Signal

$f_m \rightarrow$ Frequency of modulating Signal.

* The Instantaneous value of carrier Signal is given by

$$\boxed{C(t) = A_c \cos(2\pi f_c t)} \longrightarrow (2)$$

Where,

$A_c \rightarrow$ Maximum amplitude of the carrier Signal.

$f_c \rightarrow$ Frequency of carrier Signal.

The Standard equation for AM wave is given by

$$\boxed{S(t) = A_c\left[1+K_a m(t)\right]\cos(2\pi f_c t)} \longrightarrow (3)$$

Where,

K_a is a Constant called the amplitude Sensitivity of the modulator.

Substituting eq (1) in eq (3), we get

$$S(t) = A_c\left[1+K_a A_m\cos(2\pi f_m t)\right]\cos(2\pi f_c t)$$

$$S(t) = A_c\left[1+\mu\cos(2\pi f_m t)\right]\cos(2\pi f_c t)$$

Where, $\boxed{\mu = K_a A_m}$ is called the modulation Index or modulation factor.

$$S(t) = A_c\cos(2\pi f_c t) + \mu A_c\cos(2\pi f_c t)\cdot\cos(2\pi f_m t) \longrightarrow (4)$$

$$S(t) = A_c\cos(2\pi f_c t) + \mu A_c\cos(2\pi f_c t)\cdot\cos(2\pi f_m t) \longrightarrow (4)$$

equation (4) can be further expanded, by means of the trigonometric -al relation :

$$\boxed{\cos a\cdot\cos b = \frac{1}{2}\cos(a-b) + \frac{1}{2}\cos(a+b)}$$

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

equation (5) is the amplitude modulated Signal, consist of three frequency component

1) The first term is the carrier itself. It has a frequency 'f_c' and amplitude 'A_c'.

2) The 2nd component is $\frac{\mu A_c}{2} \cos 2\pi(f_c - f_m)t$. It has frequency $(f_c - f_m)$ called Lower Sideband (F_{LSB}) and having amplitude $\frac{\mu A_c}{2}$

3) Similarly 3rd component is $\frac{\mu A_c}{2} \cos 2\pi(f_c + f_m)t$. It has frequency $(f_c + f_m)$ called upper Sideband (F_{USB}) and having amplitude $\frac{\mu A_c}{2}$.

4 b) Determine the modulation index of AM, Percentage Modulation and Bandwidth of AM.

Modulation Index:

The ratio of change in amplitude of modulating signal to the amplitude of carrier wave is known as modulation index (or) modulation factor (or) modulation co-efficient (or) depth of modulation (or) degree of modulation 'M'.

$$\boxed{\mu = \frac{A_m}{A_c}} \quad \text{or} \quad \boxed{\mu = K_a A_m}$$

Percentage Modulation Index:

$$\boxed{\%\,\mu = \left(\frac{A_m}{A_c}\right) \times 100}$$

Note:

1. If A_m is greater than A_c then distortion is introduced in to the system.
2. The modulating signal voltage 'A_m' must be less than carrier signal voltage 'A_c' for proper amplitude modification.

Transmission Bandwidth (B_T):

The difference between upper side band and lower side band frequencies defines the transmission bandwidth 'B_T'.

$$B_T = f_{USB} - f_{LSB}$$
$$= (f_c + f_m) - (f_c - f_m)$$
$$= \cancel{f_c} + f_m - \cancel{f_c} + f_m$$
$$\boxed{B_T = 2f_m}$$

Therefore, Bandwidth requires for transmission of an Amplitude Modulation wave is twice the modulating signal frequency that is $2f_m$

5 a) Explain shortly about i) Sidebands ii) Justify the reason for selecting the DSB-SC over DSB FC.

I) **Sidebands:**

In electronic signal transmission, a sideband is the portion of a modulated carrier wave that is either above or below the basic (Baseband) signal. The portion above the baseband signal is the upper sideband; the portion below is the lower sideband. In regular amplitude modulation (AM) transmission, both sidebands are used to carry a message. In some forms of transmission, one sideband is removed (single-sideband transmission) or a portion of one sideband is removed.

There are two sidebands in the Amplitude Modulated wave. One is the Upper Side Band which is termed in-short as USB and another is the Lower Side Band, also called LSB.

The frequency of the Upper Side Band is given by,

f_{USB} = fc + fm and

The frequency of the Lower Side Band is given by,

f_{LSB} = fc - fm

II) DSB-SC and DSB-SC are the types of Amplitude Modulation Schemes. DSB-SC is an acronym for Double Sideband Suppressed Carrier and DSB-FC is an acronym for Double Sideband Full Carrier.

In DSB-FC around 67% or two-thirds of the total power is wasted by the carrier.

So, in DSB-SC the carrier is suppressed, but this suppression won't affect the message signal.

Therefore, the DSB-SC is selected over DSB-FC.

5 b) A modulating signal 10 cos $(2\pi \times 10^3 t)$ is used to modulate a carrier signal 20 cos $(2\pi \times 10^4 t)$. Compute the modulation index, % of modulation index, frequency of sideband components and their amplitudes. What will be the bandwidth of modulated signal?

Modulating Signal V_m=10 cos $(2\pi \times 10^3 t)$

Carrier Signal V_c= 20 cos $(2\pi \times 10^4 t)$

$V_m = 10$ volt $f_m = 1 \times 10^3$ HZ $= 1$ KHZ

Carrier signal $V_c = 20 \sin(2\pi \times 10^4 t)$

$v_c = V_c \sin(2\pi f_c t)$

$V_c = 20$ volt $f_c = 1 \times 10^4$ HZ $= 10$ KHZ

(a) Modulation index

$$m = \frac{V_m}{V_c} = \frac{10}{20} = 0.5$$

(b) percentage of modulation

$$m = 0.5 \times 100 = 50\,\%$$

(c) frequency of sideband components

① $f_{USB} = f_c + f_m = (10+1) = 11$ KHZ

② $f_{LSB} = f_c - f_m = 10 - 1 = 9$ KHZ

(d) Amplitude of each sideband $\frac{mV_c}{2} = \frac{0.5 \times 20}{2} = 5V$

Bandwidth of modulated signal =2fm=2*1KHz=2KHz

6 a) Illustrate the Amplitude modulation for single tone information.

A Single-tone modulating Signal $m(t)$ has a Single (tone) Frequency Component 'f_m' and is defined as follows :

$$\boxed{m(t) = A_m \cos(2\pi f_m t)} \rightarrow (1)$$

Where A_m is the amplitude of the modulating wave and f_m is the frequency of the modulating wave.

Let $\boxed{c(t) = A_c \cos(2\pi f_c t)} \rightarrow (2)$

Where A_c is the amplitude of the carrier wave and f_c is the frequency of the carrier wave.

* The time-domain expression for the Standard AM wave is

$$\boxed{S(t) = A_c[1 + K_a m(t)] \cos 2\pi f_c t} \rightarrow (3)$$

Substituting eq (1) in eq (3), we get

$$S(t) = A_c[1 + K_a A_m \cos 2\pi f_m t] \cos 2\pi f_c t$$

Since, the modulation Index $\boxed{\mu = K_a A_m}$

We get

$$S(t) = A_c[1 + \mu \cos 2\pi f_m t] \cos 2\pi f_c t \rightarrow (4)$$

equation (4) Can be further expanded, by means of the trigonometric -al relation

$$\boxed{\cos a \cdot \cos b = \frac{1}{2}\cos(a-b) + \frac{1}{2}\cos(a+b)}$$

$$s(t) = A_c \cos(2\pi f_c t) + \mu A_c \underbrace{\cos(2\pi f_c t)}_{\cos a} \cdot \underbrace{\cos(2\pi f_m t)}_{\cos b}$$

$$s(t) = A_c \cos(2\pi f_c t) + \frac{\mu A_c}{2} \cos[2\pi f_c - 2\pi f_m]t + \frac{\mu A_c}{2} \cos[2\pi f_c + 2\pi f_m]t \rightarrow (5)$$

Taking Fourier transform on both sides of eq (5), we get

$$\boxed{S(f) = \frac{A_c}{2}[\delta(f-f_c) + \delta(f+f_c)] + \frac{\mu A_c}{4}\{\delta[f-(f_c-f_m)] + \delta[f+(f_c-f_m)]\} + \frac{\mu A_c}{4}\{\delta[f-(f_c+f_m)] + \delta[f+(f_c+f_m)]\}}$$

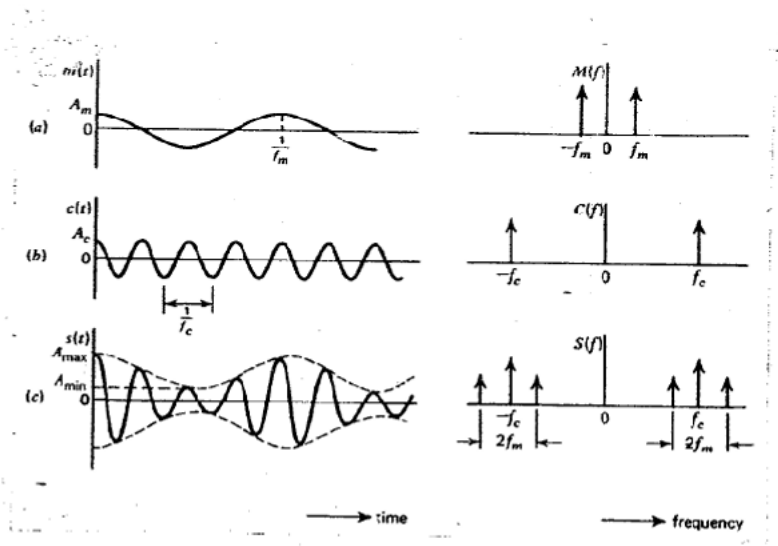


Fig (1) Illustrating the time-domain (on the left) and frequency domain (on the right) characteristics of a standard amplitude modulation produced by a single tone.
(a) Modulating wave (b) Carrier wave (c) AM wave..

6 b) Discuss the advantages and disadvantages of DSB-SC.

Advantages of SSB-SC:

1. SSB required half the bandwidth required of Amplitude wave and DSB-SC signals.
2. Due to suppression of carrier and one side band power is saved.
3. Reduced interference of noise. This is due to the reduced bandwidth as the bandwidth increases the amount of noise added to the signal with increase.
4. Fading does not occur in SSB transmission.

5. Fading means that a signal alternating increases and decreases in strength as it is picked up by the receiver.
6. It occurs because the carrier and sideband may reach the receiver shifted in time and phase with respect to each other.

Disadvantages of SSB-SC:

1. The generation and reception of SSB signal is a complex process.
2. Since carrier is absent the SSB transmitter and receiver need to have an excellent frequency stability.
3. The SSB modulation is expensive and highly complex to implement.

7 a) What is DSB-SC Modulation? Explain the Time and Frequency domain expressions of DSB-SC wave.

To overcome the drawback of power voltage is AM wave (DSB-FC) an DSB-SC method is used.

DSB-SC is a method of transmission where only the two sidebands are transmitted without the carrier (suppressing carrier)

(or)

The conventional AM wave in which the carrier is suppressed is called DSB-SC Modulation.

Time domain Representation of DSB-SC wave:

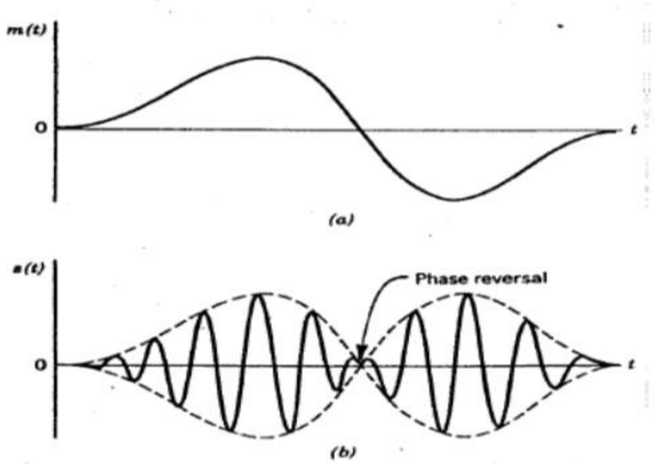


Figure
(a) Message signal. (b) DSBSC-modulated wave s(t).

* Let $m(t)$ be the message Signal having a bandwidth equal to 'W' Hz and
$c(t) = A_c \cos 2\pi f_c t$ represents the carrier, then the time-domain expression for DSB-SC wave is

$$S(t) = m(t)\,c(t)$$

$$\boxed{S(t) = A_c \cos(2\pi f_c t)\, m(t)} \longrightarrow (1)$$

* The $S(t)$ Signal undergoes a phase reversal whenever the message Signal crosses Zero.

Frequency-Domain Description :-

Taking Fourier transform on both Sides of equation (2), we get

$$\boxed{S(f) = \frac{A_c}{2}\left[M(f-f_c) + M(f+f_c)\right]} \longrightarrow (3)$$

Where $S(f)$ is the Fourier transform of the modulated wave $S(t)$

$M(f)$ is the Fourier transform of the message Signal $m(t)$.

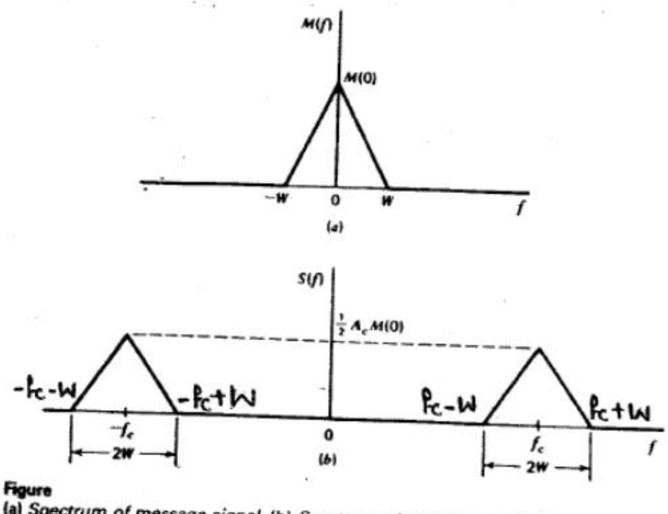


Figure
(a) Spectrum of message signal. (b) Spectrum of DSBSC modulated wave.

The amplitude spectrum drawn above exhibits the following factor

1. on either sides of $+f_c$ or $- f_c$, we have two sidebands designated as lower and upper sideband.
2. The impulse are absent at $+f_c$ or $- f_c$ in the amplitude spectrum signifying the fact that the carrier form is suppressed in the transmitted wave.

3. The minimum transmission bandwidth required is 2W that is twice the message bandwidth.

7 b) Define demodulation. Explain any one amplitude demodulation technique.

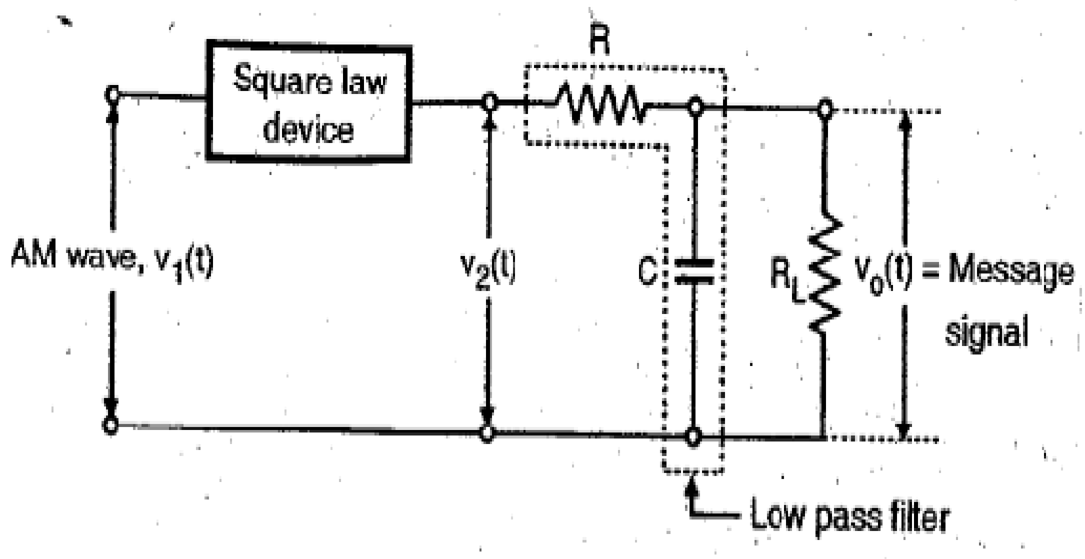
Demodulation:

Demodulation or detector is the process of recovering the original message signal from the modulated wave at the receiver. Demodulation is the reverse of the modulation process.

There are two types of detectors

1. Square law detector
2. Envelope detector

Square Law Detector:



A square law detector

* A Square-law detector is essentially obtained by using a Square-law modulator for the purpose of detection.
* An AM Signal can be demodulated by Squaring it and then passing the Squared Signal through a Low pass filter (LPF)

The transfer characteristics of a non-linear device is given by:

$$\boxed{V_2(t) = a_1 V_1(t) + a_2 V_1^2(t)} \longrightarrow (1)$$

Where,

$V_1(t) \rightarrow$ I/p voltage

$V_2(t) \rightarrow$ O/p voltage

a_1 and $a_2 \rightarrow$ are Constants.

* The I/p voltage of the AM wave is given by

$$\boxed{V_1(t) = A_c[1 + K_a m(t)] \cos 2\pi f_c t} \longrightarrow (2)$$

Substituting equation (2) in equation (1), we get

$$V_2(t) = a_1\left\{A_c[1+K_a m(t)]\cos 2\pi f_c t\right\} + a_2\left\{A_c[1+K_a m(t)]\cos 2\pi f_c t\right\}^2$$

$$V_2(t) = a_1 A_c[1+K_a m(t)]\cos 2\pi f_c t + a_2\left\{A_c^2\,[1+K_a m(t)]^2 \cos^2 2\pi f_c t\right\}$$

W.K.T

$$\boxed{(a+b)^2 = a^2 + b^2 + 2ab} \quad \text{and} \quad \boxed{\cos^2\theta = \frac{1+\cos 2\theta}{2}}$$

$$V_2(t) = a_1 A_c[1+K_a m(t)]\cos 2\pi f_c t + a_2 A_c^2 \cos^2 2\pi f_c t\left[1^2 + K_a^2 m^2(t) + 2K_a m(t)\right]$$

$$V_2(t) = a_1 A_c[1+K_a m(t)]\cos 2\pi f_c t + a_2 A_c^2\left[\frac{1+\cos 2(2\pi f_c t)}{2}\right]\left[1 + K_a^2 m^2(t) + 2K_a m(t)\right]$$

$$V_2(t) = a_1 A_c[1+K_a m(t)]\cos 2\pi f_c t + \frac{a_2 A_c^2}{2}\left[1 + K_a^2 m^2(t) + 2K_a m(t)\right](1+\cos 4\pi f_c t) \rightarrow ($$

* In eq ③ $\frac{a_2 A_c^2}{\cancel{2}} \cancel{2} K_a m(t)$ is the desired term which is due to the $a_2 V_1^2$ term. Hence the name of this detector is - Square law detector.

* The desired term is extracted by using a LPF (Fig.). Thus the o/p of LPF is $\boxed{V_0(t) = a_2 A_c^2 K_a m(t)}$

Thus the message signal $m(t)$ is recovered at the o/p of the message signal.

8 a) Explain single tone modulation for transmitting only upper side band (USB) frequency of SSB modulation.

Upper side band (USB) frequency of SSB Modulation:

* Let the modulating signal $m(t)$ is represented as

$$\boxed{m(t) = A_m \cos(2\pi f_m t)} \longrightarrow ①$$

* The hilbert transform of the modulating signal $m(t)$ is obtained by passing it through a -90° phase shifter. So the hilbert transform is given by

$$\boxed{\hat{m}(t) = A_m \sin(2\pi f_c t)} \longrightarrow ②$$

* WKT the SSB wave with only USB is given by

$$S_u(t) = \frac{A_c}{2}\left[m(t)\cos(2\pi f_c t) - \hat{m}(t)\sin(2\pi f_c t)\right] \longrightarrow ③$$

Substituting eq ① & eq ② in eq ③, we get

$$S_u(t) = \frac{A_c}{2}\left[A_m \cos(2\pi f_m t) \cdot \cos(2\pi f_c t) - A_m \sin(2\pi f_m t) \cdot \sin(2\pi f_c t)\right]$$

W.K.T

$$\boxed{\cos(A+B) = \cos(A) \cdot \cos(B) - \sin(A) \cdot \sin(B)}$$

$$S_u(t) = \frac{A_c A_m}{2}\Big[\underbrace{\cos(2\pi f_c t)}_{\cos(A)} \cdot \underbrace{\cos(2\pi f_m t)}_{\cos(B)} - \underbrace{\sin(2\pi f_c t)}_{\sin(A)} \cdot \underbrace{\sin(2\pi f_m t)}_{\sin(B)}\Big]$$

$$S_u(t) = \frac{A_c A_m}{2}\left[\cos(2\pi f_c t + 2\pi f_m t)\right]$$

$$\boxed{S_u(t) = \frac{A_c A_m}{2} \cos 2\pi\left[f_c + f_m\right] t} \longrightarrow ④$$

Equation ④ shows that the SSB wave consists of only the upper sideband of frequency $(f_c + f_m)$.

* This is exactly same as the result obtained by suppressing the lower side-frequency $(f_c - f_m)$ of the corresponding DSB-SC wave.

8 b) Explain briefly about the various applications of SSB-SC.

Application of SSB-SC:

1. SSB is used in the applications where the power saving is required in mobile systems.
2. SSB is also used in applications in which bandwidth requirements are low.

Eg: - Point to Point communication, Land, Air and Maritime Mobile Communications, Television, Telemetry, Military communications, Radio navigation and Amateur radio are the greatest users of SSB in one form or another.

9 a) Explain single tone modulation for transmitting only lower side band (LSB) frequency of SSB modulation.

Lower side band (LSB) frequency of SSB Modulation:

* Let the modulating signal $m(t)$ is represented as

$$\boxed{m(t) = A_m \cos(2\pi f_m t)} \longrightarrow (1)$$

* The hilbert transform of the modulating signal $m(t)$ is obtained by passing it through a -90° phase shifter. So the hilbert transform is given by:

$$\boxed{\hat{m}(t) = A_m \sin(2\pi f_m t)} \longrightarrow (2)$$

W.K.T the SSB wave with only LSB is given by:

$$\boxed{S_L(t) = \frac{A_c}{2}\left[m(t)\cos(2\pi f_c t) + \hat{m}(t)\sin(2\pi f_c t)\right]} \longrightarrow (3)$$

Substituting eq (1) & eq (2) in eq (3), we get

$$S_L(t) = \frac{A_c}{2}\left[A_m \cos(2\pi f_m t)\cos(2\pi f_c t) + A_m \sin(2\pi f_m t)\cdot \sin(2\pi f_c t)\right]$$

$$S_L(t) = \frac{A_c A_m}{2}\left[\underbrace{\cos(2\pi f_c t)}_{\cos(A)}\cdot \underbrace{\cos(2\pi f_m t)}_{\cos(B)} + \underbrace{\sin(2\pi f_c t)}_{\sin(A)}\cdot \underbrace{\sin(2\pi f_m t)}_{\sin(B)}\right]$$

W.K.T

$$\boxed{\cos(A-B) = \cos A\cdot \cos B + \sin A\cdot \sin B}$$

$$S_L(t) = \frac{A_c A_m}{2}\left[\cos(2\pi f_c t - 2\pi f_m t)\right]$$

$$\boxed{S_L(t) = \frac{A_c A_m}{2}\cos 2\pi\left[f_c - f_m\right]t} \longrightarrow (4)$$

Equation (4) shows that the SSB wave consists of only the Lower Side frequency $(f_c - f_m)$.

* This is exactly same as the result obtained by suppressing the upper side frequency $(f_c + f_m)$ of the corresponding DSB-SC wave.

9 b) What are the advantages and disadvantages of SSB-SC signal

Advantages of SSB-SC:

1. SSB required half the bandwidth required of Amplitude wave and DSB-SC signals.
2. Due to suppression of carrier and one side band power is saved.

3. Reduced interference of noise. This is due to the reduced bandwidth as the bandwidth increases the amount of noise added to the signal with increase.
4. Fading does not occur in SSB transmission.
5. Fading means that a signal alternating increases and decreases in strength as it is picked up by the receiver.
6. It occurs because the carrier and sideband may reach the receiver shifted in time and phase with respect to each other.

Disadvantages of SSB-SC:

1. The generation and reception of SSB signal is a complex process.
2. Since carrier is absent the SSB transmitter and receiver need to have an excellent frequency stability.
3. The SSB modulation is expensive and highly complex to implement.

10 a) Comparison of Amplitude modulation techniques.

Sl No	Parameter	DSB-FC Standard AM	DSB-SC	SSB	VSB
1)	Power	High	Medium	Less	Less than DSB-SC but greater than SSB
2)	Bandwidth	$2f_m$	$2f_m$	f_m	$f_m < BW < 2f_m$
3)	Carrier Suppression	No	Yes	Yes	No
4)	Sideband Transmission	No	No	one Sideband Completely	one Sideband Suppressed partly
5)	Transmission efficiency	Minimum	Moderate	Maximum	Moderate
6)	Receiver Complexity	Simple	Complex	Complex	Simple
7)	Modulation type	Non-linear	Linear	Linear	Linear
8)	Applications	Radio Communication	Linear Radio Communication	Linear point to point mobile Communication	Television

10 b) List the advantages and disadvantages of Double side-band Full carrier.

Advantages of DSB-FC:

1) AM Tx'ers are less Complex
2) AM receivers are Simple, detection is easy.
3) AM receivers are Cost efficient.
4) AM waves can travel a longer bandwidth.
5) Low bandwidth.

Disadvantages of DSB-FC:

1) power is wasted in the transmitted Signal.
2) AM needs larger bandwidth
3) AM waves gets affected due to noise.

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QUESTION BANK (DESCRIPTIVE)

Subject with Code: **ICS(20EC0451)** **Course& Branch**: B.Tech & CSE,CSM.CIC

Year & Sem: III-B.Tech.& I-Sem. **Regulation:** R20

UNIT- II
Angle Modulation & Demodulation

1 a) Define angle modulation. Classify different types of angle modulation and advantages of Angle modulation.

Angle Modulation:

Angle Modulation is the process in which either the frequency or phase of the carrier wave is varied in accordance with the instantaneous amplitude of the message signal is termed as Angle Modulation.

