

# Communication Engineering

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Tag	Year 3 Term 1

## Resources

- Ma'am's slide.

- Bangla :**

[https://www.youtube.com/watch?v=AJ-QfSSgXTg&list=PLMjaJoGgWV1nNjZXvcps2TrjdloXQxwbn&index=1&ab\\_channel=RojibEEEAcademy](https://www.youtube.com/watch?v=AJ-QfSSgXTg&list=PLMjaJoGgWV1nNjZXvcps2TrjdloXQxwbn&index=1&ab_channel=RojibEEEAcademy)  
(Recommended if you are following ma'am slide.)

- Superheterodyne Receive: [90,93]**

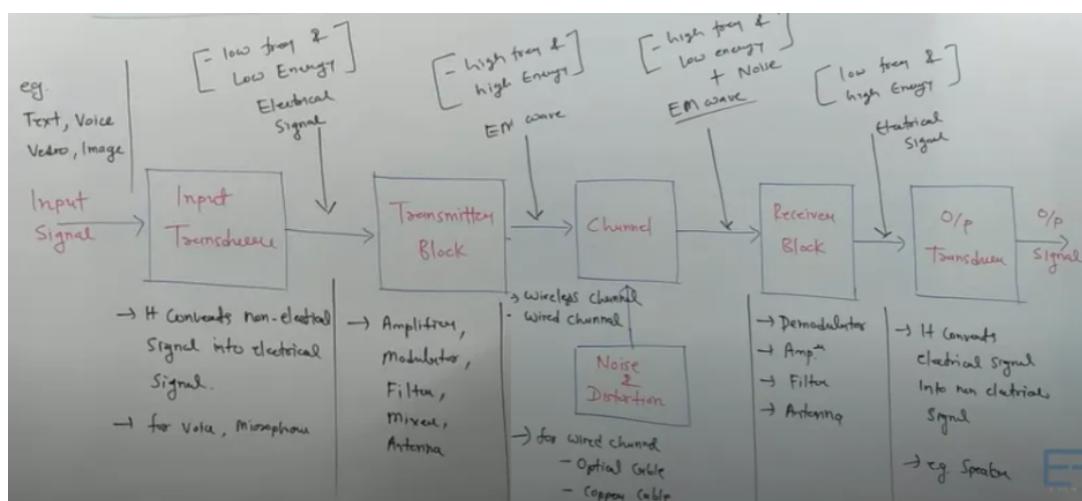
[https://www.youtube.com/watch?v=gwW0MnNVevY&list=PLc3zKsWdO93e2I625YDIXi0LlbBth1QpB&index=92&ab\\_channel=RKClasses](https://www.youtube.com/watch?v=gwW0MnNVevY&list=PLc3zKsWdO93e2I625YDIXi0LlbBth1QpB&index=92&ab_channel=RKClasses)

- English :**

[https://www.youtube.com/watch?v=kAs8OerKRmc&list=PLgwJf8NK-2e7uyUYrpgUUQowmRuKxRdw&index=1&ab\\_channel=EngineeringFunda](https://www.youtube.com/watch?v=kAs8OerKRmc&list=PLgwJf8NK-2e7uyUYrpgUUQowmRuKxRdw&index=1&ab_channel=EngineeringFunda)

- [https://www.youtube.com/watch?v=tn5NgjzNq\\_w&ab\\_channel=TechnicalGuftgu](https://www.youtube.com/watch?v=tn5NgjzNq_w&ab_channel=TechnicalGuftgu)
- [https://www.youtube.com/playlist?list=PLgluYk4ut4L1P\\_fGHgGDsyfzK28nscPcX](https://www.youtube.com/playlist?list=PLgluYk4ut4L1P_fGHgGDsyfzK28nscPcX)