Types of Angle Modulation:

Angle Modulation is classified into two types. They are

1. Frequency Modulation
2. Phase Modulation

Frequency Modulation:

The modulation in which frequency of carrier wave is varied in accordance with the instantaneous amplitude of the message signal is known as frequency modulation.

Phase Modulation:

The modulation in which Phase of carrier wave is varied in accordance with the instantaneous amplitude of the message signal is known as phase modulation.

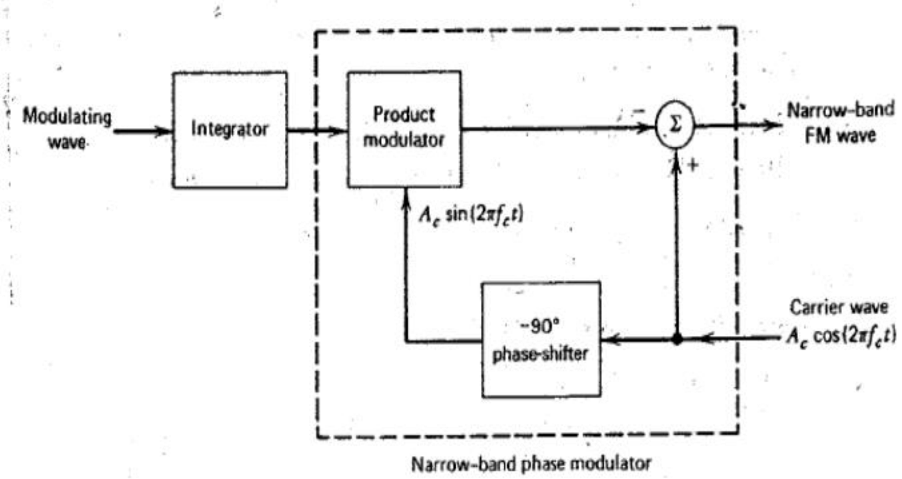
Advantages of Angle Modulation:

1. It is used in Noise Reduction.
2. It is used in Improved system fidelity.
3. It is used in efficient use of power.
4. It is used in wide bandwidth.
5. It is used in high system performance.

1 b) Analyze the expression of single tone NBFM.

Narrow Band FM:

1. A Narrow Band FM is the FM with a small bandwidth. The Modulation Index 'β' of NBFM is small as compared to one radian.
2. The NBFM has a Narrow Bandwidth which is equal to twice the message bandwidth.



Block diagram of a method for generating a narrow-band FM signal.

The time-domain expression for an FM Wave is

$$S(t) = A_c \cos\left[2\pi f_c t + \beta \sin(2\pi f_m t)\right] \longrightarrow ①$$

using the trignometric identity

$$\boxed{\cos(A+B) = \cos A . \cos B - \sin A . \sin B}$$

$$A = 2\pi f_c t \quad \& \quad B = \beta \sin(2\pi f_m t)$$

$$S(t) = A_c\left[\cos(2\pi f_c t) . \cos(\beta \sin 2\pi f_m t) - \sin(2\pi f_c t) . \sin(\beta \sin 2\pi f_m t)\right] \rightarrow ②$$

In NBFM, β is Small, hence it possible to approximate

$$\boxed{\begin{array}{l} \cos(\beta . \sin 2\pi f_m t) \approx 1 \\ \sin(\beta \sin 2\pi f_m t) \approx \beta \sin 2\pi f_m t \end{array}} \longrightarrow ③$$

Substituting eq ③ in eq ②, we get

$$S(t) = A_c \cos 2\pi f_c t \,(1) - A_c \sin 2\pi f_c t \cdot (\beta \sin 2\pi f_m t)$$

$$S(t) = A_c \cos 2\pi f_c t - \beta A_c \cdot \underline{\sin 2\pi f_c t \cdot \sin 2\pi f_m t} \rightarrow ④$$

W.K.T

$$\boxed{\sin A \cdot \sin B = \frac{1}{2}\cos(A-B) - \frac{1}{2}\cos(A+B)}$$

$$S(t) = A_c \cos 2\pi f_c t - \left[\frac{\beta A_c}{2}\cos 2\pi(f_c - f_m)t - \frac{\beta A_c}{2}\cos 2\pi(f_c + f_m)t\right]$$

$$S(t) = A_c \cos 2\pi f_c t - \frac{\beta A_c}{2}\cos 2\pi(f_c - f_m)t + \frac{\beta A_c}{2}\cos 2\pi(f_c + f_m)t \rightarrow ⑤$$

W.K.T the amplitude modulated wave is given by

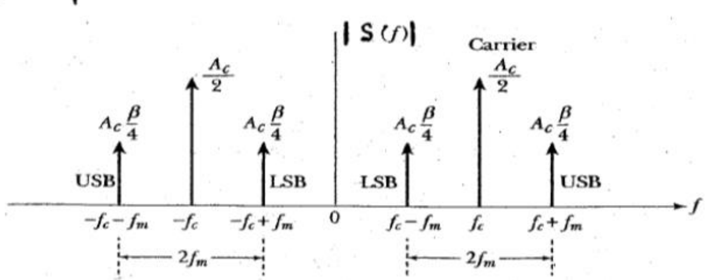
$$S(t) = A_c \cos 2\pi f_c t + \frac{\mu A_c}{2}\cos 2\pi(f_c - f_m)t + \frac{\mu A_c}{2}\cos 2\pi(f_c + f_m)t \rightarrow ⑥$$

* Comparing equation ⑤ & equation ⑥. The only difference observed between NBFM wave & AM wave is the sign reversal of the lower sideband.
* Thus NBFM requires the same bandwidth as that of AM.

Taking Fourier Transform on both sides of eq ⑤, we get

$$S(f) = \frac{A_c}{2}\left[\delta(f - f_c) + \delta(f + f_c)\right] - \frac{\beta A_c}{4}\left\{\delta[f - (f_c - f_m)] + \delta[f + (f_c - f_m)]\right\}$$
$$+ \frac{\beta A_c}{4}\left\{\delta[f - (f_c + f_m)] + \delta[f + (f_c + f_m)]\right\}$$

* The transmission bandwidth of a NBFM wave is $2f_m$.
* The NBFM wave & Conventional AM wave are identical but there is No amplitude variation in FM.

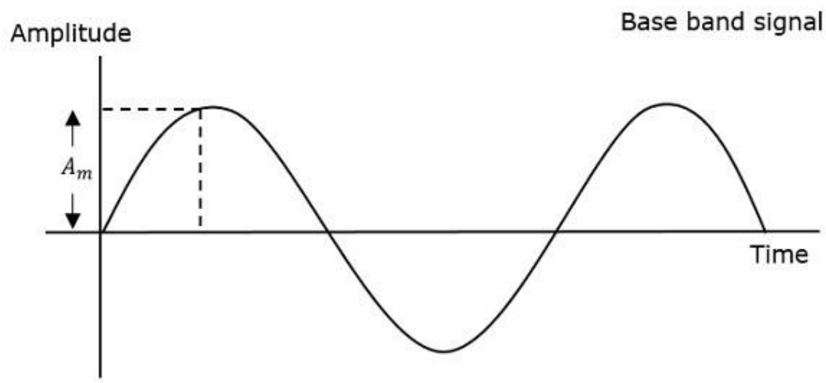


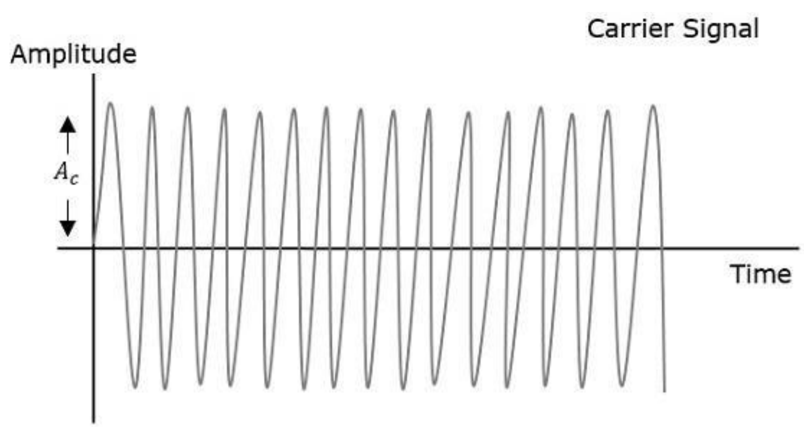
Spectral content of a NBFM wave for single-tone modulation.

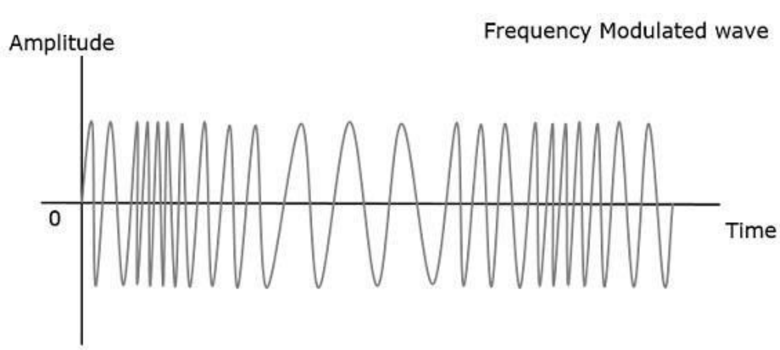
2 a) Define Frequency Modulation with necessary waveforms.

Frequency Modulation:

The modulation in which frequency of carrier wave is varied in accordance with the instantaneous amplitude of the message signal is known as frequency modulation.







Mathematical Representation

Let the carrier frequency be $\mathbf{f_c}$

The frequency at maximum amplitude of the message signal = $\mathbf{f_c} + \mathbf{\Delta f}$

The frequency at minimum amplitude of the message signal = $\mathbf{f_c - \Delta f}$

The difference between FM modulated frequency and normal frequency is termed as **Frequency Deviation** and is denoted by **Δf**.

The deviation of the frequency of the carrier signal from high to low or low to high can be termed as the **Carrier Swing**.

Carrier Swing = 2 × frequency deviation

= 2 × Δf

Equation for FM WAVE

The equation for FM wave is

$s(t) = A_c \cos [2\pi f_c t + \beta \sin(2\pi f_m t)]$

Where,

$\mathbf{A_c}$ = the amplitude of the carrier

$\mathbf{f_c}$ = frequency of the carrier

m(t) = message signal

FM can be divided into **Narrowband FM** and **Wideband FM**.

2 b) Derive the expression of Frequency modulation.

* The frequency modulated wave in time domain is given by:

$$\boxed{S(t) = A_c \cos[\theta(t)]} \longrightarrow ①$$

* The Sinusoidal modulating Signal is defined by

$$\boxed{m(t) = A_m \cos(2\pi f_m t)} \longrightarrow ②$$

* The Instantaneous frequency of the FM Signal is given by:

$$f_i(t) = f_c + K_f \underline{m(t)}$$

$$f_i(t) = f_c + \underline{K_f A_m} \cos(2\pi f_m t)$$

$$\boxed{f_i(t) = f_c + \Delta f \cos(2\pi f_m t)} \longrightarrow ③$$

Where, $\Delta f = K_f A_m$ & it is called as frequency deviation

{ The quantity Δf is called the frequency deviation. The frequency deviation Δf is proportional to the amplitude of the modulating signal & is Independent of the modulating frequency.
}

W.K.T the angular velocity $\omega_i(t)$ is the rate of change of $\theta(t)$.

$$\omega_i(t) = \frac{d}{dt}\theta(t)$$

$$2\pi f_i(t) = \frac{d}{dt}\theta(t) \longrightarrow (4)$$

Integrating eq (4) w.r.to dt

$$\int_0^t \frac{d}{\cancel{dt}}\theta(t)\cancel{dt} = \int_0^t 2\pi f_i(t).\,dt$$

$$\theta(t) = \int_0^t 2\pi \underline{f_i(t)}.\,dt \longrightarrow (5)$$

Substituting eq (3) in eq (5)

$$\theta(t) = \int_0^t 2\pi\left[f_c + \Delta f \cos(2\pi f_m t)\right] dt$$

$$= \int_0^t 2\pi f_c\,dt + \int_0^t 2\pi\,\Delta f \cos(2\pi f_m t)\,dt$$

$$= 2\pi f_c t + \cancel{2\pi}\Delta f.\frac{\sin 2\pi f_m t}{\cancel{2\pi} f_m}$$

$$\theta(t) = 2\pi f_c t + \frac{\Delta f}{f_m}\sin 2\pi f_m t$$

W.K.T

$$\boxed{\int_0^t \cos at\,dt = \frac{\sin at}{a}}$$

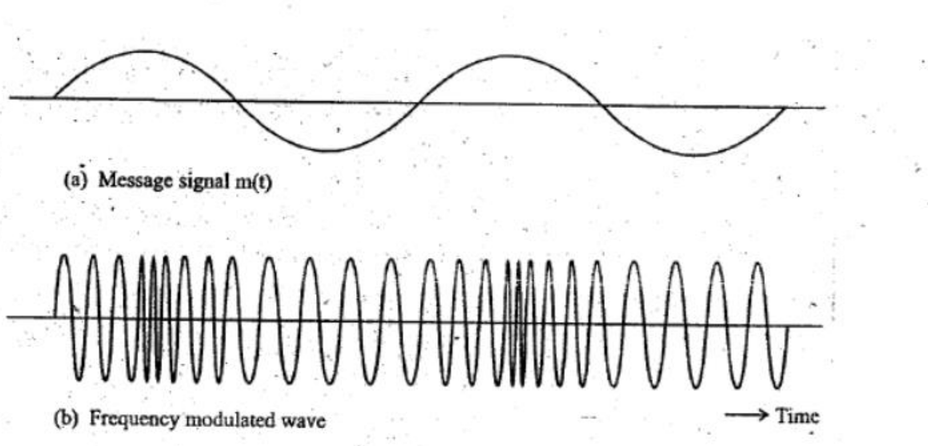
$$\boxed{\theta(t) = 2\pi f_c t + \underline{\beta}\sin(2\pi f_m t)} \longrightarrow (6)$$

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

Substitution eq ⑥ in eq ①, we get

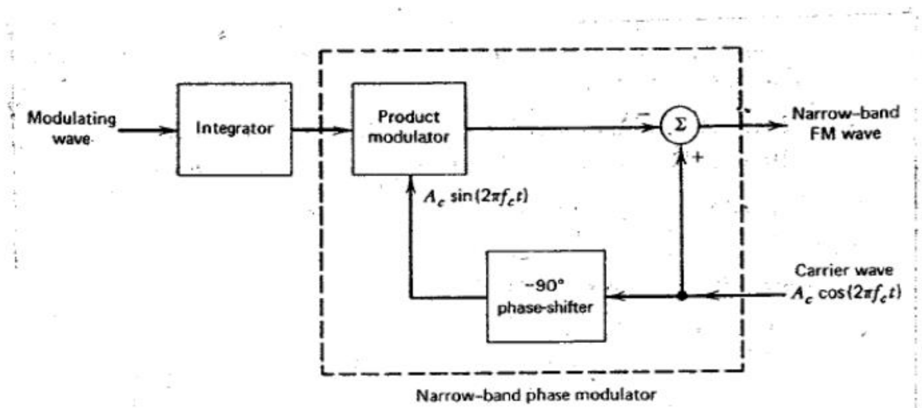
$$\therefore \boxed{s(t) = A_c \cos[2\pi f_c t + \beta \sin(2\pi f_m t)]}$$

Fm wave forms:



3 a) Explain the generation of Narrowband FM and Wideband FM.

Generation of NBFM:



Block diagram of a method for generating a narrow-band FM signal.

We know that the standard equation of FM wave is

$$s(t)=A_c\cos(2\pi f_c t+2\pi k_f\int m(t)dt) \quad (1)$$

$$s(t)=A_c\cos(2\pi f_c t)\cos(2\pi k_f\int m(t)dt) - A_c\sin(2\pi f_c t)\sin(2\pi k_f\int m(t)dt) \quad (2)$$

For NBFM,

$$|2\pi k_f\int m(t)dt| << 1 \quad (3)$$

We know that $\cos\theta\approx1$ and $\sin\theta\approx1$ when θ is very small.

By using the above relations, we will get the NBFM equation as

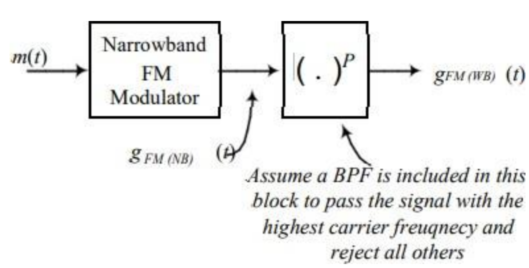
$$s(t)=A_c\cos(2\pi f_c t)-A_c\sin(2\pi f_c t)2\pi k_f\int m(t)dt \quad (4)$$

The block diagram of NBFM modulator as shown in the above figure. Here the integrator is used to integrate the modulating signal m(t)m(t). The Carrier signal $A_c\cos(2\pi f_c t)$ is the phase shifted by -90° to get $A_c\sin(2\pi f_c t)$ with the help of -90° phase shifter. The product modulator has two inputs $\int m(t)dt \int m(t)dt$ and $A_c\sin(2\pi f_c t)$. It produces an output which is the product of these two points.

Positive and Negative signs are assigned for the carrier signal and other form at the input of the summer block. Finally, the summer block produces NBFM wave.

Generation of WBFM:

Consider the following block diagram



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

3 b) What are the advantages, disadvantages, and applications of FM.

Advantages and Disadvantages of Frequency Modulation:

Advantages	**Disadvantages**
Less interference and noise.	Equipment cost is higher. Has a large bandwidth.
Power Consumption is less as compared to AM.	More complicated receiver and transmitter
Adjacent FM channels are separated by guard bands.	The antennas for FM systems should be kept close for better communication.

Applications of Frequency Modulation

If we talk about the applications of frequency modulation, it is mostly used in radio broadcasting.It offers a great advantage in radio transmission as it has a larger signal-to-noise ratio. Meaning, itresults in low radio frequency interference. This is the main reason that many radio stations use FM to broadcast music over the radio.

Additionally, some of its uses are also found in radar, telemetry, seismic prospecting and in EEG, different radio systems, music synthesis as well as in video-transmission instruments. In radio transmission, frequency modulation has a good advantage over other modulation. It has a larger signal-to-noise ratio meaning it will reject radio frequency interferences much better than an equalpower amplitude modulation (AM) signal.

4 a) Explain the generation of FM using direct method.

* In direct FM System, the Instantaneous frequency of the Carrier wave is varied directly in accordance with the message Signal by means of a device called a "voltage Controlled oscillator" (VCO).

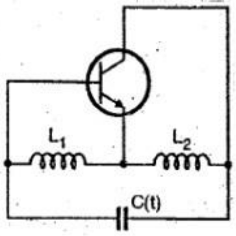
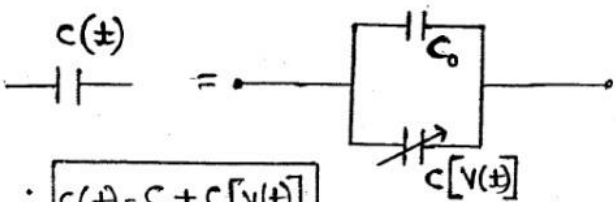


Fig ① Hartley oscillator

* Fig ① Shows a hartley oscillator in which the capacitive component of the frequency determining N/w in the oscillator consists of a Fixed capacitor Shunted by a voltage-variable capacitor.

{



$$\therefore \boxed{c(t) = C_0 + c[V(t)]}$$

}

* The frequency of oscillation of the hartley oscillator is given by:

$$\boxed{f_i(t) = \frac{1}{2\pi\sqrt{(L_1+L_2)\,c(t)}}} \longrightarrow ①$$

Where, $c(t) = C_0 + c[V(t)]$

L_1 & L_2 → are the two Inductances in the frequency-determining the oscillator.

* Assume that ~~for~~ a Sinusoidal modulating wave of frequency 'f_m', the capacitance c(t) is expressed as:

$$\boxed{c(t) = C_0 + \Delta C \cos(2\pi f_m t)} \longrightarrow ②$$

Where, C_0 is the total capacitance in the absence of modulation i.e. $f_m = 0$ &

ΔC is the maximum change in capacitance.

Substituting eq ② in eq ①, we get

$$f_i(t) = \frac{1}{2\pi\sqrt{(L_1+L_2)\quad C_0+\Delta C\cos(2\pi f_m t)}}$$

$$f_i(t) = \frac{1}{2\pi\sqrt{(L_1+L_2)\, C_0\left[1+\frac{\Delta C}{C_0}\cos(2\pi f_m t)\right]}}$$

$$f_i(t) = f_0\,\frac{1}{\sqrt{1+\frac{\Delta C}{C_0}\cos(2\pi f_m t)}}$$

Where, $f_0 = \frac{1}{2\pi\sqrt{(L_1+L_2)\,C_0}}$, unmodulated frequency of oscillation.

$$f_i(t) = f_0\,\frac{1}{\left[1+\frac{\Delta C}{C_0}\cos(2\pi f_m t)\right]^{1/2}}$$

$$f_i(t) = f_0\left[1+\frac{\Delta C}{C_0}\cos(2\pi f_m t)\right]^{-1/2}$$

Recall the binomial theorem

$[1+x]^{-1/2} \approx 1-\frac{x}{2}$

if $|x| \ll 1$

$$f_i(t) = f_0\left[1-\frac{\Delta C}{2C_0}\cos(2\pi f_m t)\right]$$

Let $\boxed{-\frac{\Delta C}{2C_0} = \frac{\Delta f}{f_0}}$

$$f_i(t) = f_0\left[1+\frac{\Delta f}{f_0}\cos(2\pi f_m t)\right]$$

$$f_i(t) = f_0 + \frac{\cancel{f_0}\,\Delta f}{\cancel{f_0}}\cos(2\pi f_m t)$$

$$\boxed{f_i(t) = f_0 + \Delta f\cos(2\pi f_m t)} \longrightarrow (3)$$

Equation (3) is the Instantaneous frequency of an FM wave, assuming Sinusoidal modulation.

4 b) What are the differences between NBFM &WBFM?

Sr No.	Narrow Band FM	Wide Band FM
1	Modulation index is less than 1	Modulation index is greater than 1
2	Frequency Deviation = 5KHz	Frequency Deviation = 75KHz
3	Modulating Frequency = 3KHz	Modulating frequency range from 30 Hz to 15KHz
4	Bandwidth = 2fm	Bandwidth 15 times NBFM, Bandwidth = 2(δ+fmmax)
5	Maximum modulation index is slightly greater than 1	Maximum modulation index between 5 to 2500
6	It is used for mobile communication.	It is used for FM broad casting.

5 a) Classify Frequency modulation techniques.

FM can be divided into **Narrowband FM** and **Wideband FM**.

Narrowband FM:

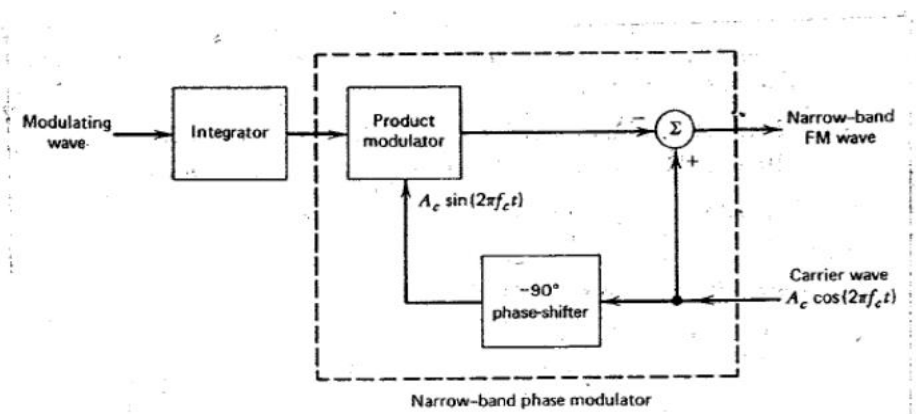
Narrow band FM, NBFM, is used for signals where the deviation is small enough that the terms in the Bessel function is small and the main sidebands are those appearing at ± modulation frequency. The sidebands further out are negligible. For NBFM, the FM modulation index must be less than 0.5, although a figure of 0.2 is often used. For NBFM the audio or data bandwidth is small, but this is acceptable for this type of communication. Narrowband FM is widely used for two-way radio communications. Although digital technologies are taking over, NBFM is still widely used and very effective. Many two way radios or walkie talkies use NBFM, especially those which conform to the license-free standards like PMR446 and FRS radio communications systems. NBFM is ideal for the low cost radio communication systems, especially those that use small walkie talkies because it can be implemented with a minimum of amount of circuitry, most of which is low cost. Although digital technology is becoming much cheaper, narrow band FM is still very cost effective.

Wideband FM:

Wideband FM is typical used for signals where the FM modulation index is above about 0.5. For these signals the sidebands beyond the first two terms are not insignificant. Broadcast FM stations use wide-band FM which enables them to transmit high quality audio, as well as other facilities like stereo, and other facilities like RDS, etc. The wide bandwidth of wide band FM is enables high quality broadcast transmissions to be made, combining a wide frequency response with low noise levels. Once the signal is sufficiently strong, the audio signal to noise ratio is very good. Sometimes high fidelity FM tuners may use a wide-band filter for strong signals to ensure the optimum fidelity and performance. Here the quieting effect of the strong signal will allow for wide-band reception and the full audio bandwidth. For lower strength signals they may switch to a narrower filter to reduce the noise level, although this will result in the audio bandwidth being reduced. However, on balance the narrower bandwidth will give a more pleasing sound when the received signal is low.

5 b) Explain the generation of Narrowband FM wave.

Generation of NBFM:



Block diagram of a method for generating a narrow-band FM signal.

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For NBFM,

$$|2\pi k_f\int m(t)dt| << 1 \quad (3)$$

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By using the above relations, we will get the NBFM equation as

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Positive and Negative signs are assigned for the carrier signal and other form at the input of the summer block. Finally, the summer block produces NBFM wave.

6 a) Discuss about transmission bandwidth and Carson's rule of FM signal.

Theoretically FM has infinite number of side bands. So, the bandwidth required for transmission is also infinite.