## Communication System



Block Diagram

## Classification of Electric Communication System

### 1. Unidirectional system or simplex system

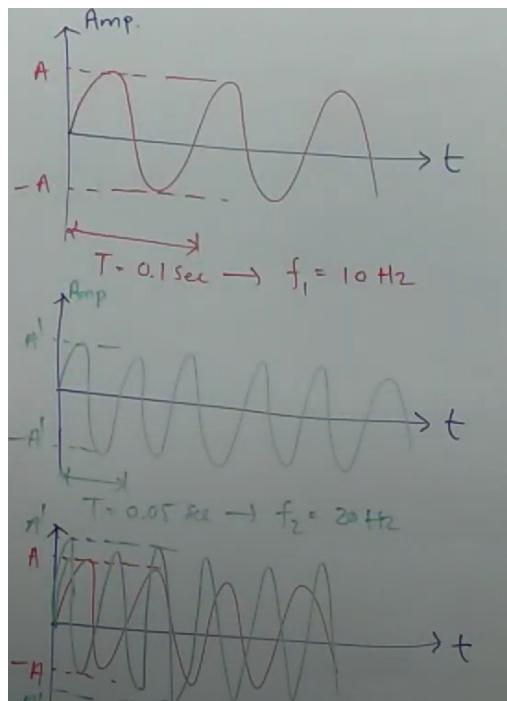
## 2. Bidirectional system or duplex system

- Half duplex system
- Full duplex system

Basis for Comparison	Simplex	Half Duplex	Full Duplex
Direction of Communication	Unidirectional	Two-directional, one at a time	Two-directional, simultaneously
Send / Receive	The sender can only send data	The sender can send and receive data, but one at a time	The sender can send and receive data simultaneously
Performance	Worst performing mode of transmission	Better than Simplex	Best performing mode of transmission
Example	Keyboard and monitor	Walkie-talkie	Telephone

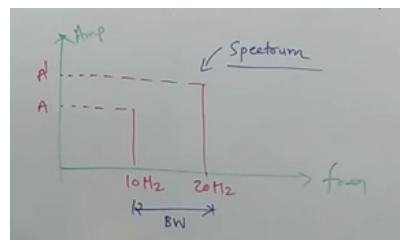
### Time Domain Signal

- It is a plot of amplitude wrt time
- multiple signals make complex



### Frequency Domain Signal

- wrt frequency
- complexity is very less



### Advantages

- Filtering
- Less complexity
- Stability
- Convolution of two signal = multiplication in frequency domain

## Modulation

It is a process of modification of carrier signal wrt modulating (message) signal.

**Carrier signal** can be defined as a high frequency signal that is modulated by the modulating or the information signal.

### Need of Modulation

1. Height of antenna : reduction of size in antenna

$$f_m, \lambda = \frac{c}{f},$$

Length of antenna,

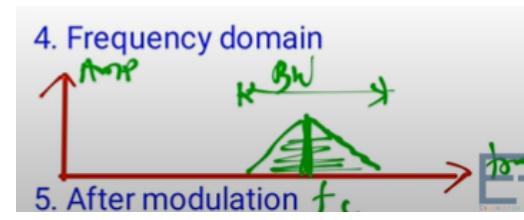
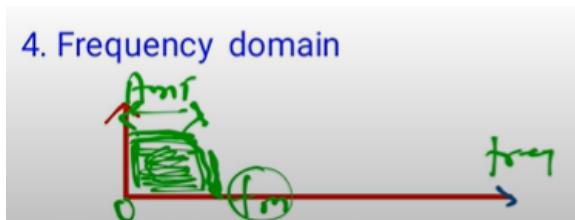
$$\text{dipole, } L = \frac{\lambda}{2}$$

$$\text{monopole, } L = \frac{\lambda}{4}$$

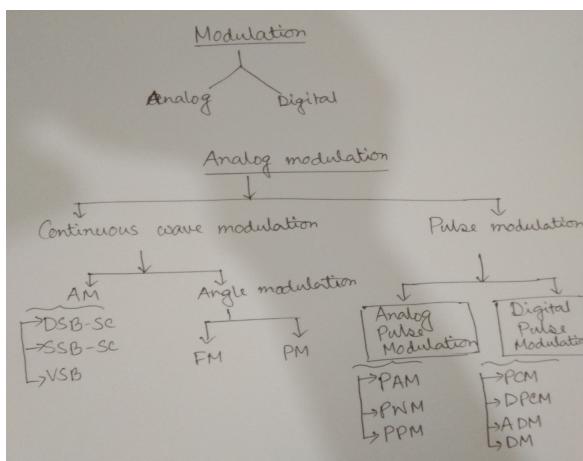
$$\text{small antenna, } L = \frac{\lambda}{50} \text{ or } \frac{\lambda}{10}$$

We can increase  $f$  to decrease the  $L$ .

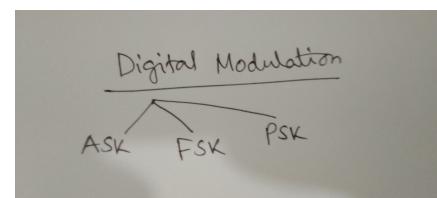
2. Radiated power by antenna:  $P_r \propto \frac{1}{\lambda^2}$ , power increases when  $f$  increases
3. Multiplexing: FDM, TDM, CDMA
4. High Bandwidth
5. Narrow Banding signal



### Classification of Modulation



Ma'am Slide



### **Amplitude Modulation**

It is the process in Amplitude of carrier signal changes wrt message/ modulating/ information signal.

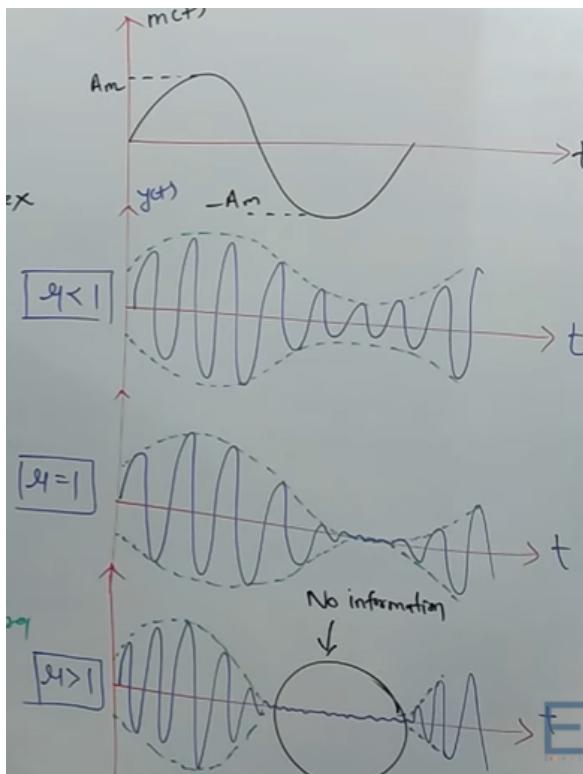
$$s(t) = A_c(1 + k_a m(t)) \cos \omega_c t$$

For single tone,

$$s(t) = A_c \cos \omega_c t + \mu \frac{A_c}{2} [\cos(\omega_c + \omega_m)t + \cos(\omega_c - \omega_m)t]$$

$$\text{where, modulation index } \mu = k_a A_m = \frac{A_m}{A_c}$$

## Modulating Index



gain,  $E_{max} = A_c[1+\mu]$  --- (iii)  
 $E_{min} = A_c[1-\mu]$  --- (iv)

(iii)  $\div$  (iv)  $\Rightarrow \frac{E_{max}}{E_{min}} = \frac{A_c[1+\mu]}{A_c[1-\mu]} = \frac{1+\mu}{1-\mu}$

$\Rightarrow \frac{E_{max} - E_{min}}{E_{max} + E_{min}} = \frac{1+\mu - 1+\mu}{1+\mu + 1-\mu} = \frac{\mu}{2}$

$\boxed{\frac{E_{max} - E_{min}}{E_{max} + E_{min}} = \mu} \quad \dots \text{(v)}$

if  $\mu = 1$  or 100%  
 if  $\mu < 1$  or 80%  
 if  $\mu > 1$  or 120%

## AM Signal Transmitted power

For the changing quantity we need to calculate average power ( $P_{avg}$ ).

$$\text{RMS value of } \cos \text{ is } = \frac{\max}{\sqrt{2}} = \frac{A_c}{\sqrt{2}}$$

$$\text{Mean square value is } = (\text{RMS})^2 = \left(\frac{A_c}{\sqrt{2}}\right)^2 = \frac{A_c^2}{2}$$

$$\text{Average value} = (\text{RMS})^2$$

$$\therefore P_{avg} = \left(\frac{A_c}{\sqrt{2}}\right)^2 + \left(\frac{A_c\mu/2}{\sqrt{2}}\right)^2 + \left(\frac{A_c\mu/2}{\sqrt{2}}\right)^2$$