Carson generalized the bandwidth formula for an FM wave. According to him the approximate formula for approximate formula for computing the bandwidth of an FM signal generated by a single tone modulating signal frequency 'f_m, is

$$\boxed{B_T \simeq 2(1+\beta) f_m} \longrightarrow (1)$$

The above Formula holds good for all values of β

The transmission bandwidth 'B_T' can also be expressed in terms of frequency deviation 'Δf'

W.K.T $\qquad \beta = \frac{\Delta f}{f_m}$

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

From equation (1)

$$B_T = 2(1+\beta) f_m$$
$$= 2f_m + 2\beta f_m$$
$$= 2f_m + 2\Delta f$$
$$= 2\Delta f\left[1 + \frac{f_m}{\Delta f}\right] \qquad \because \frac{1}{\beta} = \frac{f_m}{\Delta f}$$
$$\boxed{B_T = 2\Delta f\left[1 + 1/\beta\right]}$$

According to Carson's rule, Bandwidth of FM signal is

B=2(Δf+fm)

6 b) A 20 MHz carrier is frequency modulated by a sinusoidal signal such that the peak frequency deviation is 100 kHz. Determine the modulation index and the approximate bandwidth of the FM signal if the frequency of the modulating signal is: (i) 1kHz (ii) 15 kHz.

Given Data

F_c =20MHz,
Δf=100 kHz

i) If the frequency of the modulating signal is 1KHz then,
Modulation Index of FM signal is,

$$\beta \text{ (or) } m_f = \Delta f/fm$$
$$=100KHz/1$$
$$KHz\beta = 100$$

According to Carson's rule, Bandwidth of FM signal is

$$B=2(\Delta f+fm) =2(100KHz+1KHz)$$
$$=2(101KHz)$$
$$B=202KHz.$$

ii) If the frequency of the modulating signal is 15KHz then,
Modulation Index of FM signal is,

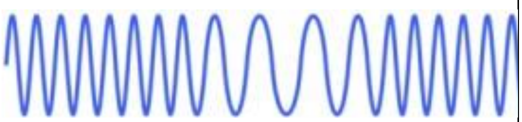
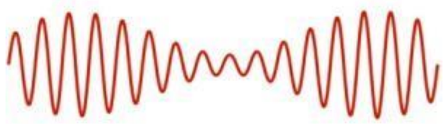
$$\beta \text{ (or) } mf = \Delta f/fm$$
$$=100KHz/15$$
$$KHz\beta = 6.66$$

According to Carson's rule, Bandwidth of FM signal is

$$B=2(\Delta f+fm) =2(100KHz+15KHz)$$
$$=2(115KHz)$$
$$B=230KHz.$$

7 a) Differentiate between the Amplitude Modulation and Frequency Modulation.

S.No	FM	AM
1.	Amplitude of FM wave is constant. It isindependent of the modulation index.	Amplitude of AM wave willchange with the modulatingvoltage.
2.	Hence, transmitted power remains constant.It is independent of m_f .	Transmitted power is dependenton the modulation index.
3.	All the transmitted power is useful.	Carrier power and one sidebandpower are useless.

4.	FM receivers are immune to noise.	AM receivers are not immune tonoise.
5.	It is possible to decrease noise further byincreasing deviation.	This feature is absent in AM.
6.	Bandwidth = $2[\Delta f+f_m]$. The bandwidthdepends on modulation index.	Bandwidth = $2f_m$. It is not dependent on the modulation index.
7.	BW is large. Hence, wide channel is required.	BW is much less than FM.
8.	Space wave is used for propagation. So,radius of transmission is limited to line of sight.	Ground wave and sky wave propagation is used. Therefore, large area is covered than FM.
9.	Hence, it is possible to operate severaltransmitters on same frequency.	Not possible to operate morechannels on same frequency.
10.	FM transmission and reception equipmentare more complex.	AM equipment's are less complex.
11.	The number of sidebands having significant amplitudes depends on modulation index m_f .	Number of sidebands in AM willbe constant and equal to 2.
12.	The information is contained in thefrequency variation of the carrier.	The information is contained in the amplitude variation of the carrier.
13.	FM Wave	AM Wave
		
14.	**Applications:** **Radio**, TV broadcasting,police wireless, point to point communications	**Applications:** Radio and TV broadcasting.

7 b) Describe the construction and functionality of balanced slope detector.

Balanced Slope Detector:

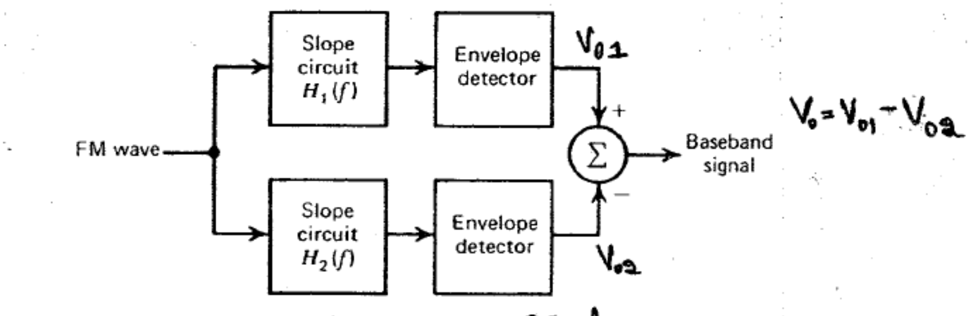


Fig: (a) Block Diagram

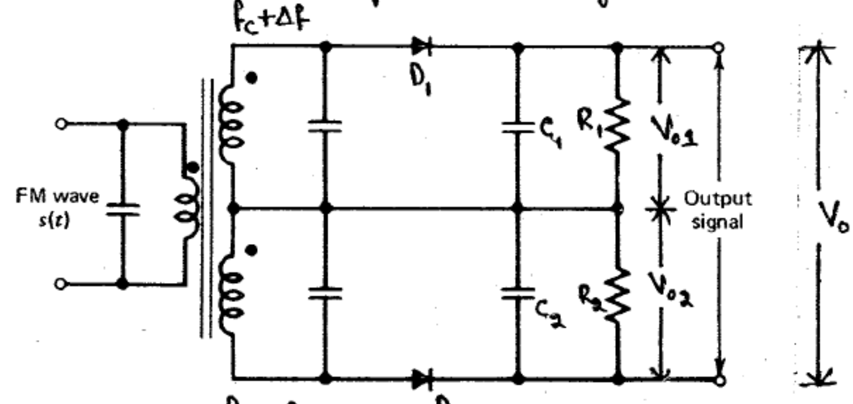


Fig (b): Ckt diagram

* The balanced Slope detector consists of Two Slope detector Ckts.
* The I/p transformer has a center tapped Secondary. Hence the I/p voltages to the two Slope detectors are 180° out of phase.
* There are 3 tuned Ckts
  i) The primary is tuned to IF i.e. f_c.
  ii) The upper tuned Ckt of the Secondary (T_1) is tuned above f_c by Δf i.e. its resonant frequency is $f_c+\Delta f$.
  iii) The lower tuned Ckt of the Secondary (T_2) is tuned below f_c by Δf i.e. its resonant frequency is $f_c-\Delta f$
* R_1C_1 & R_2C_2 are the filter Ckts.
* V_{01} & V_{02} are the o/p voltages of the two Slope detectors.
* The final o/p voltage V_o is obtained by taking the difference

of the Individual o/p voltages V_{01} & V_{02}.

i.e. $\boxed{V_0 = V_{01} - V_{02}}$

operation of the ckt:-

We can understand the operation by dividing the I/p frequency into three ranges as follows:

i) $f_{in} = f_c$:-

When I/p frequency is equal to carrier freq 'f_c', the Induced voltage in the T_1 winding of Secondary is exactly equal to that Induced in the winding T_2.

Thus the I/p voltages to both the diodes D_1 & D_2 will be Same.

∴ The dc o/p voltages V_{01} & V_{02} will also be Identical but they have opposite polarities hence $V_0 = 0V$.

ii) $f_{in} > f_c$:-

$\boxed{f_{in} > f_c \; (f_c + \Delta f)}$ i.e. $f_{in} \simeq f_c + \Delta f$

When I/p frequency is greater than 'f_c', the Induced voltage in 'T_1' winding is higher than that Induced in 'T_2'.

∴ The I/p to D_1 is higher than D_2. So +ve o/p V_{01} (of D_1) is higher than the -ve o/p V_{02} (of D_2).

Thus o/p voltage V_0 is positive. (The +ve o/p voltage increases as the I/p frequency increases towards $f_c + \Delta f$.)

iii) $f_{in} < f_c$:-

$\boxed{\text{i.e. } f_{in} \simeq f_c - \Delta f}$

When I/p frequency is less than 'f_c', the Induced voltage in 'T₂' winding is higher than in 'T₁', So I/p voltage to diode D_2 is higher than that of D_1.

Hence the -ve o/p 'V_{o2}' is greater than V_{o1}.

∴ The o/p voltage of the balanced Slope detector is -ve in this frequency range. { The -ve o/p voltage increases as f_{in} goes closer to '$f_c - \Delta f$' }

$$\therefore \boxed{V_o = \begin{cases} 0, & f_{in} = f_c \\ +ve, & f_{in} > f_c \\ -ve, & f_{in} < f_c \end{cases}}$$

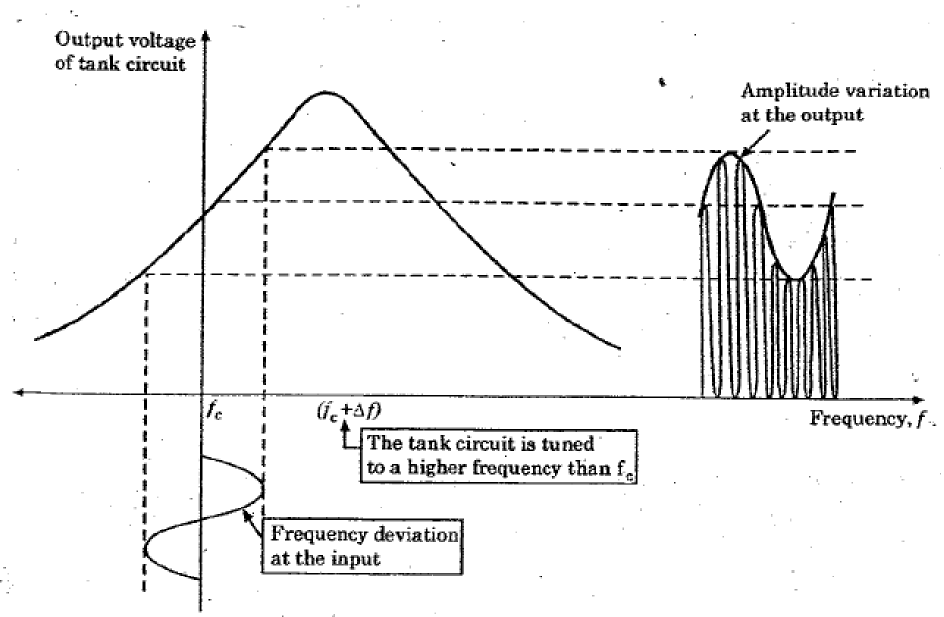
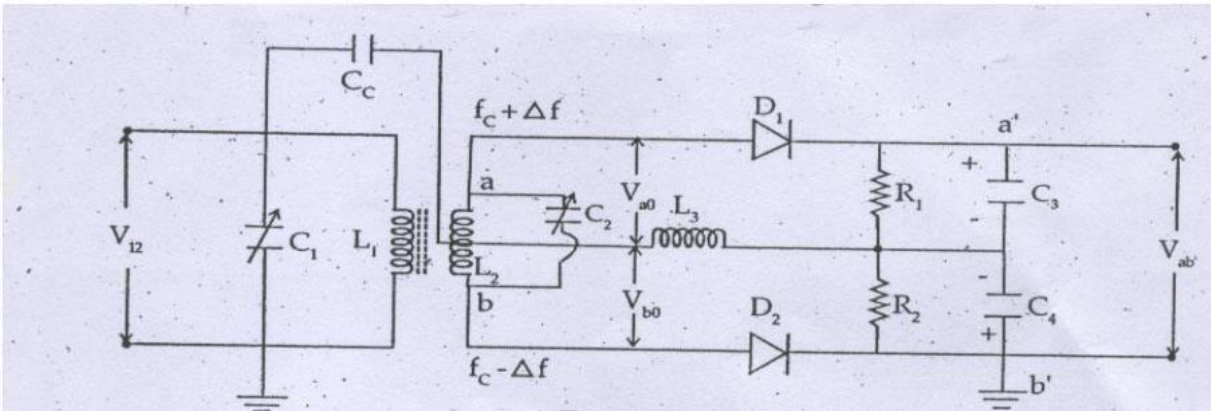


Fig ③ *Characteristics of a slope detector.*

8 a) Describe the functionality of each block of phase shift discriminator.

Phase shift discriminator:

Phase shift discriminator is also call it as a foster seeley discriminator and it converts frequency variations to amplitude variations.



Foster Seely Discriminator

The basic operation of the circuit can be explained by looking at the instances when the instantaneous input equals the carrier frequency, the two halves of the tuned transformer circuit produce the same rectified voltage and the output is zero.

If the frequency of the input changes, the balance between the two halves of the transformer secondary changes, and the result is a voltage proportional to the frequency deviation of the carrier.

Looking in more detail at the circuit,the Foster-Seeley circuit operates using a phase difference between signals. To obtain the differentphased signals a connection is made to the primary side of the transformer using a capacitor, and this is taken to the center tap of the transformer. This gives a signal that is 90° out of phase. Whenan un-modulated carrier is applied at the center frequency, both diodes conduct, to produce equaland opposite voltages across their respective load resistors.

The capacitors C1 and C2 provide a similarfiltering function. The voltage of diode D1 is not equal to voltage at diode D2.

The amplitude variations are rectified and filtered to produce a DC output voltage.

8 b) Explain the block diagram of indirect method in FM generation.

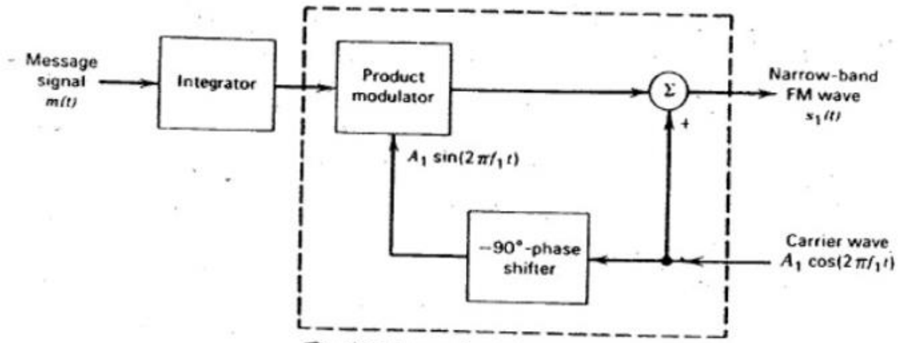


Fig ①: Narrow-band phase modulator

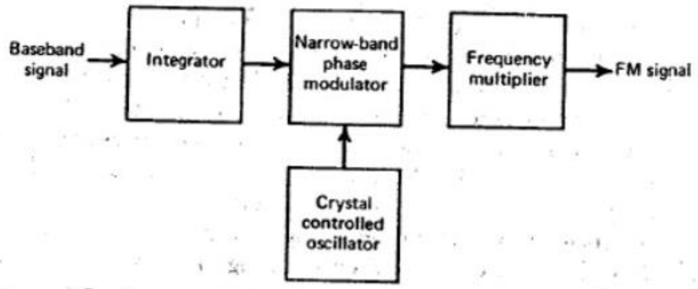


Figure ②: Block diagram of the indirect method of generating a wide-band FM signal.

Figure shows the block diagram of a indirect FM signal system.

In indirect method the message signal m(t) is first period through an integrator before applying it to phase modulator as shown in figure 1

The carrier signal is generated by using crystal oscillator.

Carrier signal is generated by using crystal oscillator because it provides very high frequency stability.

The operation of indirect method is divided into two parts as follows

1. Generate a NBFM wave using a phase modulator.

2. using the frequency multiplier of and mixer to obtain the required values of frequency deviation and modulator modulation index (that is WBFM)

In order to minimize the distortion in the phase modulator, the maximum phase deviation or modulation index 'β' is kept small there by resulting a NBFM signal.

The output of the narrow band phase modulator is then multiplied by a frequency multiplier, producing the desired WBFM wave as shown in figure 2

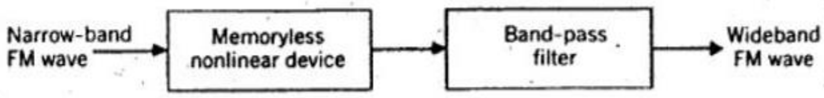


Fig ③: Frequency Multiplier

A frequency multiplier consists of a memoryless nonlinear device followed by a BPF as shown in Figure 3

9 a) Explain briefly about Phase Modulation with necessary waveforms.

Phase Modulation:

The modulation in which Phase of carrier wave is varied in accordance with the instantaneous amplitude of the message signal is known as phase modulation.

The amplitude of the carrier signal remains constant analysis

Message signal is expressed as

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

where
A_m is the amplitude of the message signal
f_m is the frequency of the message signal

Carrier signal is expressed as

$c(t)=A_c \cos (f_c t+\emptyset)$
$c(t)= A_c \cos(\theta(t))$
(or)

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

For phase modulation the instantaneous phase deviation is given as

$$\theta_i(t)= (fct+k_p m(t))$$

k_p is known as phase sensitivity

The phase modulation signal is given as

$S_{pm}(t)= A_c \cos (2\pi f_c t+kpm(t))$

Substituting m(t)

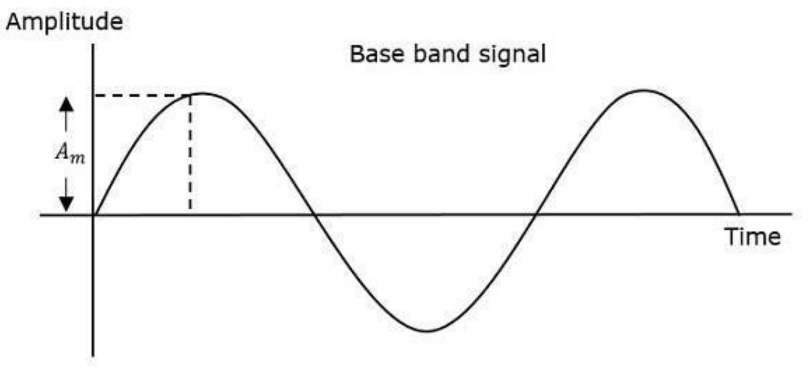
$S_{pm}(t)= A_c \cos (2\pi f_c t+kpAm\cos(2\pi fmt))$

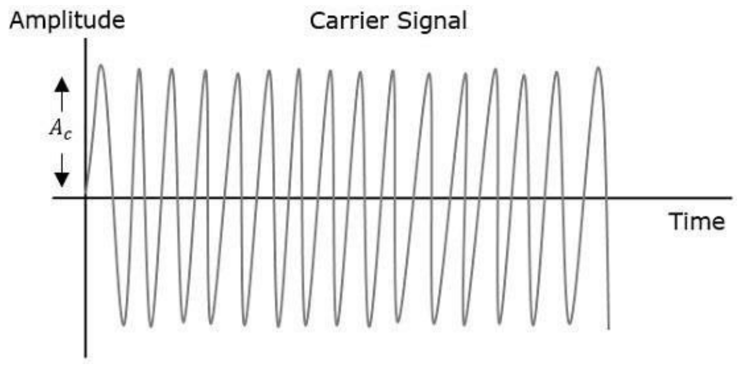
Modulation Index of PM

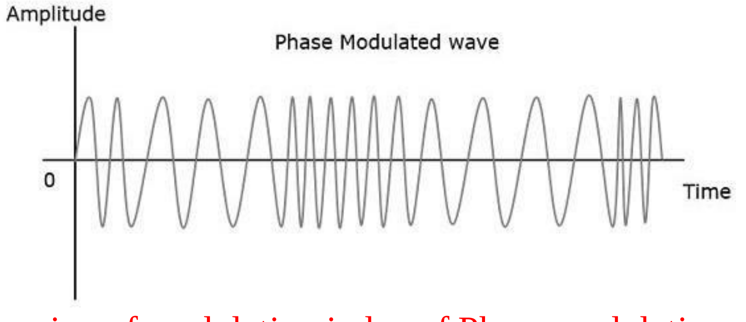
$m_p = k_p A_m$

In phase modulation, the modulation index depends only on the amplitude of the modulating signal.

Modulation index in phase modulation is directly proportional to amplitude of message signal.







9 b) Derive the expression of modulation index of Phase modulation

Mathematical Representation

The equation for instantaneous phase ϕ_i in phase modulation is

$$\varphi_i = k_p m(t)$$

Where,

- kp is the phase sensitivity
- m(t)m(t) is the message signal

The standard equation of angle modulated wave is

$$s(t) = A_c \cos(2\pi f_c t + \varphi_i)$$

Substitute, ϕ_i value in the above equation.

$$s(t) = A_c \cos(2\pi f_c t + k_p m(t))$$

This is the **equation of PM wave**.

If the modulating signal, $m(t) = A_m \cos(2\pi f_m t)$, then the equation of PM wave will be

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

Where,

- ββ = ***modulation index*** = $\Delta\phi = k_p A_m$
- $\Delta\phi$ is phase deviation

Phase modulation is used in mobile communication systems, while frequency modulation is used mainly for FM broadcasting.

10 a) Compare Phase Modulation and Frequency Modulation.

S.No	FM	PM
1.	The frequency deviation is linearly proportional to instantaneous amplitude of the modulating signal.	The phase shift of the carrier is linearly proportional to instantaneous amplitude of the modulating signal.
2.	Frequency modulation is direct method of producing FM signal	Phase modulation is In direct method of Producing FM

3.	The modulation index of an FM signal is the ratio of the frequency deviation to the modulating system.	The modulation index is proportional to the maximum amplitude of the modulating signal.
4.	Amplitude of the FM wave is constant.	Amplitude of the PM wave is constant.
5.	Noise is better suppressed in FM systems as compared to PM system	Noise immunity is interior to that of FM
6.	FM is mainly used for FM broad casting used for entertainment purposes	PM is used in mobile communication system.

10 b) Differentiate between the Frequency Modulation and Phase Modulation with its modulated waveforms.

S.N	FM	PM
1.	The frequency deviation is linearly proportional to instantaneous amplitude of the modulating signal.	The phase shift of the carrier is linearly proportional to instantaneous amplitude of the modulating signal.
2.	Frequency modulation is direct method of producing FM signal	Phase modulation is In direct method of Producing FM
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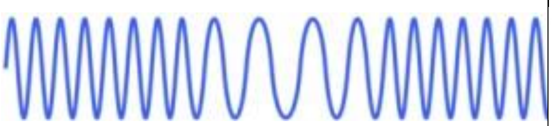
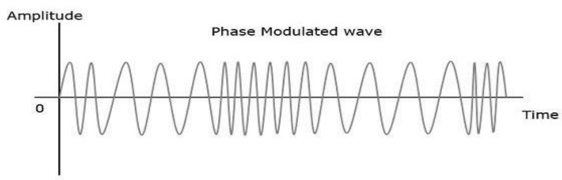
7.	FM Wave: 	PM Wave: 

COURSE CODE: 20EC0451 **R20**

SIDDARTH INSTITUTE OF ENGINEERING & TECHNOLOGY: PUTTUR

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QUESTION BANK (DESCRIPTIVE)

Subject with Code: ICS(20EC0451) **Course& Branch**: B.Tech & CSE,CSM.CIC

Year & Sem: III-B.Tech.& I-Sem. **Regulation:** R20

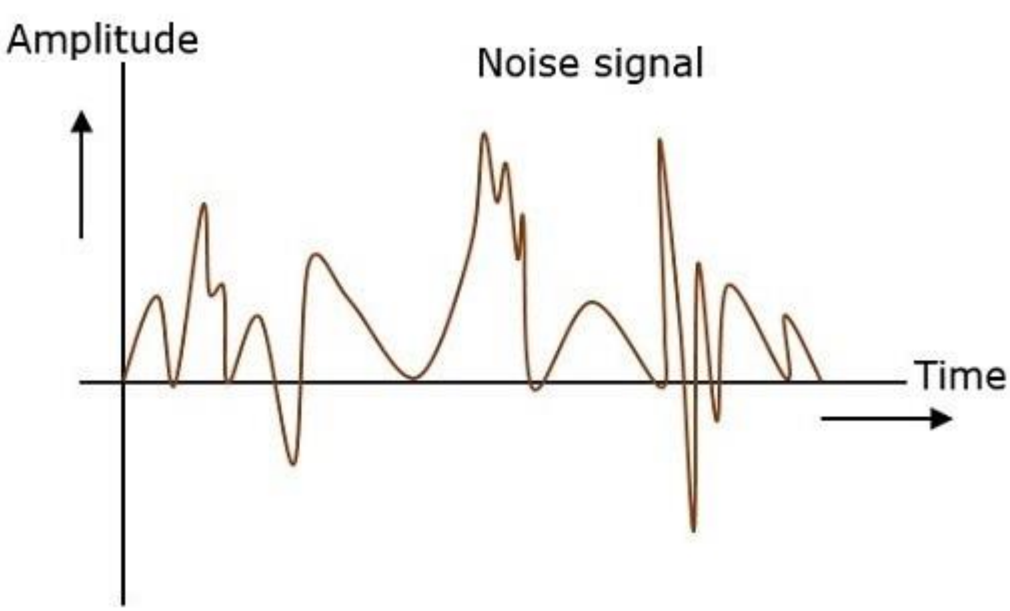
UNIT III
Noise in Communication Systems

1 a) Define Noise and list the different types of noises.

Noise:

Noise is an **unwanted signal**, which interferes with the original message signal and corrupts the parameters of the message signal. This alteration in the communication process, leads to the message getting altered. It most likely enters at the channel or the receiver.

The noise signal can be understood by taking a look at the following figure.



Hence, it is understood that the noise is some signal which has no pattern and no constant frequency or amplitude. It is quite random and unpredictable. It can't be completely eliminated through Measures are usually taken to reduce it.

Most common examples of noise are −

- Hiss sound in radio receivers

- Buzz sound amidst of telephone conversations
- Flicker in television receivers, etc

Types of Noise

The classification of noise is done depending on the type of the source, the effect it shows or the relation it has with the receiver, etc.

There are two main ways in which noise is produced. One is through some **external source** while the other is created by an **internal source**, within the receiver section.

Internal Noise:

- Thermal Noise
- Shot Noise
- Excess Noise

External Noise:

- Atmospheric Noise
- Industrial Noise
- Space or Extra-terrestrial Noise

Thermal Noise:

The noise which is produced by random motion of electrons in a conductor due to thermal agitation is termed as ‘Thermal Noise’ or ‘white noise’ or ‘Johnson noise’.

Generally, the thermal noise power (P_N) is directly proportional to the absolute temperature (T). This noise also has a direct relation with the noise power bandwidth B (i.e. the bandwidth over which the noise is measured)

Mathematically $P_N \propto T_B$

K is Proportional constant (or) Boltzmann’s constant ($1.38*10^{-23}$ joules/kelvin)

T is Absolute temperature (in kelvins)

B is Noise power bandwidth (in Hertz)

Shot Noise:

Short noise is caused due to random variation in current flow in active components.

Main source of short noise are

- Semiconductor diodes
- Transistors
- Tubes

Excess Noise:

Excess noise may be produced due to variations in the carrier density. It usually occurs in tubes, semiconductors and carbon resistors.

Atmospheric Noise:

The noise originated from the natural sources which cause disturbances in atmosphere is known as Atmospheric noise. This noise is generally caused due to spurious radio waves that include voltage in the antennas.

Lighting dis charge in thunderstorms is the best example of atmospheric noise. This noise is also termed as static because of a static electricity discharge cause during the lightening.

Industrial Noise:

Industrial noise is the noise produced due to the equipment's that produce sparks.

The common source of equipment noise are

- Automatic and aircraft ignition.
- Electric motors
- Switching equipment
- Leakage from high voltage lines
- Heavy electric machines
- Computers

Space Noise:

Space noise is further classified into two groups

- Solar Noise
- Cosmic Noise

Solar Noise:

Solar noise which is produced from the sun generates significant amount of noise.

Cosmic Noise:

Cosmic noise is the noise produced due to star or group of stars

The impact of space noise is more at very high frequency range (VHF).

1 b) Explain briefly about Noise in communication system.

Noise:

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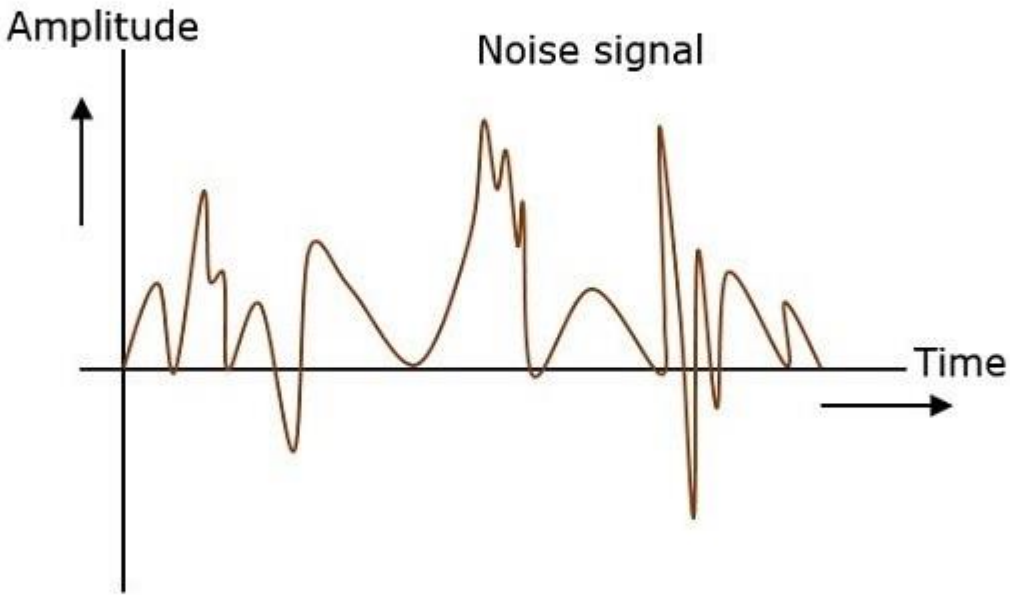

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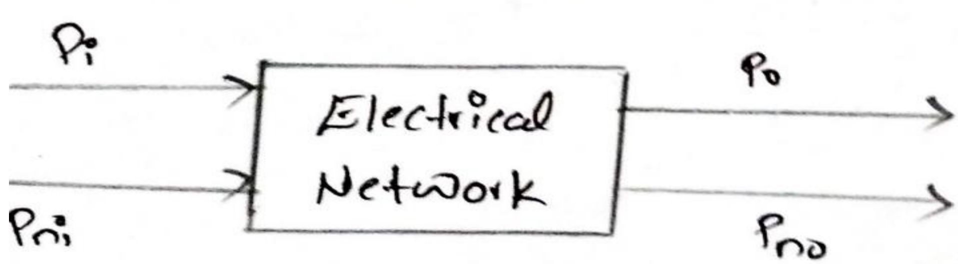
2 a) Explain noise figure and derive its expression.

Noise Figure:

The ratio of signal to noise at input to the signal to noise at the output at the receiver. It is denoted by 'NF' or 'F'.

NF= Signal to noise (S/N) input/(S/N) output



In terms of power $NF = \dfrac{P_i/P_{ni}}{P_o/P_{no}}$

$$= \frac{P_i}{P_{ni}} \cdot \frac{P_{no}}{P_o}$$

$$= \frac{P_i}{P_o} \cdot \frac{P_{no}}{P_{ni}}$$

$$= \frac{1}{G} \cdot \frac{P_{no}}{P_{ni}}$$

$G = P_o/P_i$

Interms of dB.

$$NF\ in\ dB = 10 \log_{10}\left(\frac{S/N\ input}{S/N\ output}\right)$$

$$NF\ in\ dB = (S/N)_{i/p}\ dB - (S/N)_{o/p}\ dB$$

2 b) A mixer stage has a noise figure of 20 dB and it is preceded by another amplifier with a noise figure of 9 dB and an available power gain of 15 dB. Calculate the overall noise figure referred to the input. A cellular telephone system provides a wireless connection to the PSTN for any userlocation within the radio range of the system.

Solution:

(i) First, we convert dB into equivalent power ratios as under:

F_1 = 9 dB or 9 = 10 log F_1 or F_1 = Antilog (0.9) = 7.94

F_2 = 20 dB or 20 = 10 log F_2 or F_2 = Antilog (2) = 100

G_1 = 15 dB or 15 = 10 log G_1 or G_1 = Antilog (1.5) = 31.62

(ii) Then, we calculate the overall noise factor as under:

$$F = F_1 + \frac{F_2 - 1}{G_1}$$

$F = 7.94 + \frac{100-1}{31.62} = 11.07$

(iii) The overall noise figure = 10 log F = 10 $\log_{10}$ (11.07) = 10.44 dB

3 a) Explain briefly about Signal to Noise Ratio.

Signal to Noise Ratio (SNR)

It is defined as the ratio of signal power to noise power.

$$\frac{S}{N} = \frac{P_S}{P_N}$$

$P_S \rightarrow$ Signal power

$P_N \rightarrow$ Noise Power

in dB

$$\left(\frac{S}{N}\right)_{dB} = 10 \log \left(\frac{P_S}{P_N}\right)_{dB}$$

Power in terms of voltage can be written as

$$P_S = \frac{V_S^2}{R}$$

$$P_n = \frac{V_n^2}{R}$$

So, SNR will become

$$\left(\frac{S}{N}\right)_{dB} = 10 \log \left(\frac{V_S^2/R}{V_n^2/R}\right)_{dB}$$

$$= 10 \log \left(\frac{V_S^2}{V_n^2}\right)_{dB}$$

$$= 10 \log \left(\frac{V_S}{V_n}\right)^2_{dB}$$

$$\left(\frac{S}{N}\right)_{dB} = 20 \log \left(\frac{V_S}{V_n}\right)_{dB}$$

Higher value of SNR is good for transmitter and receiver.

3 b) Calculate the input signal to noise ratio for an amplifier with an output signal to noise ratio of 16 dB and a noise figure of 5.4 dB.

Given, $(S/N)_{output} = 16$ dB

Noise Figure, $F_{dB} = 5.4$ dB

$(S/N)_{input} = ?$

We know that,

$F_{dB} = (S/N)_{output}\,dB - (S/N)_{input}\,dB$

$5.4\,dB = 16\,dB - (S/N)_{input}\,dB$

$(S/N)_{input}\,dB = 21.4\,dB$

or Input signal to noise ratio, $(S/N)_{input} = 138.04$

4 a) Explain Pulse Amplitude modulation with its waveforms.

Pulse Amplitude Modulation:

The carrier is in the form of narrow pulses having frequency fc. The uniform sampling takes place in multiplier to generate PAM signal. Samples are placed Ts sec away from each other.

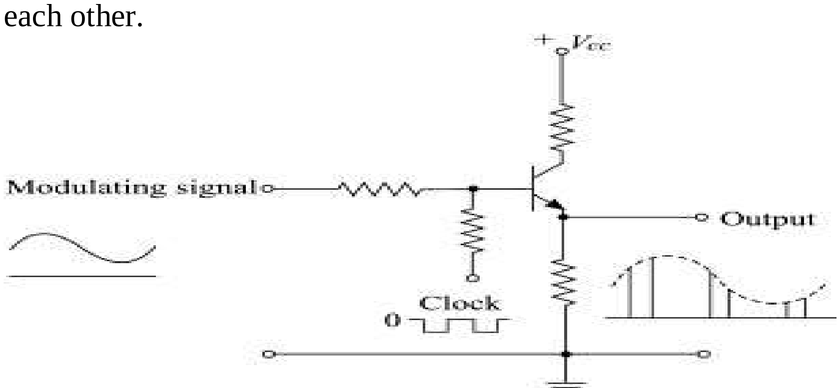


Fig. PAM Modulator

Mathematical Analysis

In a flat top PAM, the top of the samples remains constant and is equal to the instantaneous value of the baseband signal n(t) at the start of sampling.

The duration or width of each sample is τ and sampling rate is equal to,

$$f_s = \frac{1}{T_s}$$

From fig.1 (b), it may be noted that only starting edge of the pulse represents instantaneous value of the baseband signal x(t).

Also, the flat top pulse of g(t) is mathematically equivalent to the convolution of instantaneous sample and a pulse h(t) as depicted in fig.2.

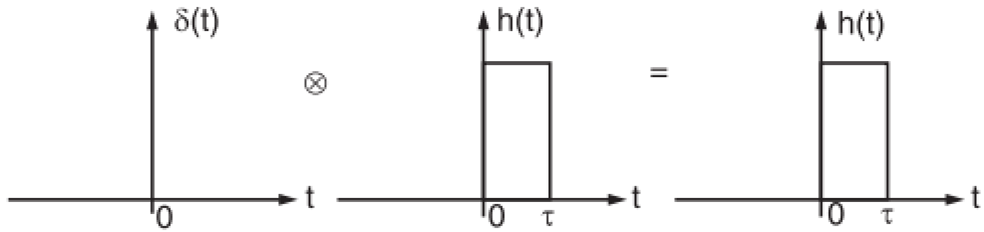


Fig.2 : Convolution of any function with delta function is equal to that function

This means that the width of the pulse in g(t) is determined by the width of h(t) and the sampling instant is determined by the delta function.

In fig.1 (b), the starting edge of the pulse represents the point where baseband signal is sampled and width is determined by function h(t).

Therefore, g(t) will be expressed as,

$$g(t) = s(t) \otimes h(t) \qquad \text{............ (1)}$$

This equation has been explained in fig.3 below.

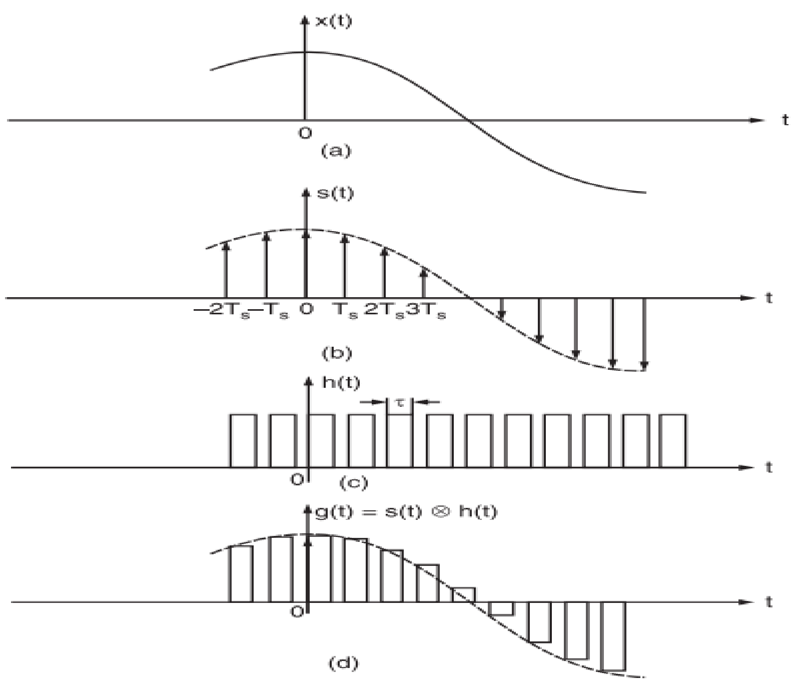
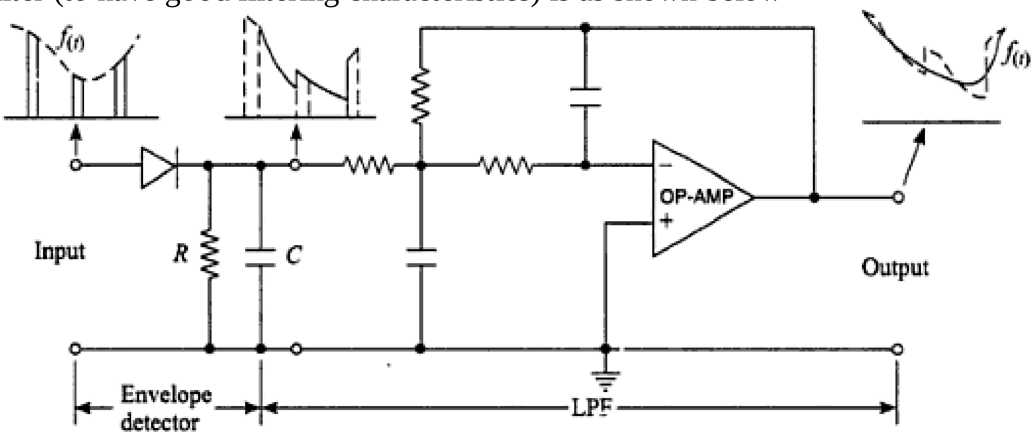


Fig.3: (a) Baseband signal x(t), (b) Instantaneously sample signal s(t), (c) Constant pulse width function h(t), (d) Flat top sampled PAM signal g(t) obtained through convolution of h(t) and s(t)

4 b) Explain the process of demodulation of a PAM signals.

PAM Demodulator:

● The PAM demodulator circuit which is just an envelope detector followed by a second order op-amp low pass filter (to have good filtering characteristics) is as shown below



5 a) What are the advantages and disadvantages of PAM signal.

Advantages of pulse amplitude modulation:

- Transmitter and receiver circuits are simple and easy to construct.
- PAM can generate other pulse modulation signals and can carry the message at the same time.
- Simple process for both modulation and demodulation is used in PAM.
- Data can be transmitted quickly, efficiently and effectively through usual copper wires.

Disadvantages of pulse amplitude modulation:

- Bandwidth should be large for transmission PAM modulation.
- Noise will be great.
- Pulse amplitude signal varies so the power required for transmission will be more.
- For this modulation, noise immunity is low as compared to other types

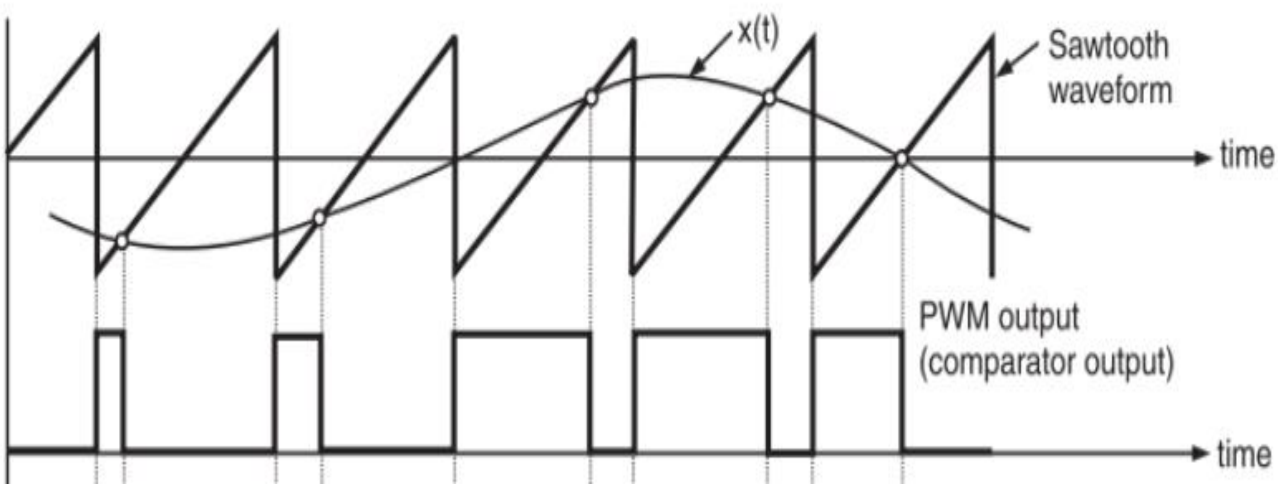
5 b) Define Pulse Width Modulation and classify it with proper diagram.

PWM is also called Pulse Duration Modulation (PDM), Pulse Length Modulation (PLM) and Definition: In PWM, Width of the pulses of the carrier pulse train is varied in accordance with the modulating signal.

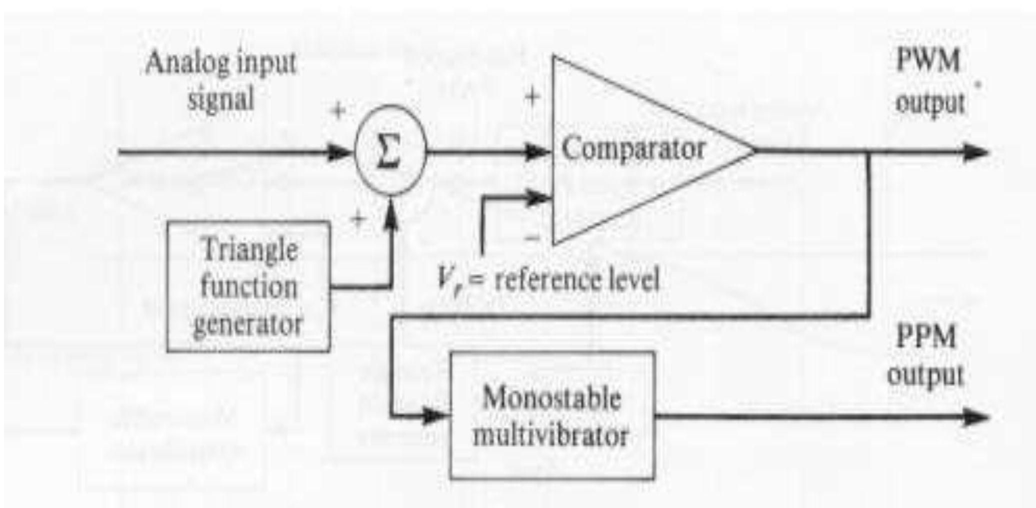
Types of Pulse Width Modulation Technique

There are three conventional types of pulse width modulation technique and they are named as follows:

- **Trail Edge Modulation –** In this technique, the signal's lead edge is modulated, and the trailing edge is kept fixed.
- **Lead Edge Modulation –** In this technique, the signal's lead edge is fixed, and the trailing edge is modulated.
- **Pulse Center Two Edge Modulation –** In this technique, the pulse centre is fixed and both edges of the pulse are modulated.



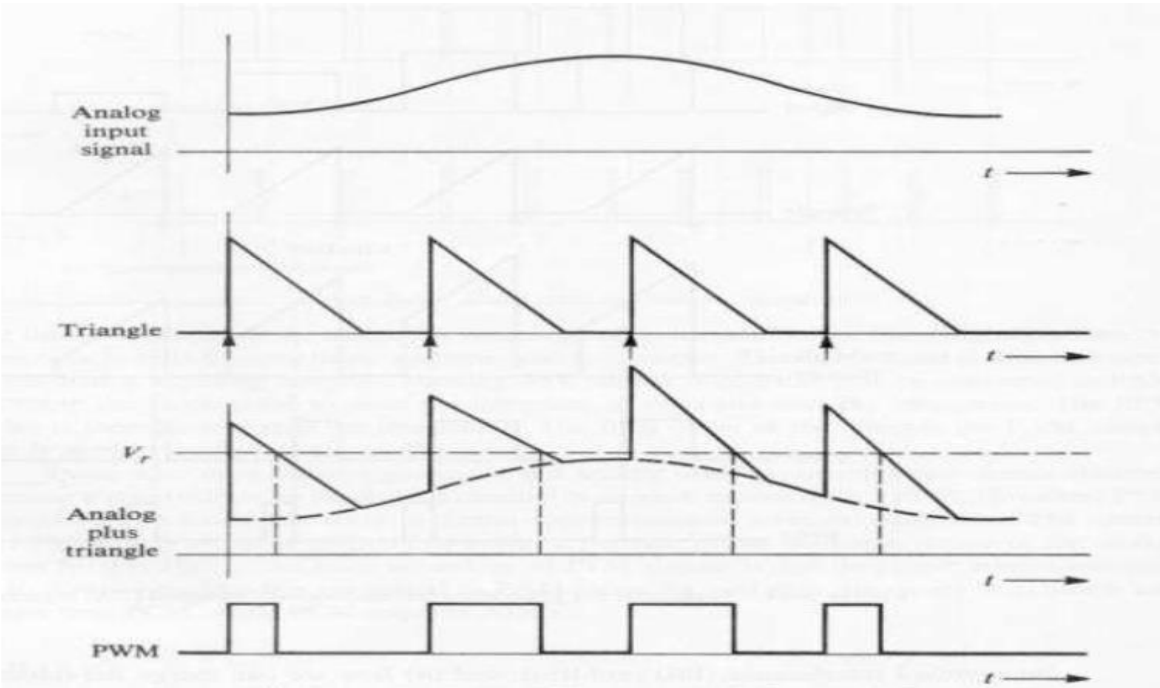
6 a) Explain the process involved in generation of PWM wave.



PWM is also called Pulse Duration Modulation (PDM), Pulse Length Modulation (PLM) and Definition: In PWM, Width of the pulses of the carrier pulse train is varied in accordance with the modulating signal.

1.The PWM pulses obtained at the comparator output are applied to a mono stable multi vibrator which is negative edge triggered.
2.Hence for each trailing edge of PWM signal, the monostable output goes high.
3.It remains high for a fixed time decided by its RC components.

4.Thus, as the trailing edges of the PWM signal keeps shifting in proportion with the modulating signal, the PPM pulses also keep shifting.
5.Therefore, all the PPM pulses have the same amplitude and width. The information is conveyed via changing position of pulses.



6 b) Describe the demodulation technique of PWM signal.

Demodulation:

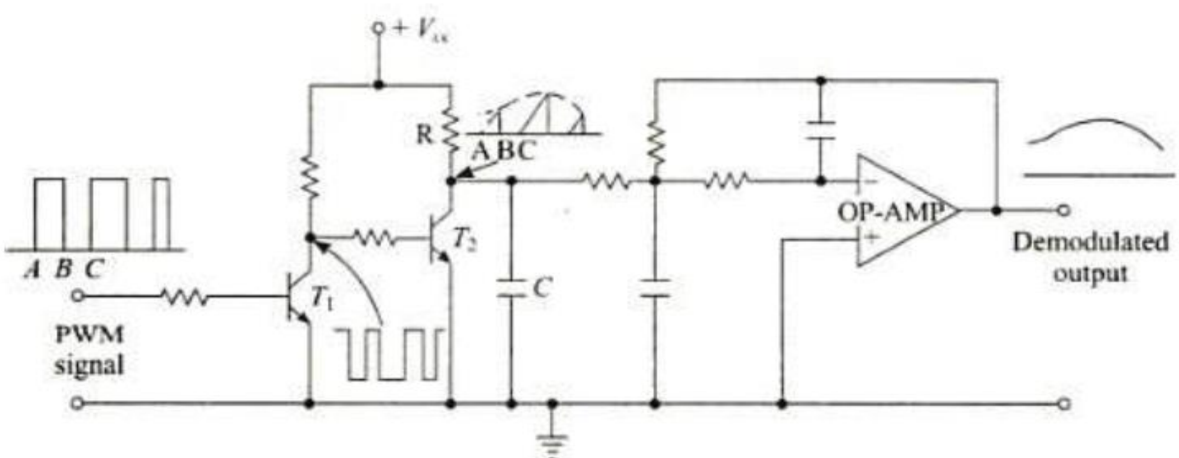


Fig. PWM Demodulator

- Transistor T1 works as an inverter.
- During time interval A-B when the PWM signal is high the input to transistor T2 islow.
- Therefore, during this time interval T2 is cut-off and capacitor C is charged throughan R-C combination.
- During time interval B-C when PWM signal is low, the input to transistor T2 is high,and it gets saturated.
- The capacitor C discharges rapidly through T2.The collector voltage of T2 during B-C is low.
- Thus, the waveform at the collector of T2is similar to saw-tooth waveform whoseenvelope is the modulating signal.
- Passing it through 2nd order op-amp Low Pass Filter, gives demodulated signal.

7 a) What are the advantages and disadvantages of PWM signal?

Advantages of pulse width modulation:

- The demodulation process employed in PWM system is simple
- PWM has good noise immunity as amplitude of the signal is kept constant.
- PWM system does not require synchronization between transmitter and receiver.