From Ma'am slide.  $P_{avg} = P_t$

$$\text{Total Transmitted Power, } P_t = P_c + P_{USB} + P_{LSB} = P_c + P_s$$

Power of carrier,  $P_c = \frac{A_c^2}{2}$ , It is redundancy  $\rightarrow$  it doesn't have information

$P_s$  has information, it is called power of sideband.

$$\text{Power of Upper Sideband, } P_{USB} = \frac{1}{2} \left(\frac{A_c\mu}{2}\right)^2 = \frac{1}{8} A_c^2 \mu^2$$

$$\text{Power of Lower Sideband, } P_{LSB} = \frac{1}{8} A_c^2 \mu^2$$

$$\text{Power of sideband, } P_s = P_{USB} + P_{LSB} = \frac{1}{4} A_c^2 \mu^2 = \frac{1}{2} P_c \mu^2$$

$$\text{Total Power, } P_t = \frac{A_c^2}{2} \left(1 + \frac{\mu^2}{2}\right)$$

### Efficiency of AM Signal

It is based on  $P_s$ . Because, it has information.

$$\eta = \frac{P_s}{P_t} = \frac{\mu^2}{2+\mu^2}$$

### Redundancy

$$D = 1 - \mu = \frac{2}{2+\mu^2}$$

### DSB-SC : Double Side Band Suppressed Carrier

Transmission in which frequencies produced by amplitude modulation are symmetrically spaced above and below the carrier frequency and the carrier level is reduced to the lowest practical level, ideally completely suppressed.

- We don't send carrier signal
- Only LSB and USB are there
- It has 180 degree phase reversal at zero lossy of modulating signal

### Advantages

- Lower Power Consumption

### Disadvantages

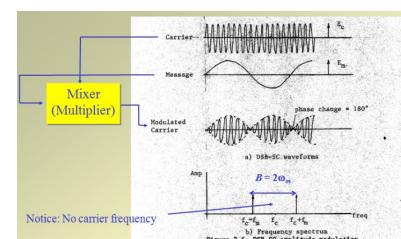
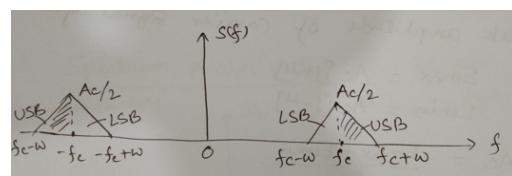
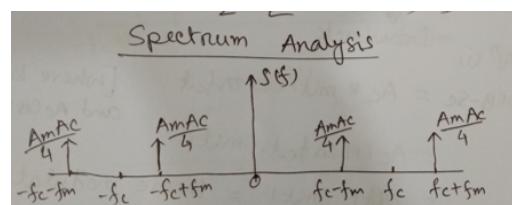
- Complex detection

### Applications

- Analogue TV

AM signal,

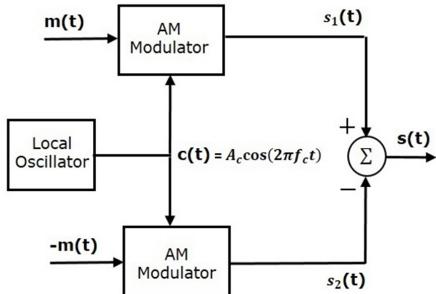
$$\begin{aligned} s(t) &= A_c[1 + k_a m(t)] \cos \omega_c t = A_c m(t) \cos \omega_c t \quad [k_a = 1, A_c \cos \omega_c = 0] \\ &= A_c A_m \cos \omega_m t \cos \omega_c t = \frac{A_m A_c}{2} \cos(\omega_c + \omega_m)t \cos(\omega_c - \omega_m)t \\ P_{avg} &= \left(\frac{\frac{A_m A_c}{2}}{\sqrt{2}}\right)^2 + \left(\frac{\frac{A_m A_c}{2}}{\sqrt{2}}\right)^2 = \frac{A_m^2 A_c^2}{4} \end{aligned}$$



### Balance Modulator DSB-SC

- Consists of two identical AM modulation

- one input  $m(t)$
- another input  $-m(t)$
- The summer block is used to sum those



$$s_1(t) = A_c[1 + k_a m(t)] \cos \omega_c t$$

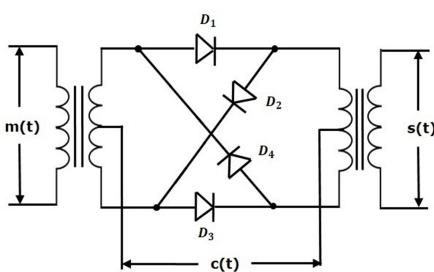
$$s_2(t) = A_c[1 - k_a m(t)] \cos \omega_c t$$

$$s(t) = s_1(t) - s_2(t)$$

$$= 2k_a m(t) \cos \omega_c t$$

By comparing with the standard form of DSB-SC, we will get the scaling factor as  $2k_a$

### Ring Modulator DSB SCs



#### For Positive half cycle

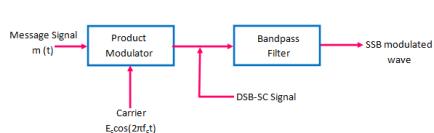
- D1, D3 = on, D2, D4 = off
- message signal multiplied by +1

#### For Negative half cycle

- D2, D4 = on, D1, D3=off
- message signal multiplied by -1

We will get DSB SC wave  $s(t)$ , which is just the product of the carrier signal  $c(t)$  and the message signal  $m(t)$ , which represents DSB-SC wave and is obtained at the output transformer of the ring modulator.

### SSB-SC : Single Sideband Suppressed Carrier



- Removes any of sideband along with carrier
- Bandwidth =  $\omega_m = f_m$

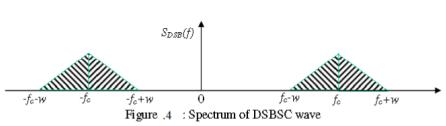


Figure 4 : Spectrum of DSBSC wave

$$s(t) = \frac{A_m A_c}{2} \cos(\omega_c + \omega_m)t$$

or,  $s(t) = \frac{A_m A_c}{2} \cos(\omega_c - \omega_m)t$

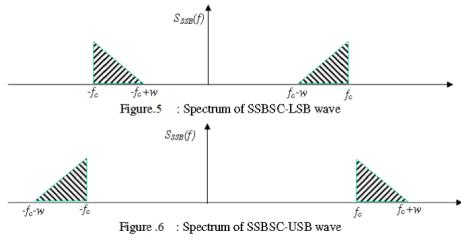


Figure 5 : Spectrum of SSBSC-LSB wave

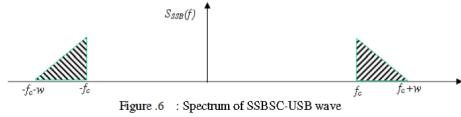
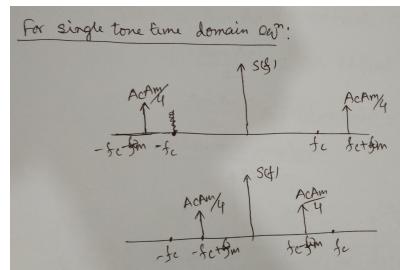
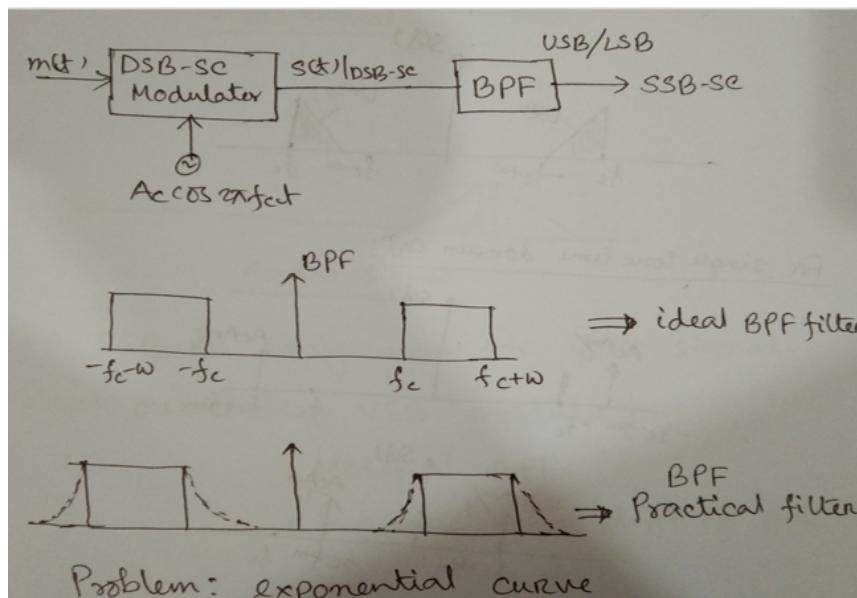