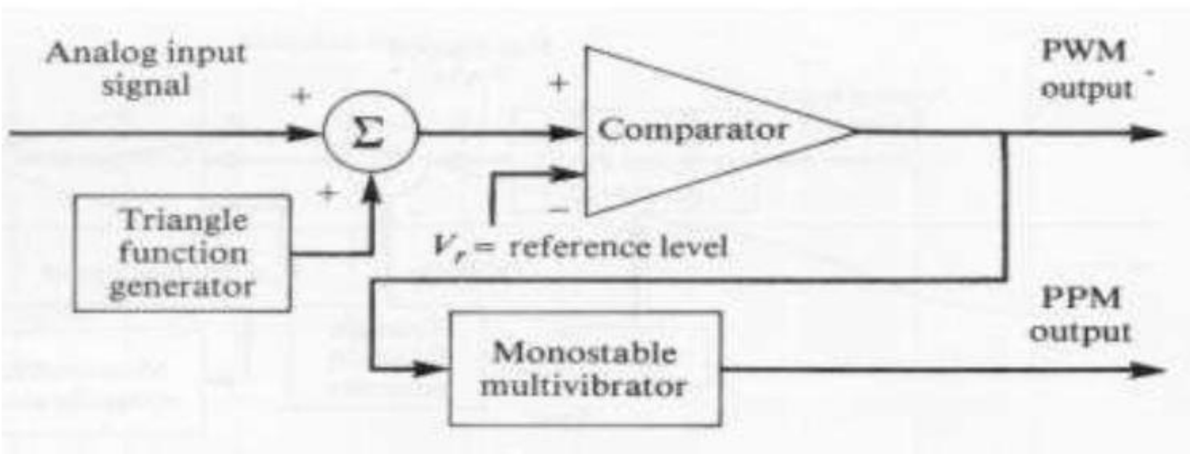
Disadvantages of pulse width modulation:

- Due to pulses varying in width, the signal power is also varied.
- The transmission of PWM signal requires large bandwidth when compared to PAM signal.
- PWM is not suitable for time division multiplexing.

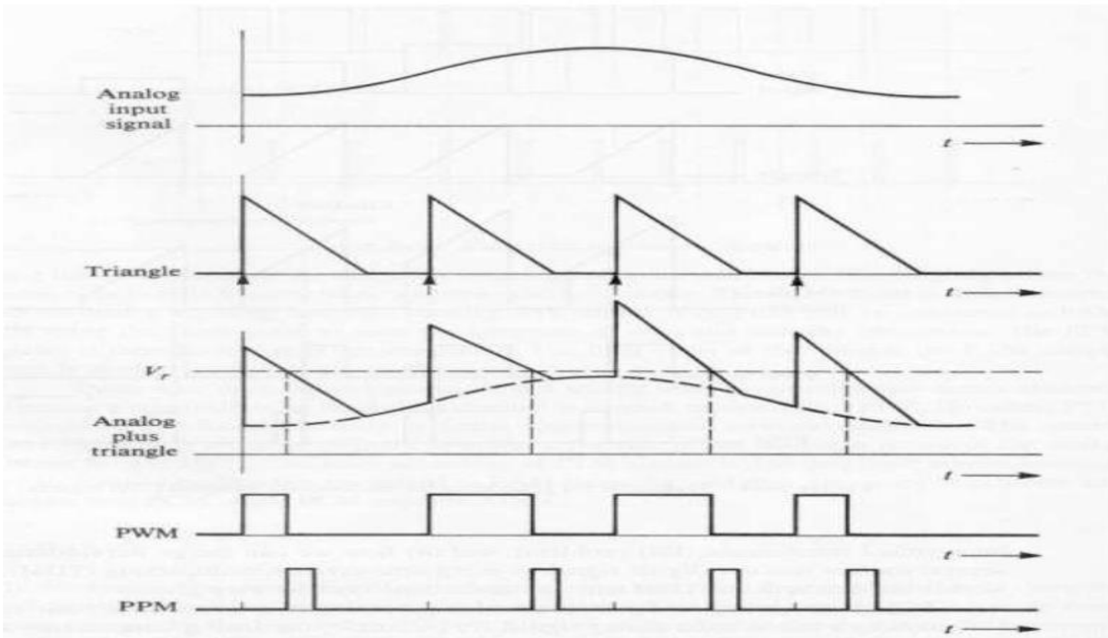
7 b) Differentiate between the Pulse Amplitude Modulation and Pulse Width Modulation with its modulated waveforms.

PAM	PWM
Amplitude is varied	Width is varied
Bandwidth depends on the width of the pulse	Bandwidth depends on the rise time of the pulse
Instantaneous transmitter power varies with the amplitude of the pulses	Instantaneous transmitter power varies with the amplitude and width of the pulses
System complexity is high	System complexity is low
Noise interference is high	Noise interference is low
It is similar to amplitude modulation	It is similar to frequency modulation

8 a) Explain about the generation of PPM signal.



1.The PWM pulses obtained at the comparator output are applied to a mono stable multi vibrator which is negative edge triggered.
2.Hence for each trailing edge of PWM signal, the monostable output goes high.
3.It remains high for a fixed time decided by its RC components.
4.Thus, as the trailing edges of the PWM signal keeps shifting in proportion with the modulating signal, the PPM pulses also keep shifting.
5.Therefore, all the PPM pulses have the same amplitude and width. The information is conveyed via changing position of pulses.



8 b) Elaborate demodulation of PPM signal.

PPM Demodulator:

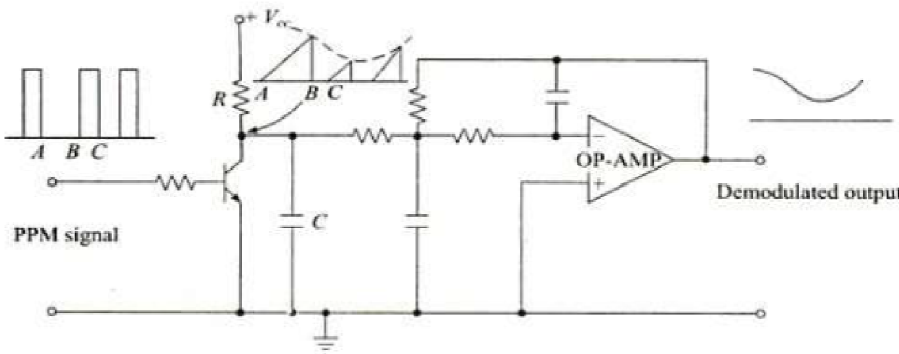


Fig. PPM Demodulator

- The gaps between the pulses of a PPM signal contain the information regarding themodulating signal.
- During gap A-B between the pulses the transistor is cut-off and the capacitor C getscharged through R-C combination.
- During the pulse duration B-C the capacitor discharges through transistor and thecollector voltage becomes low.
- Thus, waveform across collector is saw-tooth waveform whose envelope is the modulating signal.
- Passing it through 2nd order op-amp Low Pass Filter, gives demodulated signal.

9 a) What are the advantages and disadvantages of PPM signal

Advantages of pulse Position modulation:

- Demodulation process is simple in PPM.
- In PPM, noise interference is less as amplitude is kept constant.
- Because of the constant pulse widths and amplitudes, signal power is also constant in PPM.

Disadvantages of pulse Position modulation:

- PPM signals require synchronization between transmitter and receiver.
- The transmission bandwidth required for a PPM signal is large when compared to that of PAM signal.
- PPM is not suitable for time division multiplexing.

9 b) Differentiate between the Pulse Position Modulation and Pulse Width Modulation with its modulated waveforms.

PWM	PPM
Width is varied	Position is varied
Bandwidth depends on the rise time of the pulse	Bandwidth depends on the rise time of the pulse
Instantaneous transmitter power varies with the amplitude and width of the pulses	Instantaneous transmitter power remains constant with the width of the pulses
System complexity is low	System complexity is low
Noise interference is low	Noise interference is low
It is similar to frequency modulation	It is similar to phase modulation

10 a) Define pulse modulation and different types of pulse modulation in analog and digital communication.

Pulse Modulation: Pulse Modulation can be defined as the variation of amplitude or width or position of a higher frequency discrete carrier signal in accordance with the amplitude of an analog modulating signal.

Pulse Modulation can be broadly classified into two major types. They are

- Analog pulse modulation
- Digital pulse modulation

Analog Pulse Modulation: In analog pulse modulation technique amplitude or time of a carrier is varied in accordance with the instantaneous value of analog modulating signal.

Analog pulse modulation technique is further classified into two types namely

Pulse Amplitude Modulation (PAM): A modulation technique in which the amplitude of the carrier signal consisting of periodic train or pulses is varied linearly with amplitude of the message signal is known as pulse amplitude modulation.

Pulse Time Modulation (PTM): The modulation technique in which the timing of the carrier pulse is changed with respect to the amplitude of the message signal is known as pulse time modulation.

The pulse time modulation is further divided into two types. They are

Pulse Width Modulation (PWM): A modulation technique in which, the width of the carrier signal consisting of periodic train pulse is varied linearly with the amplitude of the message signal is known as pulse width modulation. It is also termed as Pulse Duration Modulation(PDM).

Pulse Position Modulation (PPM): A modulation technique in which, the position of the carrier signal consisting of periodic train pulse is varied linearly with the amplitude of the message signal is called pulse position modulation.

Digital Pulse Modulation: In digital pulse modulation technique, analog modulation signal is converted into discrete signal by changing the amplitude of carrier pulse train. These discrete levels are then represented by digital codes for transmission.

Digital pulse modulation technique is further classified into two types. They are

- Pulse Code Modulation (PCM)
- Delta Modulation (DM)

Pulse Code Modulation (PCM): Pulse code modulation can be defined as a signal encoding technique, where in an analog information signal is sampled and amplitude of these samples is approximated to the nearest value among the finite set of discrete levels. This approximation is carried out such that both amplitude and times is indicated in discrete format.

PCM is further classified into Differential Pulse Code Modulation (DPCM). In this type of modulation, difference in the amplitude levels of two successive samples is transmitted instead of the absolute value of the actual sample.

Delta Modulation (DM): Delta modulation is the simplest form of DPCM wherein difference between successive samples are encoded into data streams of n-bits. It employs single bit DPCM code to digitally transmit analog signals.

It is further classified into Adaptive Delta Modulation (ADM) technique. ADM can be defined as a delta modulation technique which varies step size of the signal, based on the amplitude characteristics of the applied analog signal.

10 b) Compare PAM, PWM and PPM techniques.

PAM	PWM	PPM
Amplitude is varied	Width is varied	Position is varied
Bandwidth depends on the width of the pulse	Bandwidth depends on the rise time of the pulse	Bandwidth depends on the rise time of the pulse
Instantaneous transmitter power varies with the amplitude of the pulses	Instantaneous transmitter power varies with the amplitude and width of the pulses	Instantaneous transmitter power remains constant with the width of the pulses
System complexity is high	System complexity is low	System complexity is low
Noise interference is high	Noise interference is low	Noise interference is low
It is similar to amplitude modulation	It is similar to frequency modulation	It is similar to phase modulation

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QUESTION BANK (DESCRIPTIVE)

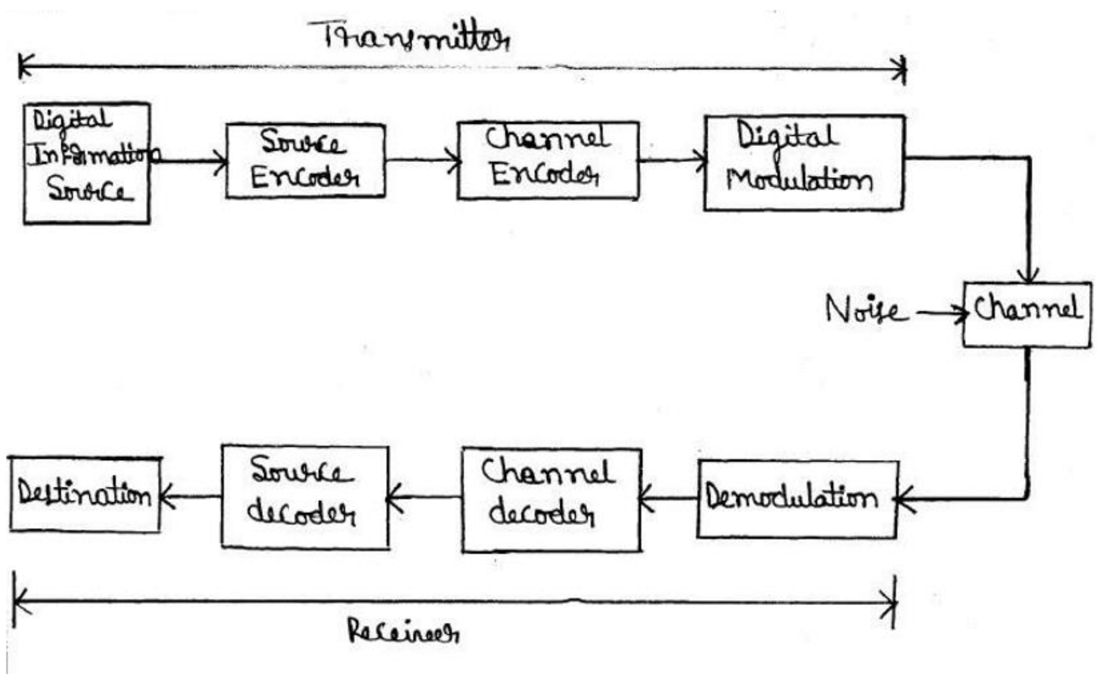
Subject with Code: **ICS(20EC0451)** **Course& Branch**: B.Tech & CSE,CSM.CIC

Year & Sem: III-B.Tech.& I-Sem. **Regulation:** R20

UNIT IV
Digital Communication

1 a) Define Digital Communication and draw the basic block diagram of Digital communication system.

Digital Communication: Digital communication is defined as the process of exchanging information between two or more communicating points using digital signals.



1 b) Explain the function of each block of Digital communication system.

Source of information: There are two types of source information.

- Analog Information Source

- Digital information source

Analog Information source: Microphone actuated by a speech, T.V channel, camera, scanning a sense continuous amplitude signal

Digital Information source: Analog information source can be transferred from discrete to sampling.

Source Encoder and Decoder: The symbol can't be transmitted directly and then converted into 0's and 1's (Binary) is called source encoder. It converts binary output of the channel.

Channel of the decoder into a symbol form is called source decoder.

Channel Encoder and Decoder: Channel encoder adds some binary bits to the input signal is called channel encoder.

Channel decoder at the receiver is able to reconstruct error sequence and reduce the distortion is called channel decoder.

Modulator: Modulator can be used to minimized the effect of channel noise to match the frequency of spectrum and capability for many signals is called Modulator.

Demodulator: The extraction of the message from the information waveforms produced by the modulator and reduce the demodulator output of the demodulation is bit stream.

Channel: Channel provides the connection between sources and destination is called channel.

2 a) Explain the Process of Quantization with suitable example.

Quantization: Quantization of signals can be defined as the conversion of an analog information signal into discrete form, where in infinite number of levels are transformed into finite number of conditions. In the process of quantization, the peak to peak range of input sample values is divided into decision levels or thresholds of finite set (or value). Among the available finite set of representation levels, the output is allotted with a discrete value in the quantization process.

During the rounding off process of analog sample values, a significant amount of error or noise known as quantization error is produced in the quantizer. This error is directly proportional to the difference between consecutive quantization levels and inversely proportional to the number of levels for amplitude range.

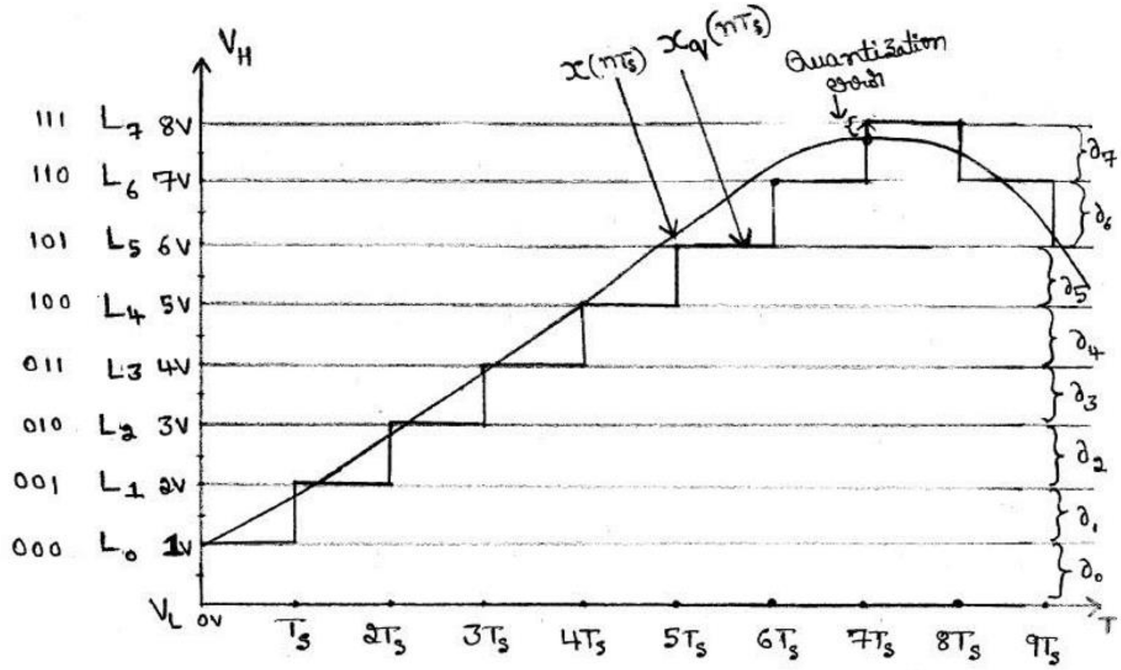


Fig ①: Quantizing operation.

2 b) Discuss the different types of Quantization in detail.

Types of Quantization: Based on the distribution of quantization levels to be spaced uniformly, the process of quantization is classified into two types. They are

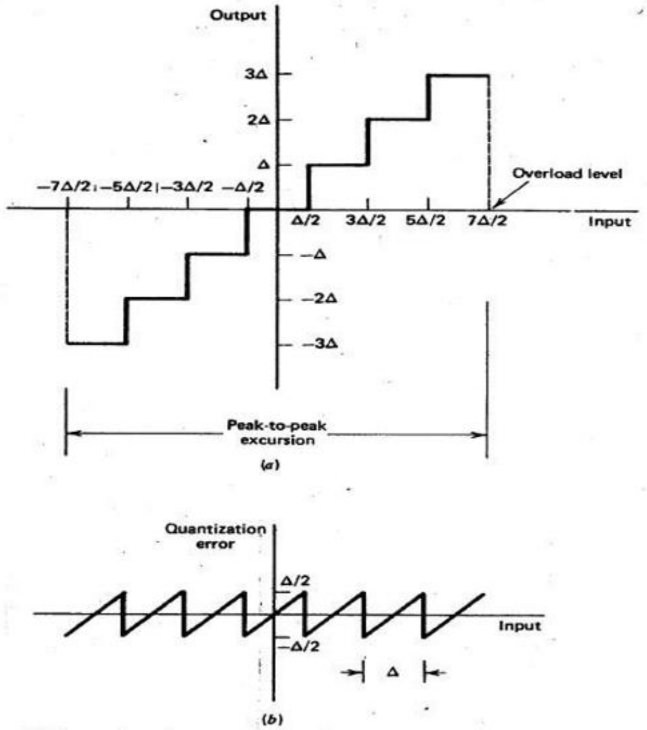
- Uniform Quantization
- Non uniform Quantization

Uniform Quantization: The process of quantization wherein the quantization levels (or step size) are spaced uniformly over the entire range of input signal is known as uniform quantization.

Based on the position of origin, uniform quantization is again subdivided into two types namely,

- Midtread uniform quantization
- Midriser uniform quantization

Midtread Uniform Quantization: In this type of quantization, origin lies in the middle of a tread of quantization staircase process as shown in figure (i).



(*a*) Transfer characteristic of quantizer of midtread type. (*b*) Variation of the quantization error with input.

It can be observed from figure(i) that the gap between representation levels and decision thresholds are designated by a common value known as step size (Δ).

The origin lies at the middle of a staircase riser when representation levels are at +Δ/2 and -Δ/2, +3Δ/2 and -3Δ/2, +5Δ/2 and -5Δ/2, +7Δ/2 and -7Δ/2….. and decision thresholds are maintained at 0, +Δ and -Δ, +2Δ and -2Δ, +3Δ and -3Δ, ………

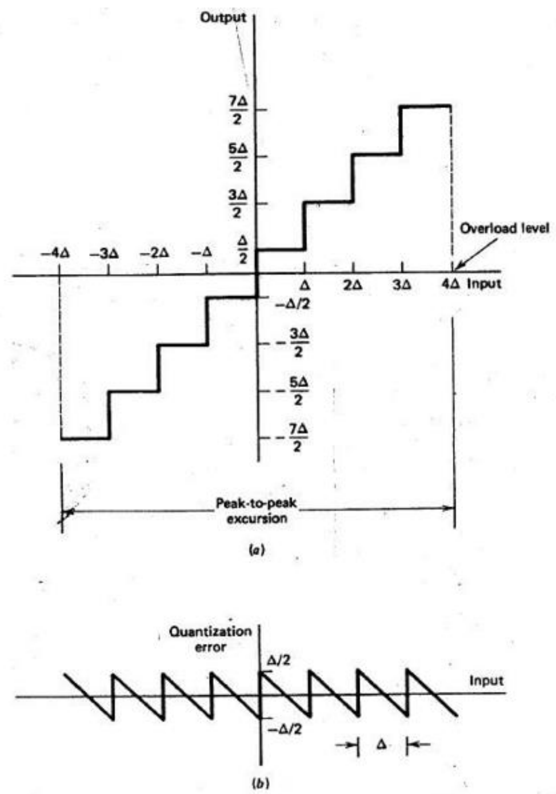
The number of quantized levels in the midtread quantization is an odd number and is given as,

Number of quantized level=2^m-1

Where m is number of bits used for encoding a sample.

This type of quantization process finds its major application in voice communications.

Midriser Uniform Quantization: In this type of quantization origin lies in the middle of a rise of quantization staircase process as shown in figure (ii).



(a) Transfer characteristic of quantizer of midriser type. *(b)* Variation of the quantization error with input.

The number of quantized levels in the midriser quantization is an even number and is given as,

Number of quantized level=2^m-1

Where m is number of bits used for encoding a sample.

Non Uniform Quantization: The process of quantization where in the quantization levels are not spaced uniformly and step size varies with respect to the relative amplitude level of sampled value is known as non-uniform quantization.

3 a) Illustrate the different types of Quantization noise.

Quantization Noise: The difference between the original sampled value are quantized signal is known as quantization error.

$$E=x_q(nT_s)-x(nT_s)$$

Let us consider the input signal α(t) is continuous and linearly varied with the range of $-x_{max}$ to x_{max}

The input signal x(t) is sampled to produce $x(nT_s)$ and quantized into q quantization levels

The total amplitude= $x_{max} - (-x_{max}) = x_{max} + x_{max} = 2\, x_{max}$

Let the total amplitude is divided into q quantization levels then the step size is

$$\text{Step size } \Delta = \frac{2\ \text{xmax}}{q}$$

If the input is normalized to the amplitude of unity, then x_{max} =1 and - x_{max} =-1 and Δ=2/q

If the step size is small, then the quantization error e will be an uniformly distributed variable therefore the maximum quantization error is

$$E_{max}=\Delta/2$$

Therefore, the quantization error depends on step size. If step size is large the quantization error is high. To reduce the quantization error of PCM, step size Δ should be less than or equal to one

$$\Delta\geq1$$

Types of Quantization: Based on the distribution of quantization levels to be spaced uniformly, the process of quantization is classified into two types. They are

- Uniform Quantization
- Non uniform Quantization

Uniform Quantization: The process of quantization wherein the quantization levels (or step size) are spaced uniformly over the entire range of input signal is known as uniform quantization.

Non Uniform Quantization: The process of quantization where in the quantization levels are not spaced uniformly and step size varies with respect to the relative amplitude level of sampled value is known as non-uniform quantization.

3 b) State sampling theorem. What is Nyquist rate and Nyquist interval?

Sampling Theorem: According to sampling theorem, a continuous time signal can be completely represented in its samples and recovered back, if the sampling frequency f_s is greater than (or) equal to the twice of highest frequency component of the message signal f_m.

i.e. $f_s \geq 2f_m$

Nyquist Rate: The minimum sampling rate at which both sampling and reconstruction of a signal (from its samples) can be performed without any distortion is called Nyquist rate. It is denoted by f_s and is given as

i.e. $f_s=2f_m$

Nyquist Interval: Nyquist interval can be defined as the maximum time interval between the equally spaced samples of the signal during which sampling rate is equal to Nyquist rate.

Nyquist interval is equal to reciprocal of Nyquist rate and is given by

$$T_s=1/2f_{max}$$

4 a) Illustrate with a neat block diagram explain PCM transmitter and receiver.

Pulse code Modulation (PCM): Pulse code modulation is an analog to digital convertor. The block diagram of pulse code modulation system is shown in figure

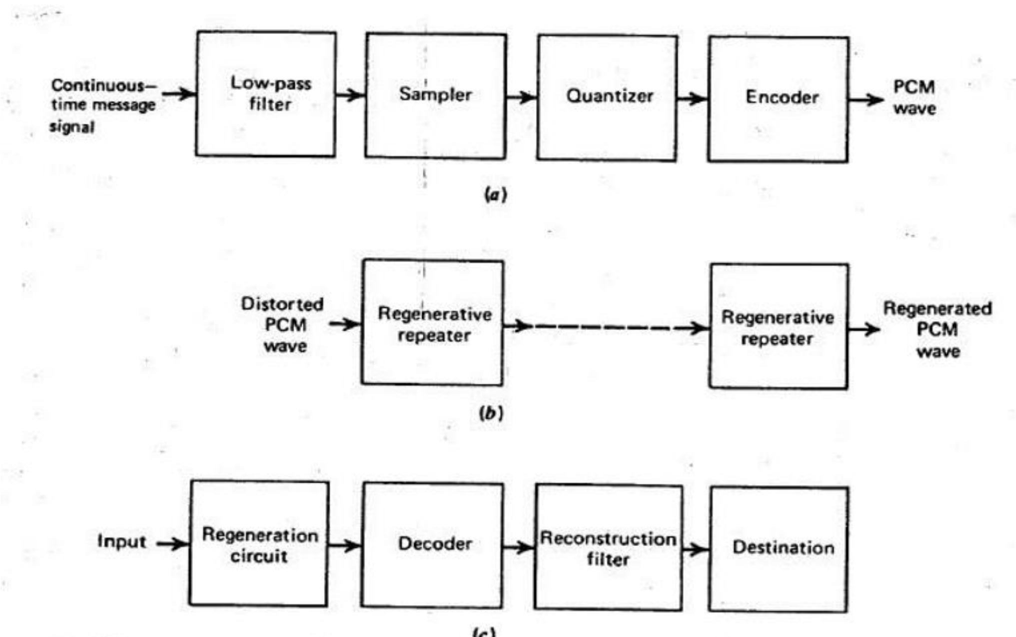


Fig ① : Basic elements of a PCM system. (a) Transmitter. (b) Transmission path. (c) Receiver.

The block diagram of a pulse code modulation system is shown in figure. It consists of

- Transmitter
- Regenerative repeater
- Receiver

PCM Transmitter: In PCM Transmitter consisting Low pass filter, samples, quantizer and encoder and finally it generated PCM signal.