Figure 6 : Spectrum of SSBSC-USB wave

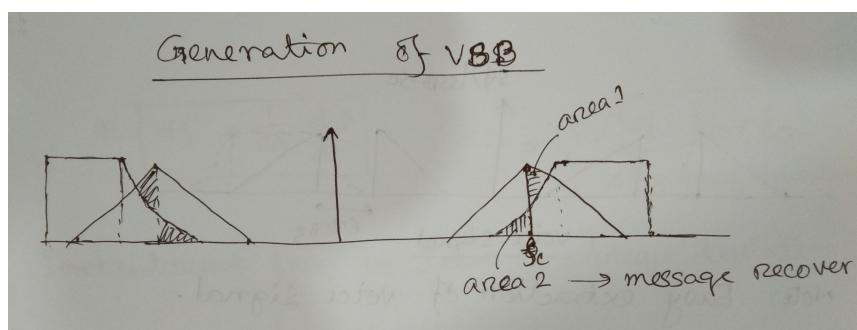


We can use SSB-SC for Audio, Video signal but voice signal. Because of gap.

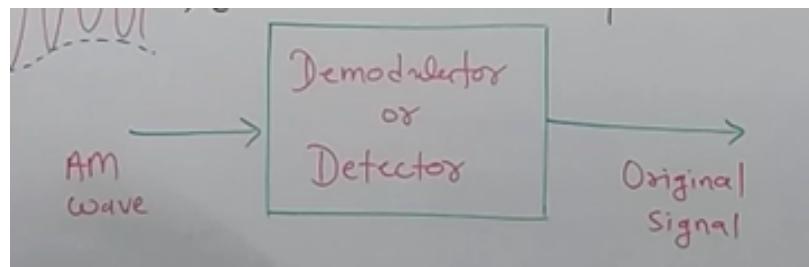


### VSB : Vestigial Sideband Generator

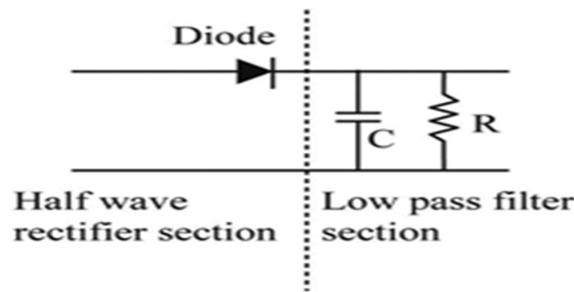
- Used to transmit video signal
- Modify the BPF of SSB-SC
- Voice : 300 - 3500 Hz (SSB-SC)
- Audio Signal (20-2000 Hz)
- Video Signal (0-4.5 MHz) (VSB)



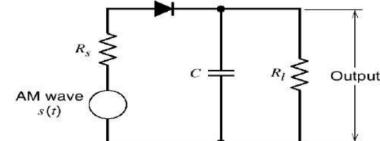
## Demodulation of AM



### Envelope Detector



- It is a simple and highly effective system. This method is used in most of the commercial AM radio receivers