Transmission Path: In transmission path one major circuitry is there that is Regenerative repeater circuit.

PCM Receiver: PCM Receiver consisting decoder Low pass filter and destination finally it gives D/A (Digital to Analog) conversion and get the original signal.

4 b) What are the advantages & disadvantages of PCM?

Advantages of Pulse Code Modulation:

- It has a higher noise immunity.
- It has a higher transmitter efficiency.
- Easily multiplexed.
- Uniform transmission quality.
- Low manufacturing effect.
- Due to its digital nature we can easily store PCM signals.

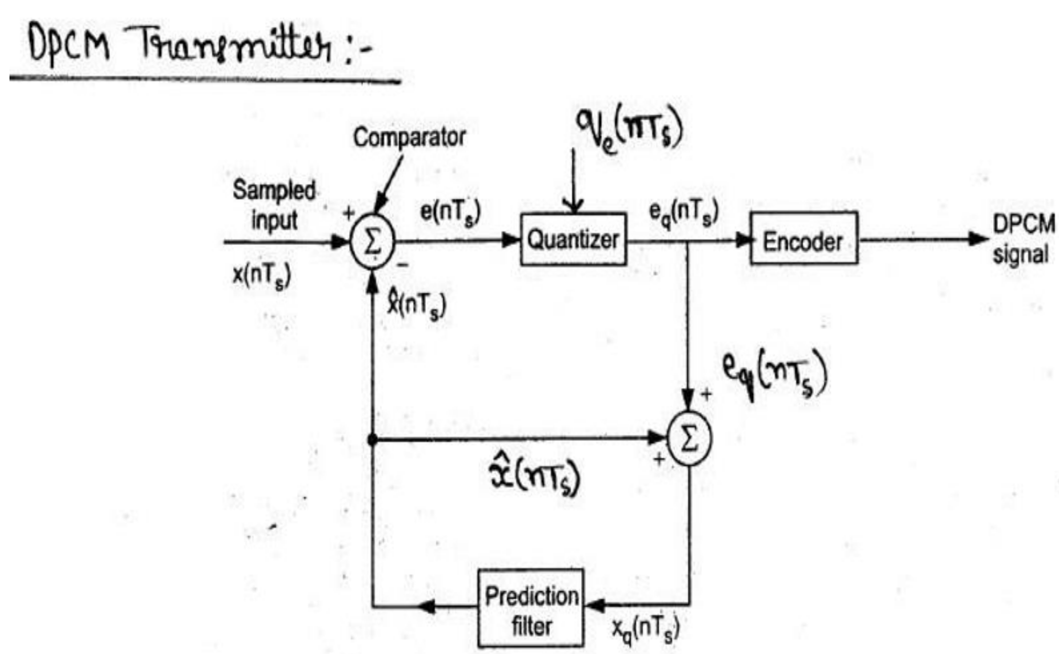
Disadvantages of Pulse Code Modulation:

- It requires large bandwidth as compared to other method called an analog system.
- Over loaded appears when modulation signal change between sampling by an amount greater than the size of the step.
- The difference between the original analog signal and the translated digital signal is called quantization error.
- Encoding, decoding and also have quantizing circuit of PCM is very complex.
- Noise and cross talk leave low but rise attenuation.

5 a) Explain DPCM system with neat diagram.

Differential Pulse Code Modulation: In the use of Pulse Code Modulation for the digization of a voice or video signal, the signal is sampled of a rate slightly higher than the Nyquist rate. The resulting sampled signal is then found to exhibit a high correlation between adjacent samples. It is standard PCM system.

The base band signal x(t) is sampled of the rate $f_s=1/T_s$ sequences of correlated samples T_s second a part.



Differential pulse code modulation transmitter

From fig ①, $x(nT_s)$ represents the Sampled version of the analog Signal $x(t)$.

* The o/p of the Comparator is the difference between the unquantized Sampled I/p & prediction of its $\hat{x}(nT_s)$

i.e. $$\boxed{e(nT_s) = x(nT_s) - \hat{x}(nT_s)} \longrightarrow ①$$

Where $\hat{x}(nT_s)$ is the prediction of $x(nT_s)$.

* The prediction error $e(nT_s)$ is the quantized to produce $e_q(nT_s)$. In the quantizer the noise $q_e(nT_s)$ gets added.

∴ The o/p of quantizer can be written as :

$$\boxed{e_q(nT_s) = e(nT_s) + q_e(nT_s)} \longrightarrow ②$$

* From fig ①, the I/p to the prediction filter may be written as :

$$x_q(nT_s) = \hat{x}(nT_s) + \underline{e_q(nT_s)} \longrightarrow ③$$

Substituting eq ② in eq ③, we get

$$x_q(nT_s) = \hat{x}(nT_s) + \underline{e(nT_s)} + q_e(nT_s) \longrightarrow ④$$

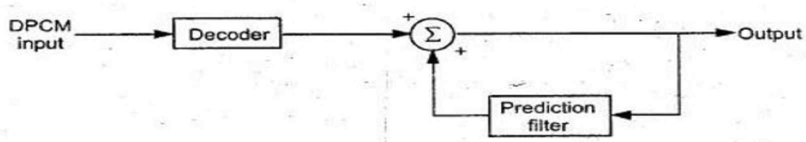
Substituting eq ① in eq ④, we get

$$x_q(nT_s) = \cancel{\hat{x}(nT_s)} + x(nT_s) - \cancel{\hat{x}(nT_s)} + q_e(nT_s)$$

$$\boxed{x_q(nT_s) = x(nT_s) + q_e(nT_s)} \longrightarrow ⑤$$

Where $x_q(nT_s)$ is the quantized version of $x(nT_s)$.

DPCM Receiver :-



DPCM receiver

* The decoder 1st reconstruct the quantized error signal from incoming binary signal.
* The prediction filter o/p & quantized error signals are summed up to give the quantized version of the original signal.

* Thus the signal at the receiver differs from actual signal by quantization error $q_e(nT_s)$, which is introduced permanently in the reconstructed signal.

5 b) What are the advantages & disadvantages of DPCM.

Advantages of DPCM:

- Bandwidth Requirement of DPCM is less compared to PCM.
- Quantization error is reduced because of prediction filter.
- Number of bits used to represent one sample value of are also reduced compared to PCM.
- Reduce the redundance information.

Disadvantages of DPCM:

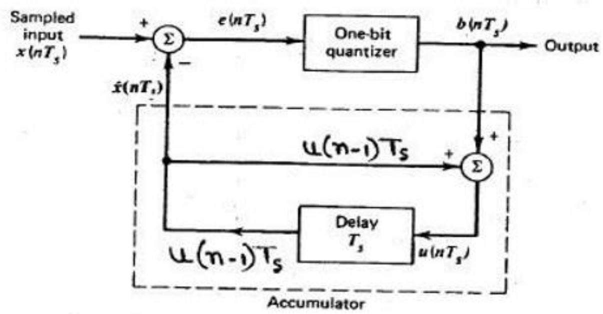
- High bit rate.
- Needs prediction filter circuit to be used which is very high complex.
- Practical usage is limit.

6 a) Explain DM (delta modulation system) with suitable diagrams.

Delta Modulation (DM): Delta Modulation transmits only one bit per sample that is the present sample value is compared with the previous sample value and the indication whether the amplitude is increased or decreased is sent.

The input signal x(t) is approximated to step signal by the data modulator. The difference is between input signal x(t) and staircase approximated signal is quantized into only two levels that is +Δ or -Δ

DM Transmitter :-



* The error between the Sampled value $x(nT_s)$ & last approximated Sample is given by

$$\boxed{e(nT_s) = x(nT_s) - \hat{x}(nT_s)} \longrightarrow (1)$$

* Let $u(nT_s)$ be the present Sample approximation of Staircase o/p.

From Fig (3): $\hat{x}(nT_s) = u(n-1)T_s$

$$\boxed{\hat{x}(nT_s) = u(nT_s - T_s)} \longrightarrow (2)$$

Substituting eq (2) in eq (1), we get

$$\boxed{e(nT_s) = x(nT_s) - u(nT_s - T_s)} \longrightarrow (3)$$

* The binary quantity $b(nT_s)$ is the algebraic sign of the error $e(nT_s)$, except for the scaling factor δ.

i.e. $$\boxed{b(nT_s) = \delta \, \text{sgn}[e(nT_s)]} \longrightarrow (4)$$

$b(nT_s)$ depends on the sign of error $e(nT_s)$, the sign of step-size 'δ' will be decided

i.e. $$b(nT_s) = +\delta \,, \quad \text{if } x(nT_s) \geq \hat{x}(nT_s)$$
$$b(nT_s) = -\delta \,, \quad \text{if } x(nT_s) \leq \hat{x}(nT_s)$$

* If $b(nT_s) = +\delta$, then binary '1' is transmitted

If $b(nT_s) = -\delta$, then binary '0' is transmitted.

$$\therefore \quad \boxed{u(nT_s) = u[nT_s - T_s] + b(nT_s)}$$

* The previous sample approximation $u[nT_s - T_s]$ is restored by delaying one sample period 'T_s'.

DM Receiver :-

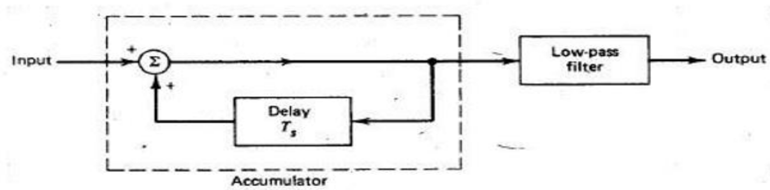


Fig shows the block diagram of DM receiver.

* The accumulator generates the staircase approximated signal o/p & is delayed by one sampling period 'T_s'. It is then added to the I/p signal.

* If I/p is binary '1' then it adds +δ Step to the previous O/P
* If I/p is binary '0' then one Step 'δ' is Subtracted from the delayed Signal.

6 b) Compare PCM, DPCM, and DM.

Parameters	PCM	DPCM	DM
Number of bits	It can use 4, 8 or 16 bits per sample	Bits can be more than one but less than PCM	It uses only one bit for one Sample.
levels and step size	The number of levels depends on number of bits. level size is fixed	step size is kept fixed and cannot be varied Number of levels is fixed.	step size is kept fixed and cannot be varied
complexity of implementation	System is complex	Simple	Simple

Quantization error and distortion	Quantization error depends on number of levels used	Slope overload distortion and quantization noise is present	Slope overload distortion and granular noise are present.
Transmission bandwidth	Highest bandwidth is required since number of bits are high	Bandwidth required is less than PCM	lowest bandwidth is required.
Feedback	There is no feed back in transmitter or receiver	Feedback exists	Feedback exists in transmitter

7 a) Draw the block diagram of ASK modulator and demodulator and explain the operation.

Amplitude Shift Keying: Figure (i) illustrates the block diagram of a ASK waveform generator that employs aa product modulator with a unipolar binary wave m(t) and carrier wave $A_c \cos(2\pi f_c t)$ as its inputs.

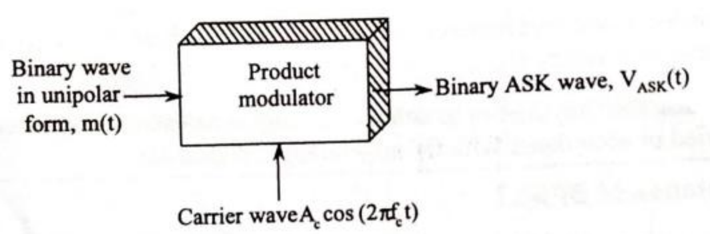


Figure (i): Generation of ASK Waveform

In the above modulation process, amplitude of the carrier signal switches between the binary values 1 and 0. During the binary value 1 a sinusoidal carrier of amplitude A_c and frequency f is transmitted for the bit duration of T_b secs. On the other hand, the binary value 0, the carrier wave is switched off for the bit duration of T_b secs.

Figure(ii) illustrates the generation of ASK signal waveform for digital input signal and sinusoidal analog carrier signal.

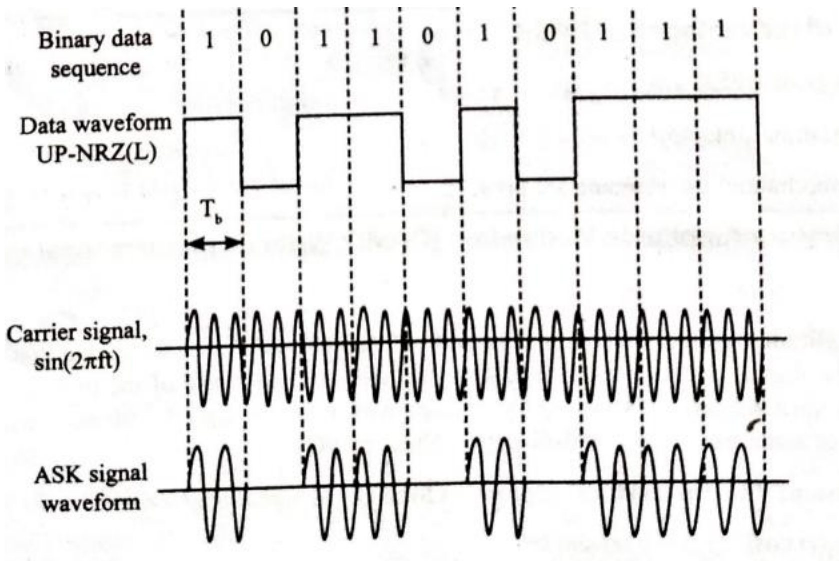


Figure (ii): ASK Signal Waveform

It can be observed from figure(ii) that

- ASK signal waveform is undergoes one change for every variation in the binary stream from logic 1 to logic 0 or from logic 0 to logic 1.
- Logic 1 of the input binary data produces constant amplitude ASK and constant frequency carrier signals.
- Logic 0 of input binary data produces zero ASK signal and no carrier signal at the ASK system.

The mathematical expression of ASK signal is given as,

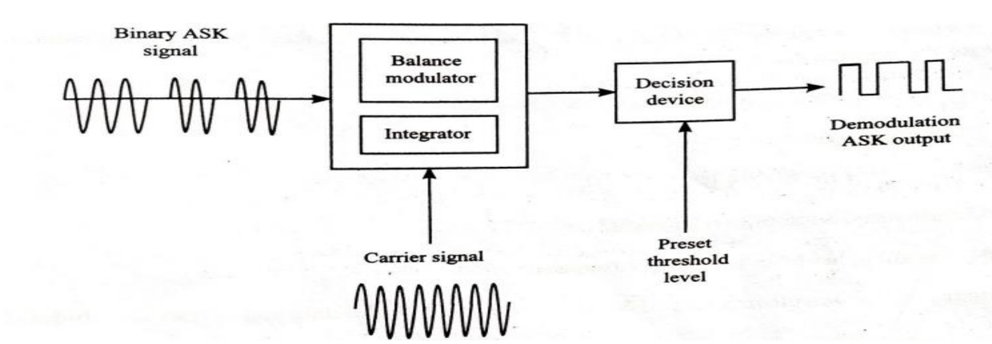
The mathematical expression of ASK signal is given as,

$$v_{ASK}(t) = \begin{cases} V_c \cos(2\pi f_c t) & \text{for binary value 1} \\ 0 & \text{for binary value 0} \end{cases}$$

Where, V_c – Maximum amplitude of analog carrier signal

f_c – Carrier frequency.

ASK Demodulator:



It can be observed from figure that the ASK demodulator consists of three elements namely balance modulator, integrator and a decision device.

- Balance modulator is provided with a binary ASK signal along with a high frequency sinusoidal carrier signal. The output of the balance modulator is operated by the integrator circuit for successive bit intervals (T_b). this integrator basically performs the function of a low pass filter.
- The output of the integrator circuit is fed to the decision making device. This device compares the integrator output with a preset threshold level. It produces symbol 1 when the threshold value is exceeded and produced symbol 0 when the threshold value does not exceed.

Thus, the output of the ASK demodulator produces ASK demodulated output or original digital data.

7 b) Explain with suitable waveforms Amplitude Shift Keying.

Amplitude Shift Keying: Figure (i) illustrates the block diagram of a ASK waveform generator that employs aa product modulator with a unipolar binary wave m(t) and carrier wave $A_c \cos(2\pi f_c t)$ as its inputs.

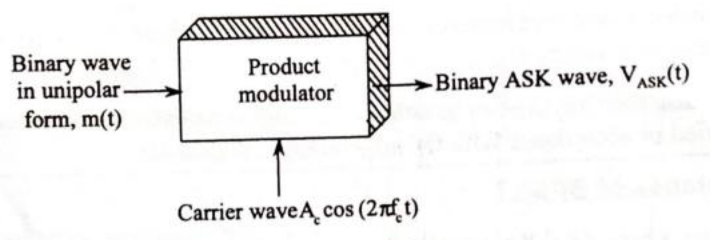


Figure (i): Generation of ASK Waveform

In the above modulation process, amplitude of the carrier signal switches between the binary values 1 and 0. During the binary value 1 a sinusoidal carrier of amplitude A_c and frequency f is transmitted for the bit duration of T_b secs. On the other hand, the binary value 0, the carrier wave is switched off for the bit duration of T_b secs.

Figure(ii) illustrates the generation of ASK signal waveform for digital input signal and sinusoidal analog carrier signal.

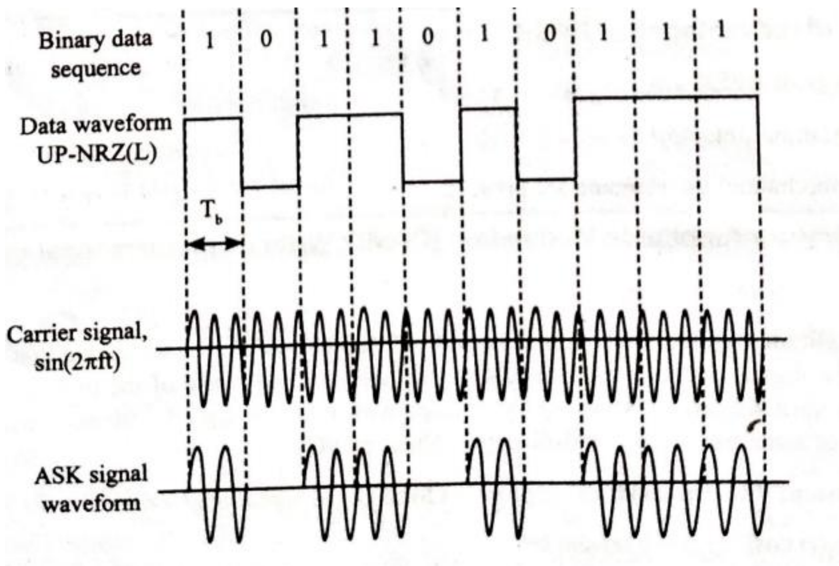


Figure (ii): ASK Signal Waveform

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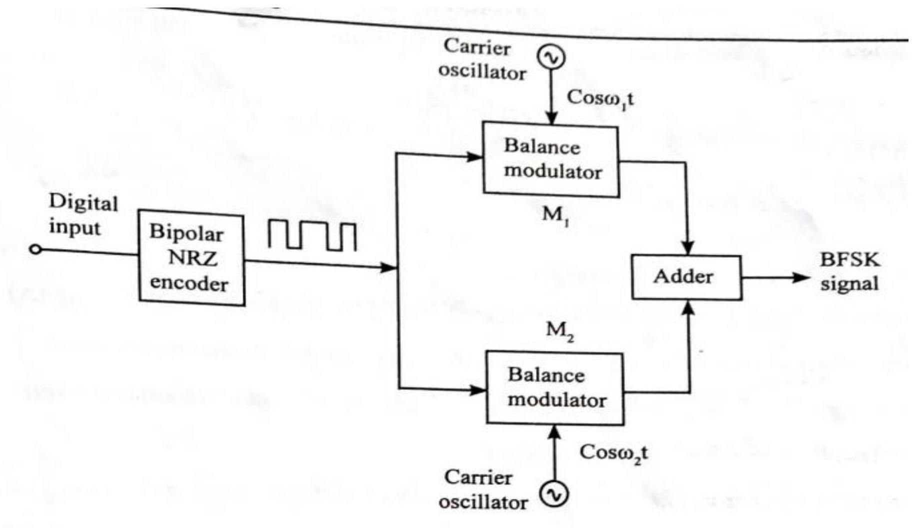
$$v_{ASK}(t) = V_c \cos(2\pi f_c t) \text{ for binary value 1}$$
$$0 \quad \text{for binary value 0}$$

Where, V_c – Maximum amplitude of analog carrier signal

f_c – Carrier frequency.

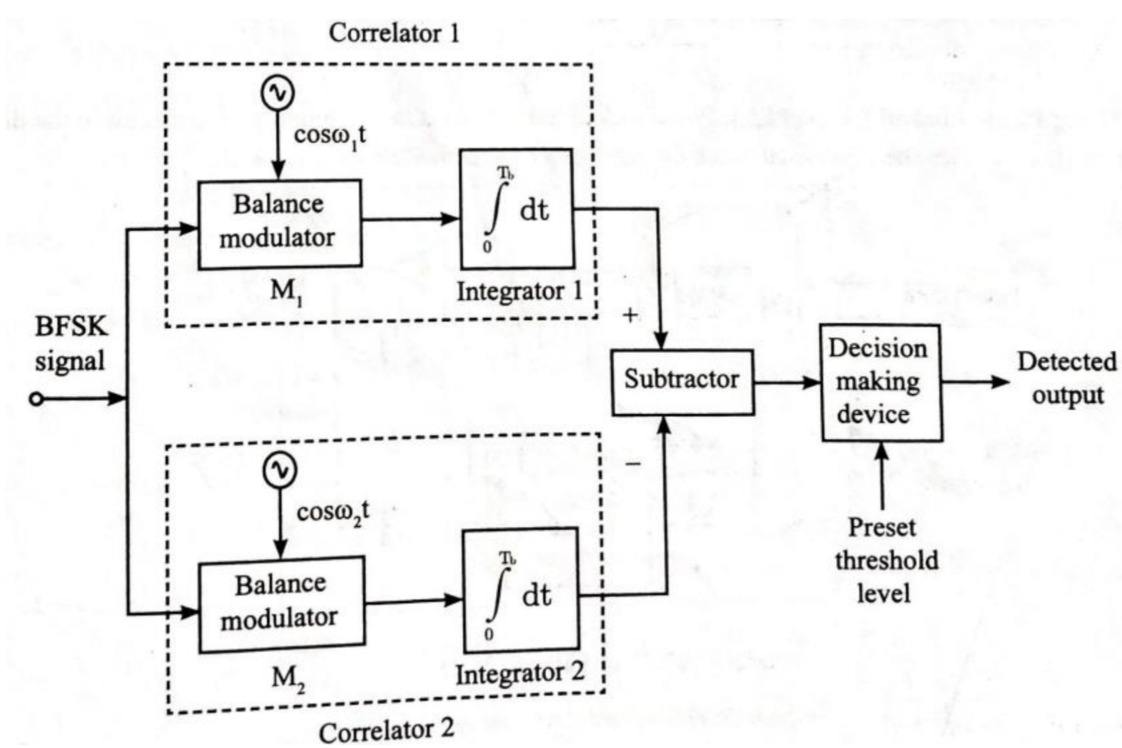
8 a) Explain the Binary Frequency shift keying in detail.

BFSK Modulator: Figure illustrates the functional block diagram of a BFSK Modulator.



It can be observed from figure (a) that the digital input data is processed through a bipolar NRZ encoder. This encoded signal is fed to two independent balance modulator (M_1 and M_2) which multiply the signal with a high frequency carrier signal. A linear adder circuit is employed to add the outputs of the two balance modulators. Thus, a BFSK signal is obtained is obtained at the adder circuit which has a frequency shift from f_1 to f_2.

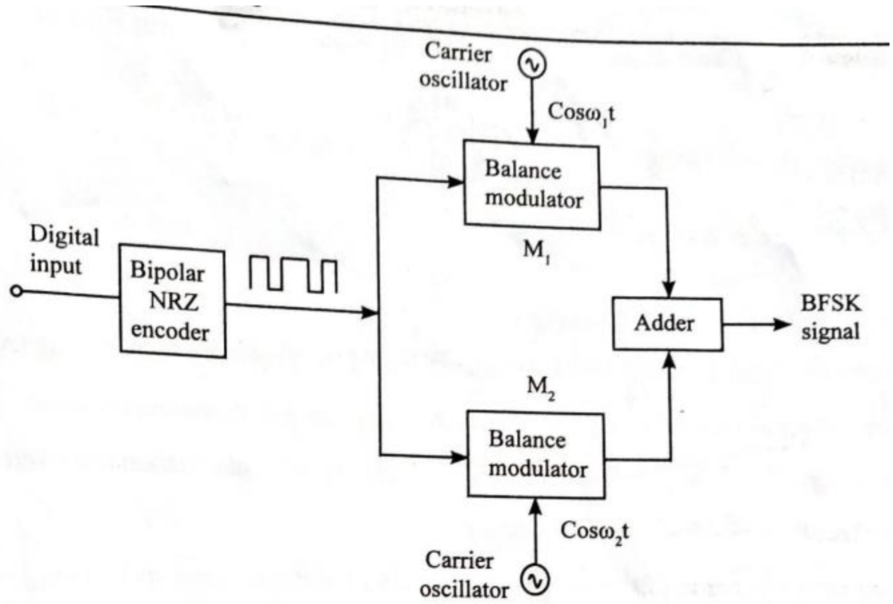
BFSK Demodulator: A BFSK signal can be recovered at the receiver by employing BFSK demodulator figure illustrates the block diagram of a BFSK demodulator.



It can be observed from figure that a BFSK demodulator consists of two correlators a subtraction and a decision making device. Each correlator has one balance modulator and one integrator circuit. A subtractor is employed to perform subtraction operation of two correlator outputs. The output of the subtractor is then compared with a preset threshold level (usually a zero volt) if using decision value. The output detected is equal to binary 1 if the compared signal greater than 0 volt. On the other hand, the output is equal to binary 0 if the signal is less than 0 volt.

8 b) Explain with suitable waveforms Binary Frequency Shift Keying.

BFSK Modulator: Figure illustrates the functional block diagram of a BFSK Modulator.



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The mathematical representation of a BFSK signal is given as,

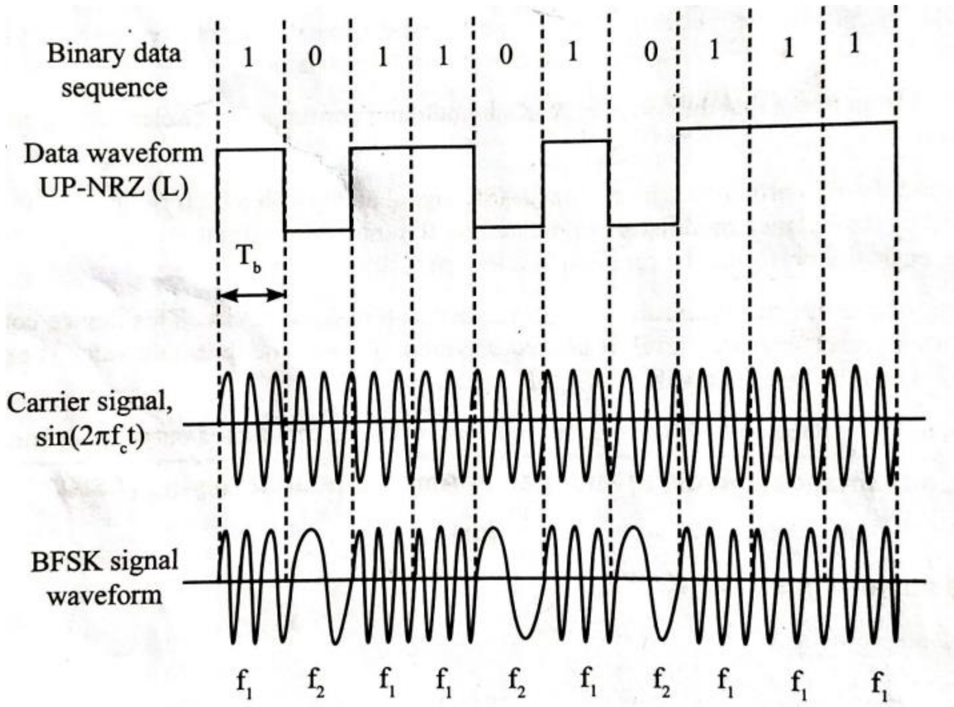
$$v_{BFSK}(t) = V_c \cos(2\pi f_1 t) \text{ for binary } 1$$

$$V_c \cos(2\pi f_2 t) \text{ for binary } 0$$

Where V_c – Maximum amplitude of carrier signal

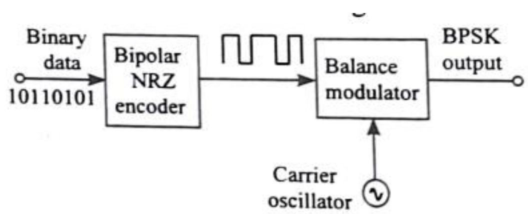
f_1, f_2 – Offset frequencies from the carrier frequency (f_c).

Illustrates the waveforms of a BFSK signal for input digital data and carrier sinusoidal signal.



9 a) Explain the Binary Phase Shift Keying modulator and demodulator.

BPSK Modulator: Figure illustrates the functional block diagram of a BPSK modulator.



It can be observed from figure that the binary input data is converted into its corresponding bipolar NRZ signals using NRZ encoder. This output is fed to the balance modulator along with a higher frequency sinusoidal carrier signal. The balance modulator produces BPSK output signal depending on the phase relationship with the carrier oscillator.

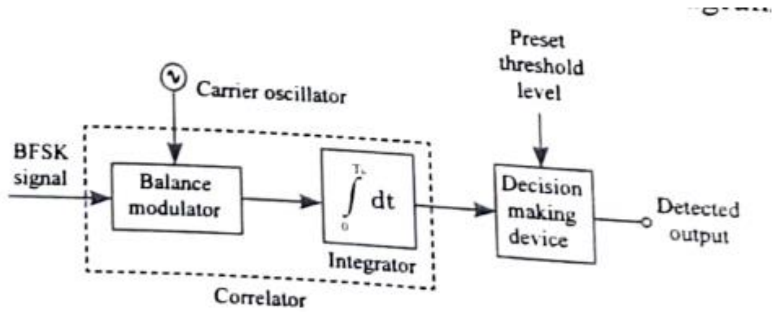
- For binary data 1- Modulator output is in phase with the reference carrier oscillator.
- For binary data 0- Modulator output is 180^o out of phase with reference carrier oscillator.

In a Binary Shift Keying(BPSK) technique, the phase of a carrier sinusoidal signal is varied in accordance with the applied digital data input.

The phase of the sinusoidal carrier signal is usually varied from 0^o to 180^o. BPSK is known as biphase modulation or phase reversal keying technique.

BPSK signal undergoes a phase shift from 0^o to 180^o whenever a transition occurs at the digital binary data input.

BPSK Demodulator: figure illustrates the functional block diagram of a BPSK demodulator



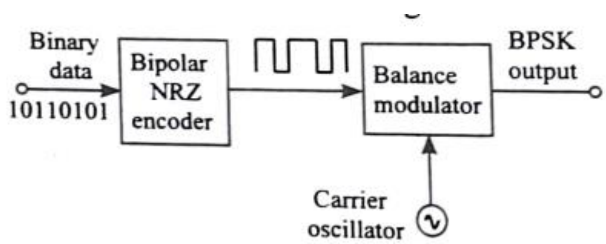
It can be observed from figure that the BPSK is fed to the correlator circuit which comprises one balance modulator and one integrator.

A carrier signal generated from carrier oscillator is also provided as input to the balance modulator. The correlator output is then compared with a preset threshold level (usually zero volt) using decision making device.

- The detected output is equal to 1 when the input of decision making device is greater than 0 volt.
- The detected output is equal to 0 when the input of decision making is less than 0 volt.

9 b) Explain with suitable waveforms Binary Phase Shift Keying.

BPSK Modulator: Figure illustrates the functional block diagram of a BPSK modulator.



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The phase of the sinusoidal carrier signal is usually varied from 0° to 180°. BPSK is known as biphase modulation or phase reversal keying technique.

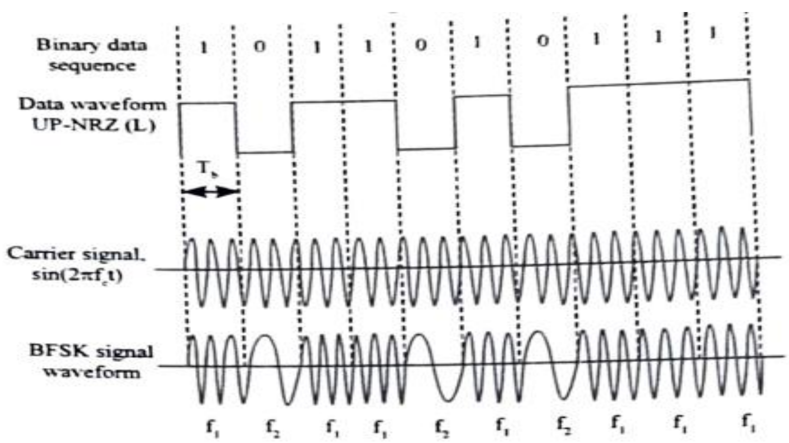
The mathematical representation of BPSK signal is given as,

$$v_{\text{BPSK}} = v_C \sin(2\pi f_c t) \text{ for binary 1}$$
$$= v_C \sin(2\pi f_c t + \phi) \text{ for binary 0}$$

Where, V_c – Maximum amplitude of carrier signal

f_c – Carrier frequency

BPSK signal undergoes a phase shift from 0° to 180° whenever a transition occurs at the digital binary data input.



10 a) Explain Slope overload distortion & Granular Noise.

Slope overload distortion:

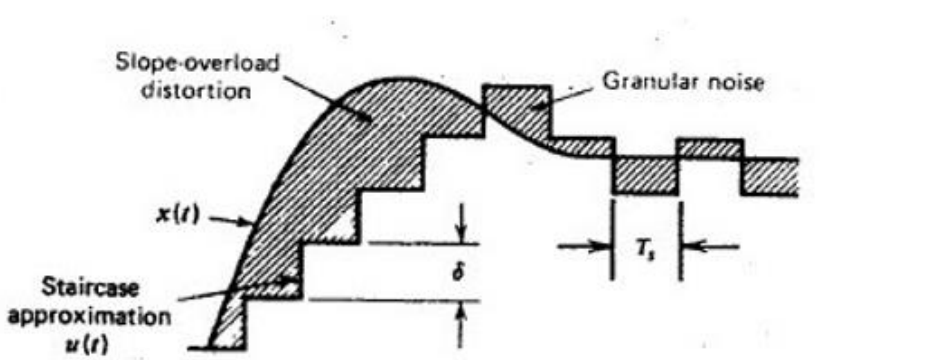


Illustration of quantization error in delta modulation.

- Slope overload distortion arises because of the large dynamic range of the input signal.

- In figure it can be seen that the rate of rise of input signal x(t) is so high that the staircase signal cannot approximate it the step size δ becomes the small stair case signal x(t) to follow the stop segment of x(t). thus large error between the stair case approximated signal and the original input signal x(t). this error is called slope overloaded distortion.
- To reduce this error, the step size should be increased when slope of the signal x(t) is high.

That is, slope of the staircase u(t)≥ slope of the message signal

$$\delta/_{Ts} \geq \max [\ d/dt\ x(t)]$$

Granular Noise:

- This noise occurs when the step size is too large compared to small variations in the input signal i.e. for very small variations in the input signal, the staircase signal is changed by large amount because of large step size δ.
- The error between the input and approximated signal is called Granular noise. The solution of this problem is to make step size small.

10 b) Compare ASK, FSK, and PSK.

ASK	PSK	FSK
In Amplitude Shift Keying (ASK), amplitude of analog carrier signal is varied in accordance with the digital data.	In ~~Frequency~~ Phase shift Keying (PSK), ~~frequency~~ Phase of analog carrier signal is varied in accordance with the digital data	In frequency shift keying (FSK), frequency of analog carrier signal is varied in accordance with the digital data.
Frequency and Phase of the carrier signal remain constant	Amplitude and frequency of the carrier signal remain constant	Amplitude and phase of the carrier signal remain constant

ASK modulation technique is less immune to noise	PSK modulation technique has better noise immunity than that of ASK and FSK	FSK modulation technique has good noise immunity
ASK system is simple to design	PSK system has high system complexity	FSK system has moderate system complexity
ASK system have high probability of occurrence of error	In PSK systems, probability of occurrance of error is comparatively low	In FSK systems, probability of occurrance of error is comparatively low.

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QUESTION BANK (DESCRIPTIVE)

Subject with Code: ICS(20EC0451) **Course& Branch**: B.Tech & CSE,CSM.CIC

Year & Sem: III-B.Tech.& I-Sem. **Regulation:** R20

UNIT-V
Introduction to Wireless Communication Systems

1 a) Discuss briefly about the evolution of Mobile radio communication.

- Wireless communications is enjoying its fastest growth period in history, due to enabling technologies which permit widespread deployment.
- The ability to provide wireless communications to an entire population was not evenconceived until Bell Laboratories developed the cellular concept in the 1960s and 1970s.
- With the development of highly reliable, miniature, solid-state radio frequency hardware in the 1970s, the wireless communications era was born.
- Following figure illustrates how mobile telephony has penetrated our daily lives compared with other popular inventions of the 20th century.

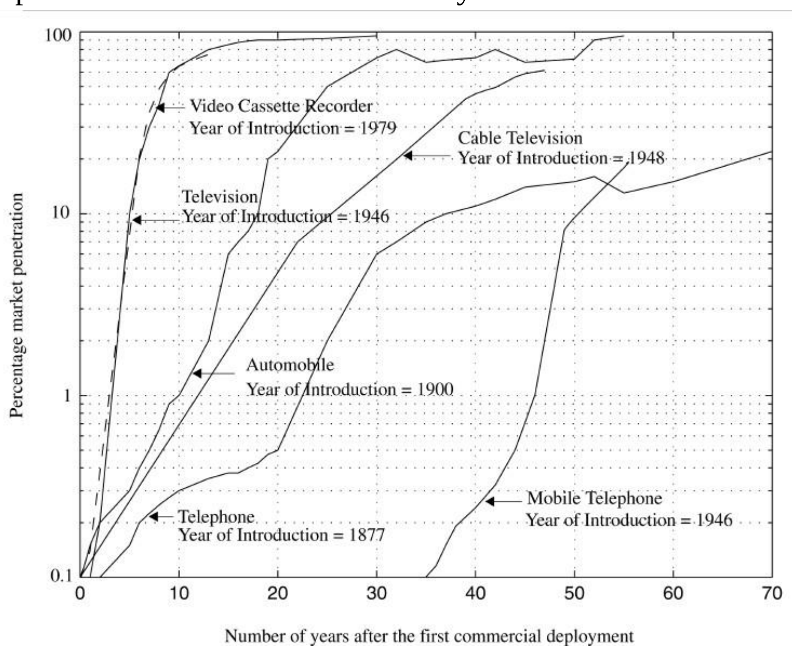


Fig: The growth of mobile telephony as compared with other popular inventions of the20th century.

- In 1935, Edwin Armstrong demonstrated frequency modulation (FM) for the firsttime.
- Since the late 1930s, FM has been the primary modulation technique used for mobile communication systems throughout the world.
- The vast majority of mobile users in the 1960s were not connected to the public switched telephone network (PSTN), and thus were not able to directly dial telephonenumbers from their vehicles.
- With the boom in CB radio and cordless appliances such as garage door openers and telephones, the number of users of mobile and portable radio in 1995 was about 100million.
- In the first few years of the 21st century, it is clear there will be an equal number of wireless and conventional wireline customers throughout the world.
- At the beginning of the 21st century, over 1% of the worldwide wireless subscriber population had already abandoned wired telephone service for home use, and had begun to rely solely on their cellular service provider for telephone access.

1 b) Explain second generation (2G) cellular networks.

- Unlike first generation cellular systems that relied exclusively on FDMA/FDD and analog FM, second generation standards use digital modulation formats and TDMA/FDD and CDMA/FDD multiple access techniques.
- The most popular second generation standards include the following three TDMA standards and one CDMA standard:
- (a) Global System Mobile (GSM), which supports eight time slotted users for each 200 kHz radio channel and has been deployed widely by service providers in Europe,Asia, Australia, South America, and some parts of the US.
- (b) Interim Standard 136 (IS-136), also known as North American Digital Cellular (NADC), which supports three time slotted users for each 30 kHz radio channel andis a popular choice for carriers in North America, South America, and Australia.
- (c) Pacific Digital Cellular (PDC), a Japanese TDMA standard that is similar to IS- 136 with more than 50 million users and
- (d) the popular 2G CDMA standard Interim Standard 95 Code Division Multiple Access (15-95), also known as cdma-One, which supports up to 64 users that are orthogonally coded and simultaneously transmitted on each 1.25MHz. channel.
- CDMA is widely deployed by carriers in North America, as well as in Korea, Japan,China, South America, and Australia.
- Second generation systems were first introduced in the early 1990s, and evolved fromthe first generation of analog mobile phone systems (e.g., AMPS, ETACS, and JTACS).
- Following figure illustrates how the world subscriber base was divided between the 1G and 2G technologies of late 2001.

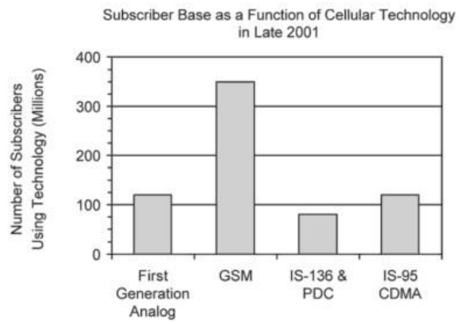


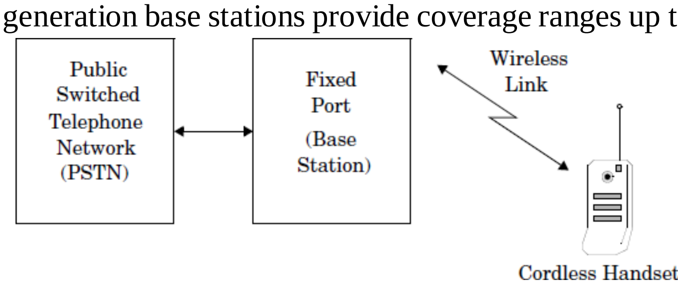
Fig: Worldwide subscriber base as a function of cellular technology (2001)

- In many countries, 2G wireless networks are designed and deployed for conventionalmobile telephone service, as a high capacity replacement for, or in competition with,existing older first generation cellular telephone systems.
- Modern cellular systems are also being installed to provide fixed (non-mobile) telephone service to residences and businesses in developing nations—this is particularly cost effective for providing plain old telephone service (POTS).
- The 2G technologies offer at least a three-times increase in spectrum efficiency as compared to first generation analog technologies, the need to meet a rapidly growingcustomer base justifies the gradual, ongoing change out of analog to digital 2G technologies in any growing wireless network.

2 a) Explain cordless telephone systems.

Cordless Telephone Systems:

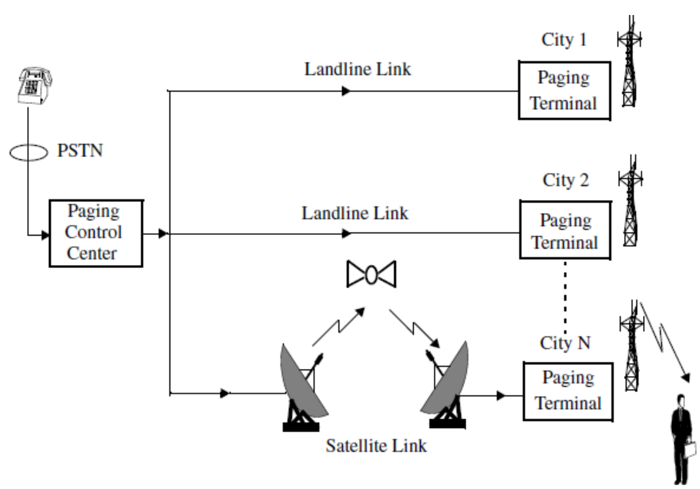
- Cordless telephone systems are full duplex communication systems that use radio toconnect a portable handset to a dedicated base station, which is then connected to a dedicated telephone line with a specific telephone number on the public switched telephone network (PSTN).
- In first generation cordless telephone systems, the portable unit communicates only to the dedicated base unit and only over distances of a few tens of meters.
- Early cordless telephones operate solely as extension telephones to a transceiver connected to a subscriber line on the PSTN and are primarily for in-home use.
- Second generation cordless telephones allow subscribers to use their handsets at many outdoor locations within urban centers.
- Modern cordless telephones are sometimes combined with paging receivers so that a subscriber may first be paged and then respond to the page using the cordless telephone.
- Cordless telephone systems provide the user with limited range and mobility, as it isusually not possible to maintain a call if the user travels outside the range of the basestation.
- Typical second generation base stations provide coverage ranges up to a few hundredmeters.



2 b) Explain paging systems.

Paging Systems:

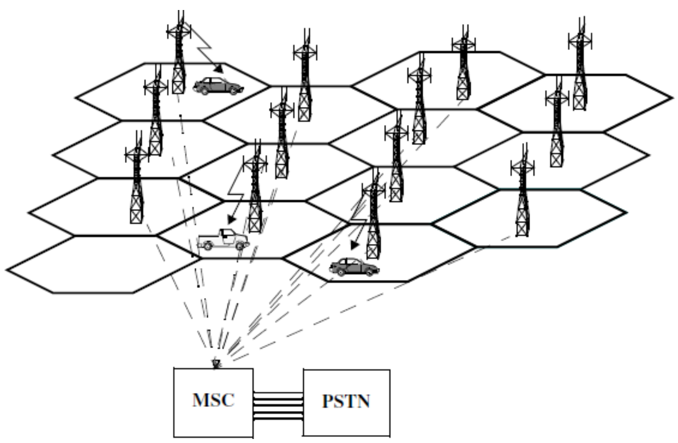
- Paging systems are communication systems that send brief messages to a subscriber.
- Depending on the type of service, the message may be either a numeric message, an alphanumeric message, or a voice message.
- Paging systems are typically used to notify a subscriber of the need to call a particular telephone number or travel to a known location to receive further instructions.
- In modern paging systems, news headlines, stock quotations, and faxes may be sent.
- A message is sent to a paging subscriber via the paging system access number (usually a toll-free telephone number) with a telephone keypad or modem. The issuedmessage is called a page.
- The paging system then transmits the page throughout the service area using base stations which broadcast the page on a radio carrier.
- Paging systems vary widely in their complexity and coverage area.
- While simple paging systems may cover a limited range of 2 to 5 km.
- Wide area paging systems can provide worldwide coverage.
- Wide area paging systems consist of a network of telephone lines, many base station transmitters, and large radio towers that simultaneously broadcast a page from each base station (this is called simulcasting).
- Paging systems are designed to provide reliable communication to subscriberswherever they are; whether inside a building, driving on a highway, or flying in an airplane.
- This necessitates large transmitter powers (on the order of kilowatts) and low data rates (a couple of thousand bits per second) for maximum coverage from each base station.



3 a) Explain cellular telephone system.

- A cellular telephone system provides a wireless connection to the PSTN for any userlocation within the radio range of the system.

- Cellular systems accommodate a large number of users over a large geographic area,within a limited frequency spectrum.
- Cellular radio systems provide high quality service that is often comparable to thatof landline telephone systems.
- High capacity is achieved by limiting the coverage of each base station transmitter toa small geographic area called a cell.
- A sophisticated switching technique called a handoff enables a call to proceed uninterrupted when the user moves from one cell to another.
- The following figure shows a basic cellular system that consists of mobile stations,base stations and a mobile switching center (MSC).



- The mobile switching center is sometimes called a mobile telephone switching office(MTSO), since it is responsible for connecting all mobiles to the PSTN in a cellular system.
- Each mobile communicates via radio with one of the base stations and may be handed-off to any number of base stations throughout the duration of a call.
- The mobile station contains a transceiver, an antenna, and control circuitry, and maybe mounted in a vehicle or used as a portable hand-held unit.
- The base stations consist of several transmitters and receivers which simultaneouslyhandle full duplex communications.
- The base station serves as a bridge between all mobile users in the cell and connectsthe simultaneous mobile calls via telephone lines or microwave links to the MSC.
- The MSC coordinates the activities of all of the base stations and connects the entirecellular system to the PSTN.
- A typical MSC handles 100,000 cellular subscribers and 5,000 simultaneous conversations at a time, and accommodates all billing and system maintenance functions, as well.
- In large cities, several MSCs are used by a single carrier.

3 b) Discuss about frequency division duplexing in wireless communication.

Frequency division duplex (FDD):

- Full-duplex systems, allow simultaneous radio transmission and reception between a subscriber and abase station, by providing two simultaneous but separate channels (frequency division duplex, or FDD) or adjacent time slots on a single radio channel (time division duplex, or TDD) for communication to and from the user.

- FDD provides simultaneous radio transmission channels for the subscriber and the base station, so thatthey both may constantly transmit while simultaneously receiving signals from one another.
- At the base station, separate transmit and receive antennas are used to accommodate the two separatechannels.
- At the subscriber unit, a single antenna is used for both transmissions to and reception from the base station.
- A device called a duplexer is used inside the subscriber unit to enable the same antenna to be used forsimultaneous transmission and reception.
- To facilitate FDD, it is necessary to separate the transmit and receive frequencies by about 5% of thenominal RF frequency.
- FDD is used exclusively in analog mobile radio systems.

4 a) Explain third generation (3G) wireless networks.

Third Generation (3G) Wireless Networks

- 3G systems promise unparalleled wireless access in ways that have never been possible before.
- Multi-megabit Internet access, communications using Voice over Internet Protocol (VoIP), voice-activated calls, unparalleled network capacity, and ubiquitous “always-on” access are just some of the advantages being touted by 3G developers.
- Companies developing 3G equipment cenvision users having the ability to receive live music, conduct interactive web sessions, and have simultaneous voice and data access with multiple parties at the same time using a single mobile handset.
- The International Telecommunications Union (ITU) formulated a plan to implementa global frequency band in the 2000 MHz range that would support a single,ubiquitous wireless communication standard for all countries throughout the world.
- This plan, called International Mobile Telephone 2000 (IMT-2000), has been successful in helping to cultivate active debate and technical analysis for new high speed mobile telephone solutions when compared to 2G.
- However, as can be seen in following figure, the hope for a single worldwide standardhas not materialized, as the world-wide user community remains spit between two camps: GSM/IS-136/PDC and CDMA.

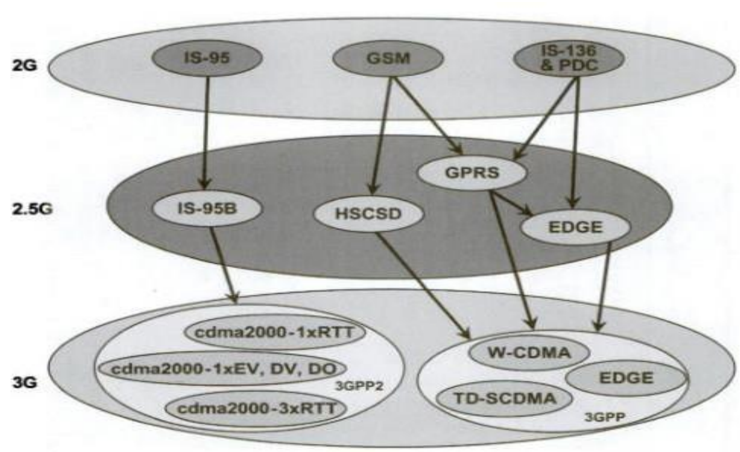


Fig: Various upgrade paths for 2G-3G technologies

- The eventual 3G evolution for CDMA systems leads to cdma2000. Several variantsof CDMA 2000 are currently being developed, but they all are based on the fundamentals of 1-95 and IS-95B technologies.
- The eventual 3G evolution for GSM, IS-136, and PDC systems leads to Wideband CDMA (W-CDMA), also called Universal Mobile Telecommunications Service (UMTS).
- W-CDMA is based on the network fundamentals of GSM, as well as the merged versions of GSM and IS-136 through EDGE.
- The ITU IMT-2000 standards organizations are currently separated into two major organizations reflecting the two 3G camps:
- (i) 3GPP (3G Partnership Project for Wideband CDMA standards based on backward compatibility with GSM and IS-136/PDC) and
- (ii) 3GPP2 (3G Partnership Project for cdma2000 standards based on backward compatibility with IS-95).
- Countries throughout the world are currently determining new radio spectrum bands to accommodate the 3G networks that will likely be deployed in the 2004-2005 timeframe.
- ITU's 2000 World Radio Conference established the 2500-2690 MHz, 1710-1885 MHz, and 806-960 MIHZ bands as candidates for 3G.
- In the US, additional spectrum in the upper UHF television bands near 700 MHz is also being considered for 3G.

4 b) A spectrum of 30 MHz of bandwidth is allocated to a particular FDD cellular telephone system which uses two 25 kHz simplex channels to provide full duplex voice and control channels, compute the number of channels available per cell if a system uses (i) four-cell reuse, (ii) seven-cell reuse, and (iii) 12-cell reuse. If 1 MHz of the allocated spectrum is dedicated to control channels, determine an equitable distribution of control channels and voice channels in each cell for each of the three systems.

Given, Total bandwidth=30 MHz
Channel bandwidth = 25 kHz × 2 simplex channels = 50 kHz/duplex channelTotal available channels = 30,000/50 = 600 channels
(a) For N = 4,
Total number of channels available per cell = 600/4 = 150 channels.
(b) For N = 7,
Total number of channels available per cell = 600/7 = 85 channels.
(c) For N = 12,
Total number of channels available per cell = 600/12 = 50 channels.

1 MHz spectrum for control channels implies that there are 1000/50 = 20 control channels out of the 600 channels available. To evenly distribute the control and voice channels, simplyallocate the same number of voice channels in each cell wherever possible.

(a) For N = 4, we can have five control channels and 145 voice channels per cell. In practice, however, each cell only needs a single control channel (the control channelshave a greater reuse distance than the voice channels). Thus, one control channel and145 voice channels would be assigned to each cell.

(b) For N = 7, four cells with three control channels and 82 voice channels [(600- 20)/7=82], two cells with three control channels and 90 voice channels, and one cellwith two control channels and 92 voice channels could be allocated.

(c) For N = 12, we can have eight cells with two control channels and 48 voice channels,and four cells with one control channel and 49 voice channels each. In an actual system, each cell would have one control channel, eight cells would have 48 voicechannels, and four cells would have 49 voice channels.

5 a) Explain the multiple access schemes for narrowband systems.

Multiple access techniques (FDMA, TDMA, CDMA & SDMA) can be grouped as narrowband and wideband systems, depending on how the available bandwidth is allocated to the users.

Narrowband Systems:

- The term narrowband is used to relate the bandwidth of a single channel to the expected coherence bandwidth of the channel.
- In a narrowband multiple access system, the available radio spectrum is divided into a large number of narrowband channels.
- The channels are usually operated using FDD.
- To minimize interference between forward and reverse links on each channel, the frequency separation is made as great as possible within the frequency spectrum.
- In narrowband FDMA, a user is assigned a particular channel which is not shared by otherusers in the vicinity, and if FDD is used (that is, each duplex channel has a forward and reverse simplex channel), then the system is called FDMA/FDD.
- Narrowband TDMA, on the other hand, allows users to share the same radio channel but allocates a unique time slot to each user in a cyclical fashion on the channel, thus separating a small number of users in time on a single channel.
- For narrowband TDMA systems, there generally are a large number of radio channels allocated using either FDD or TDD, and each channel is shared using TDMA. Such systems are called TDMA/FDD or TDMA/TDD access systems.

5 b) Discuss about time division duplexing in wireless communication.

- Time division multiple access (TDMA) systems divide the radio spectrum into time slots,and in each slot only one user is allowed to either transmit or receive.
- It can be seen from figure that each user occupies a cyclically repeating time slot, so a channel may be thought of as a particular time slot that reoccurs every frame, where N time slots comprise a frame.

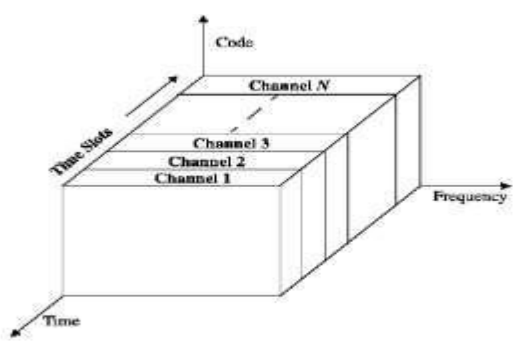


Figure 3 TDMA scheme where each channel occupies a cyclically repeating time slot.

- TDMA systems transmit data in a buffer-and-burst method, thus the transmission for anyuser is non-continuous.
- The transmission from various users is interlaced into a repeating frame structure as shownin Figure 4. It can be seen that a frame consists of a number of slots. Each frame is madeup of a preamble, an information message, and tail bits.
- Guard times are utilized to allow synchronization of the receivers between different slots and frames.

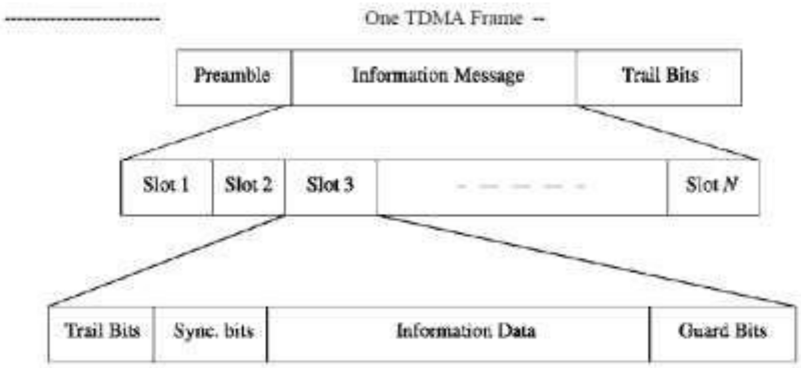


Figure 4 TDMA frame structure. The frame is cyclically repeated over time.

6 a) Explain the multiple access schemes for wideband systems.

Wideband system:

- In wideband systems, the transmission bandwidth of a single channel is much larger thanthe coherence bandwidth of the channel.
- Thus, multipath fading does not greatly vary the received signal power within a wideband channel, and frequency selective fades occur in only a small fraction of the signal bandwidth at any instance of time.
- In wideband multiple access systems, a large number of transmitters are allowed to transmit on the same channel.
- TDMA allocates time slots to the many transmitters on the same channel and allows onlyone transmitter to access the channel at any instant of time, whereas spread spectrum CDMA allows all of the transmitters to access the channel at the same time.
- TDMA and CDMA systems may use either FDD or TDD multiplexing techniques.

6 b) Draw the TDMA frame structure and briefly explain the fields.

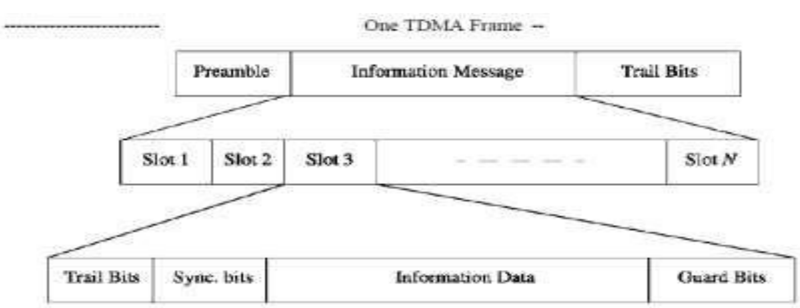


Figure 4 TDMA frame structure. The frame is cyclically repeated over time.

- Preamble consists of Address and synchronization bits used by base stations and mobile stations to identity each other.
- Information message consists of slots(users)-user information is located.
- Each slot will have trail bits, sync bits, Information data and guard bits.
- Trail bits helps in power control mechanism which help in transmission of the signal.
- Sync bits helps in synchronizing the transmitting and receiver at the time of communications.
- Information data for each slot have the data about that user.
- Guard bits are guard bands between the user to avoid interference between the users.

7 a) Describe the features of the frequency division multiple access (FDMA) scheme.

- Frequency division multiple access (FDMA) assigns individual channels to individual users.
- It can be seen from figure that each user is allocated a unique frequency band or channel. These channels are assigned on demand to users who request service.

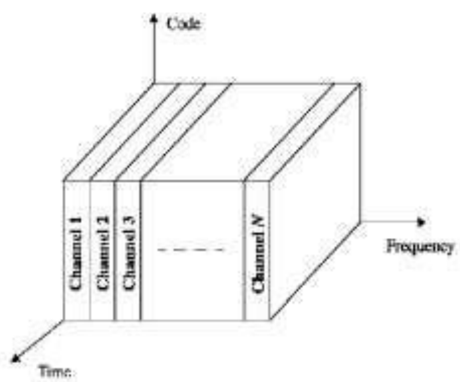


Figure: FDMA where different channels are assigned different frequency bands

- During the period of the call, no other user can share the same channel.
- In FDD systems, the users are assigned a channel as a pair of frequencies; one frequencyis used for the forward channel, while the other frequency is used for the reverse channel.

The features of FDMA are as follows:

- The FDMA channel carries only one phone circuit at a time.
- If an FDMA channel is not in use, then it sits idle and cannot be used by other users to increase or share capacity. It is essentially a wasted resource.
- After the assignment of a voice channel, the base station, and the mobile transmit simultaneously and continuously.
- The bandwidths of FDMA channels are relatively narrow (30 kHz in AMPS) as each channel supports only one circuit per carrier. That is, FDMA is usually implemented in narrowband systems.
- The complexity of FDMA mobile systems is lower when compared to TDMA systems, though this is changing as digital signal processing methods improve for TDMA.
- Since FDMA is a continuous transmission scheme, fewer bits are needed for overhead purposes (such as synchronization and framing bits) as compared to TDMA.

- FDMA systems have higher cell site system costs as compared to TDMA systems, becauseof the single channel per carrier design.
- The FDMA mobile unit uses duplexers since both the transmitter and receiver operate at the same time.
- This results in an increase in the cost of FDMA subscriber units and base stations.
- FDMA requires tight RF filtering to minimize adjacent channel interference.

7 b) Describe the features of code division multiple access (CDMA) scheme.

- In code division multiple access (CDMA) systems, the narrowband message signal is multiplied by a very large bandwidth signal called the spreading signal.
- The spreading signal is a pseudo-noise code sequence that has a chip rate which is ordersof magnitudes greater than the data rate of the message.
- All users in a CDMA system, as seen from Figure, use the same carrier frequency andmay transmit simultaneously.

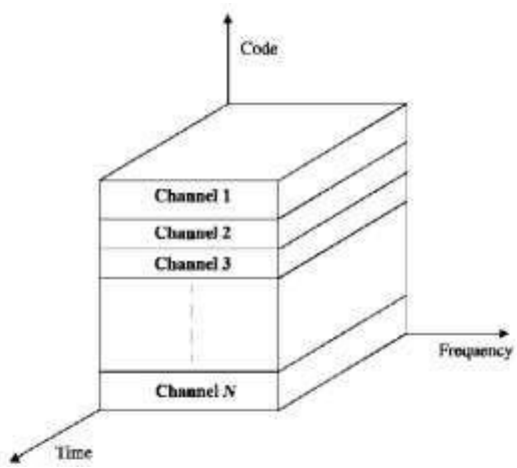


Figure: Spread spectrum multiple access in which each channel is assigned a unique PNcode which is orthogonal or approximately orthogonal to PN codes used by other users.

- Each user has its own pseudorandom code word which is approximately orthogonal to all other code words.
- The receiver performs a time correlation operation to detect only the specific desired code word. All other code words appear as noise due to de-correlation.

The features of CDMA including the following:

- Many users of a CDMA system share the same frequency. Either TDD or FDD may be used.
- Unlike TDMA or FDMA, CDMA has a soft capacity limit. Increasing the number of usersin a CDMA system raises the noise floor in a linear manner.
- Thus, there is no absolute limit on the number of users in CDMA.
- Rather, the system performance gradually degrades for all users as the number of users is increased, and improves as the number of users is decreased.
- Multipath fading may be substantially reduced because the signal is spread over a large spectrum.

- If the spread spectrum bandwidth is greater than the coherence bandwidth of the channel,the inherent frequency diversity will mitigate the effects of small-scale fading.
- Channel data rates are very high in CDMA systems.
- A RAKE receiver can be used to improve reception by collecting time delayed versionsof the required signal.
- Self-jamming is a problem in the CDMA system. Self-jamming arises from the fact thatthe spreading sequences of different users are not exactly orthogonal.
- The near-far problem occurs at a CDMA receiver if an undesired user has a high detected power as compared to the desired user.

8 a) Describe the features of time division multiple access (TDMA) scheme.

- Time division multiple access (TDMA) systems divide the radio spectrum into time slots,and in each slot only one user is allowed to either transmit or receive.
- It can be seen from figure that each user occupies a cyclically repeating time slot, so a channel may be thought of as a particular time slot that reoccurs every frame, where N time slots comprise a frame.

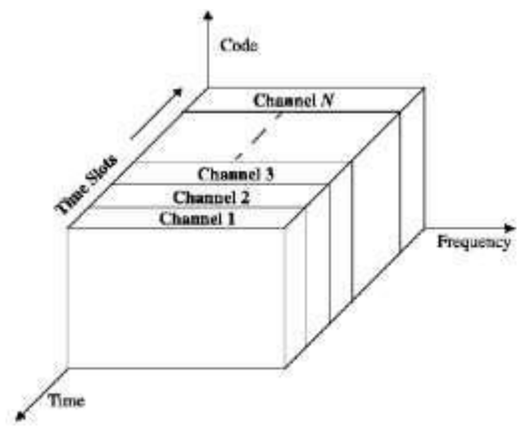


Figure 3 TDMA scheme where each channel occupies a cyclically repeating time slot.

- TDMA systems transmit data in a buffer-and-burst method, thus the transmission for anyuser is non-continuous.
- The transmission from various users is interlaced into a repeating frame structure as shown Figure 4. It can be seen that a frame consists of a number of slots. Each frame is madeup of a preamble, an information message, and tail bits.
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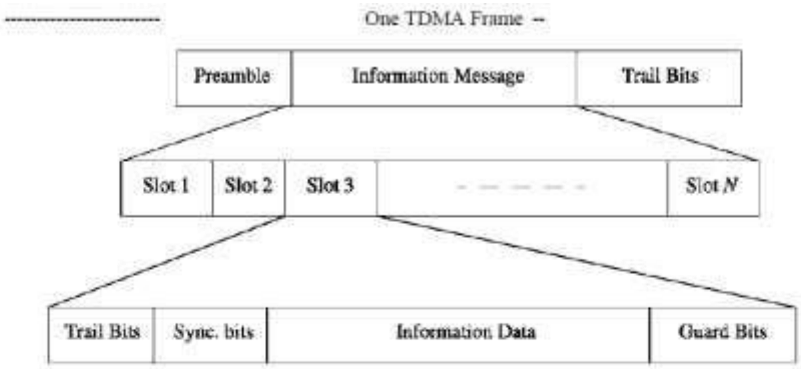

Figure 4 TDMA frame structure. The frame is cyclically repeated over time.

The features of TDMA include the following:

- TDMA shares a single carrier frequency with several users, where each user makes use of non-overlapping time slots.
- Data transmission for users of a TDMA system is not continuous but occurs in bursts.
- This results in low battery consumption, since the subscriber transmitter can be turned off when not in use (which is most of the time).
- Because of discontinuous transmissions in TDMA, the handoff process is much simplerfor a subscriber unit, since it is able to listen for other base stations during idle time slots.
- TDMA uses different time slots for transmission and reception, thus duplexers are not required.

8 b) Evaluate the efficiency of time division multiple access (TDMA) scheme.

Efficiency of TDMA:

- The efficiency of a TDMA system is a measure of the percentage of transmitted data that contains information as opposed to providing overhead for the access scheme.
- The frame efficiency η_f, is the percentage of bits per frame which contain transmitted data.
- Note that the transmitted data may include source and channel coding bits, so the raw end-user efficiency of a system is generally less than η_f.
- The frame efficiency can be found as follows.

The number of overhead bits per frame is,

$$b_{OH} = N_r b_r + N_t b_p + N_t b_g + N_r b_g$$

where N_r is the number of reference bursts per frame, N_t is the number of traffic bursts per frame, b_r is the number of overhead bits per reference burst, b_p is the number of overhead bits per preamble in each slot, and b_g is the number of equivalent bits in each guard time interval. The total number of bits per frame, b_T, is

$$b_T = T_f R$$

where T_f is the frame duration, and R is the channel bit rate. The frame efficiency η_f is thus given as

$$\eta_f = \left(1 - \frac{b_{OH}}{b_T}\right) \times 100\%$$

Number of channels in TDMA system – The number of TDMA channel slots that can be provided in a TDMA system is found by multiplying the number of TDMA slots per channel by the number of channels available and is given by

$$N = \frac{m(B_{tot} - 2B_{guard})}{B_c}$$

where m is the maximum number of TDMA users supported on each radio channel. Note that two guard bands, one at the low end of the allocated frequency band and one at the high end, are required to ensure that users at the edge of the band do not "bleed over" into an adjacent radio service.

9 a) Differentiate FDMA, TDMA and CDMA.

FDMA	TDMA	CDMA
FDMA stands for Frequency Division Multiple Access.	TDMA stands for Time Division Multiple Access.	CDMA stands for Code Division Multiple Access.
In this, sharing of bandwidth among different stations takes place.	In this, only the sharing of time of satellite transponder takes place.	In this, there is sharing of both i.e. bandwidth and time among different stations takes place.
There is no need of any code word.	There is no need of any code word.	Code word is necessary.
In this, there is only need of guard bands between the adjacent channels are necessary.	In this, guard time of the adjacent slots are necessary.	In this, both guard bands and guard time are necessary.
Synchronization is not required.	Synchronization is required.	Synchronization is not required.

FDMA	TDMA	CDMA
The rate of data is low.	The rate of data is medium.	The rate of data is high.
Mode of data transfer is continuous signal.	Mode of data transfer is signal in bursts.	Mode of data transfer is digital signal.
It is little flexible.	It is moderate flexible.	It is highly flexible.

9 b) Illustrate with a timing diagram how call initiated by a mobile user is established.

When a cellular phone is turned ON, but not yet engaged in a call, it first scans the group of forward control channels to determine the one with the strongest signal, and then monitors that control channel until the signal level drops below a usable level.

Call initiation by a landline (PSTN) subscriber to mobile user:

- The mobile switching centre (MSC) dispatches the request to all base station in a cellular system.
- The Mobile Identification Number (MIN) which is subscriber telephone number is then broadcast as a paging message over all of the forward control channels throughout the cellular system.
- The mobile receives the paging message sent by BS which s monitors, and responds by identifying itself over the RCC.
- The BS relays the acknowledgement sent by the mobile and informs the MSC of handshake.
- The MSC instructs the BS to move the call to an unused voice channel pair within the cell.
- The BS signals the mobile to change frequencies to an unused forward and reverse voice channel pair.
- Another data message is transmitted on forward channel to instruct the mobile telephone to ring and mobile user to answer the phone.
- Figure below shows sequence of events involved in call connection.

MSC		Receives call from PSTN Sends the requested MIN to all base Stations			Verifies that the mobile has a Valid MIN ESN Pair.	Requests BS to move mobile to unused voice channel pair		Connects the mobile with the calling party on the PSTN
BASE Station	FCC		Transmitts page (MIN) for specified User				Transmits data message for mobile to move to specific voice channel	
	RCC			Receives MIN ESN Station class Mark and Passes to MSC				
	FVC							Begin Voice Transmission
	RVC							Begin Voice reception
Mobile	FCC		Receives page and matches the MIN with its OWN MIN				Receives data messages to move to specified voice channel	
	RCC			Acknowledges receipt of MIN and sends ESN and station class Mark				
	FVC							Begin Voice reception
	RVC							Begin Voice Transmission

10 a) Explain various hybrid spread spectrum techniques in CDMA.

In addition to the frequency hopped and direct sequence, spread spectrum multiple access techniques, there are certain other hybrid combinations that provide certain advantages. These hybrid techniques are described below.

Hybrid FDMA/CDMA (FCDMA) — This technique can be used as an alternative to the DS-CDMA techniques discussed above. Figure 8.6 shows the spectrum of this hybrid scheme. The available wideband spectrum is divided into a number of subspectras with smaller bandwidths. Each of these smaller subchannels becomes a narrowband CDMA system having processing gain lower than the original CDMA system. This hybrid system has an advantage in that the required bandwidth need not be contiguous and different users can be allotted different subspectrum bandwidths depending on their requirements. The capacity of this FDMA/CDMA technique is calculated as the sum of the capacities of a system operating in the subspectra [Eng93].

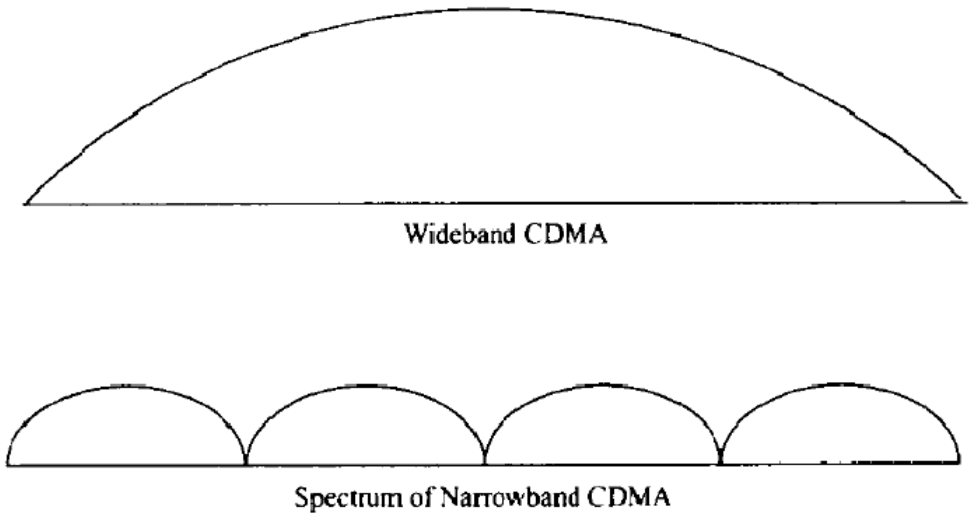

Figure 8.6
Spectrum of wideband CDMA compared to the spectrum of a hybrid, frequency division, direct sequence multiple access.

Hybrid Direct Sequence/Frequency Hopped Multiple Access (DS/FHMA) — This technique consists of a direct sequence modulated signal whose center frequency is made to hop periodically in a pseudorandom fashion. Figure 8.7 shows the frequency spectrum of such a signal [Dix94]. Direct sequence, frequency hopped systems have an advantage in that they avoid the near-far effect. However, frequency hopped CDMA systems are not adaptable to the soft handoff process since it is difficult to synchronize the frequency hopped base station receiver to the multiple hopped signals.

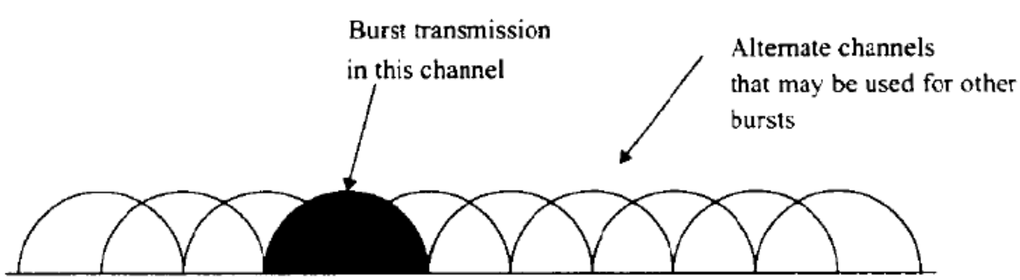

Figure 8.7
Frequency spectrum of a hybrid FH/DS system.

Time Division CDMA (TCDMA) — In a TCDMA (also called TDMA/CDMA) system, different spreading codes are assigned to different cells. Within each cell, only one user per cell is allotted a particular time slot. Thus at any time, only one CDMA user is transmitting in each cell. When a handoff takes place, the spreading code of the user is changed to that of the new cell. Using

TCDMA has an advantage in that it avoids the near-far effect since only one user transmits at a time within a cell.

Time Division Frequency Hopping (TDFH) — This multiple access technique has an advantage in severe multipath or when severe co-channel interference occurs. The subscriber can hop to a new frequency at the start of a new TDMA frame, thus avoiding a severe fade or erasure event on a particular channel. This technique has been adopted for the GSM standard, where the hopping sequence is predefined and the subscriber is allowed to hop only on certain frequencies which are assigned to a cell. This scheme also avoids co-channel interference problems between neighboring cells if two interfering base station transmitters are made to transmit on different frequencies at different times. The use of TDFH can increase the capacity of GSM by several fold [Gud92].

10 b) Describe space division multiple access (SDMA) scheme.

- Space division multiple access (SDMA) controls the radiated energy for each user in space.
- It can be seen from Figure that SDMA serves different users by using spot beam antennas.

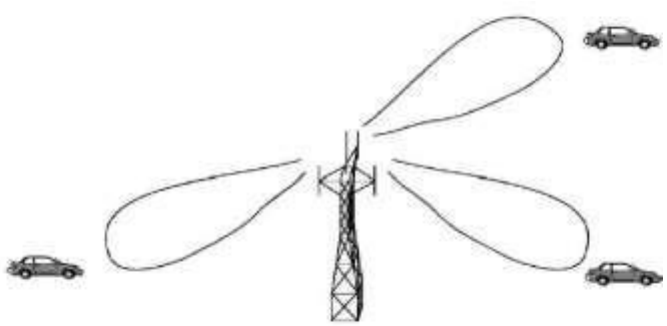

Figure: A spatially filtered base station antenna serving different users by using spot beams.

- These different areas covered by the antenna beam may be served by the same frequency(in a TDMA or CDMA system) or different frequencies (in an FDMA system).
- Sectorized antennas may be thought of as a primitive application of SDMA.
- The reverse link presents the most difficulty in cellular systems for several reasons.
- First, the base station has complete control over the power of all the transmitted signals on the forward link.
- Second, transmit power is limited by battery consumption at the subscriber unit, thereforethere are limits on the degree to which power may be controlled on the reverse link.
- If the base station antenna is made to spatially filter each desired user so that more energyis detected from each subscriber, then the reverse link for each user is improved and less power is required.
- Adaptive antennas used at the base station (and eventually at the subscriber units) promiseto mitigate some of the problems on the reverse link.
- In the limiting case of infinitesimal beam-width and infinitely fast tracking ability, adaptive antennas implement optimal SDMA, thereby providing a unique channel that isfree from the interference of all other users in the cell.
- With SDMA, all users within the system would be able to communicate at the same timeusing

the same channel.

- In addition, a perfect adaptive antenna system would be able to track individual multipath components for each user and combine them in an optimal manner to collect all of the available signal energy from each user.
- The perfect adaptive antenna system is not feasible since it requires infinitely large antennas.