### Positive Half Cycle

- Diode forward biased
- Capacitor charges rapidly
- Charge time constant  $(r_f + R_s)C$  must be shorter than the carrier period

$$(r_f + R_s)C \ll \frac{1}{f_c}$$

### Negative/ Input signal falls

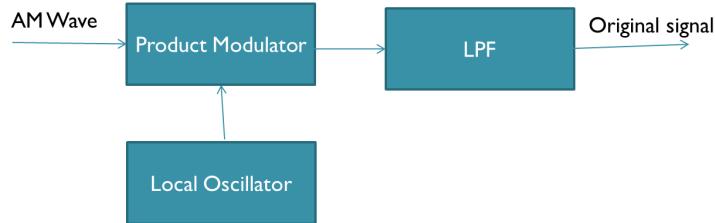
- Reverse
- Discharges through load resistor
- it continues until the positive next positive half cycle
- Discharging time constant  $R_L C$  must be long enough to ensure that the capacitor discharges slowly through  $R_l$  between the positive peak of the carrier wave.

$$\frac{1}{f_c} \ll R_L C \ll \frac{1}{W}$$

[ $W$  is bandwidth of the message signal]

## Synchronous Detection

**if  $\mu > 1$ , we should use this.**



### Local Oscillator

- generates carrier wave
- It is phase synchronous by Transmitter and Receiver

### Advantages:

- Better quality than modulation
- less affected by noise

### Draw Backs

- Transmitter and receiver are required
- More Complexity

## Angle Modulation

Angle modulation is a modulation process in which **the angle of the carrier wave or signal is changed** with respect to the message signal or baseband signal.

Carrier Signal for Angle modulation,

$$s(t) = A_c \cos(\omega_c t + \phi)$$

Amplitude,  $A_c$

$$\text{Angle, } \theta = \omega_c t + \phi$$

Frequency,  $f_c$  (in  $\omega_c$ )

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

Phase =  $\psi$

$$\theta = \omega_c t + \phi$$

$$f_{ins} = f_c + \Delta f$$

$$\Rightarrow \frac{d\theta}{dt} = \omega_c + \frac{d\phi}{dt}$$

$$(f_{ins})_{max} = f_c + (\Delta f)_{max} \dots (i)$$

$$\Rightarrow \omega_{ins} = \omega_c + \frac{d\phi}{dt}$$

$$(f_{ins})_{min} = f_c - (\Delta f)_{max} \dots (ii)$$

$$\Rightarrow f_{ins} - f_c = \frac{1}{2\pi} \frac{d\phi}{dt}$$

$$f_c = \frac{(f_{ins})_{max} + (f_{ins})_{min}}{2}$$

$$\Rightarrow \Delta f = f_{ins} - f_c = \frac{1}{2\pi} \frac{d\phi}{dt}$$

$$\text{Carrier swing, } (\Delta f)_{max} = \frac{(f_{ins})_{max} - (f_{ins})_{min}}{2}$$

$\omega_{ins}$  = instantaneous angular frequency

$f_{ins}$  = instantaneous frequency

$\Delta f$  = Frequency derivation

## Frequency Modulation

**Frequency Modulation (FM)** is that form of angle modulation in which the *instantaneous frequency  $f_{ins}$  is varied linearly with the baseband signal  $m(t)$ .*

$$\Delta f \propto m(t) \Rightarrow \Delta f = k_f m(t), \text{ Unit of } k_f = \frac{\text{Hz}}{\text{Volt}}$$

where  $k_f$  = frequency sensitivity of modulation

### Expression

$s(t) = A_c \cos(\omega_c t + \phi)$ , we have to convert  $\phi$  to frequency.

$$\Delta f = \frac{1}{2\pi} \frac{d\phi}{dt} \Rightarrow \int k_f m(t) dt = \frac{1}{2\pi} \phi$$

$$\Rightarrow \phi = 2\pi \int k_f m(t) dt$$

$$s(t) = A_c \cos[\omega_c t + 2\pi k_f \int m(t) dt] [k_f = \text{Hz/Volt}]$$

$$s(t) = A_c \cos[\omega_c t + k_f \int m(t) dt] [k_f = \text{rad/Volt}]$$

### For single tone

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

$$s(t) = A_c \cos[\omega_c t + 2\pi k_f \int A_m \cos 2\pi f_m t]$$

$$= A_c \cos[\omega_c t + \frac{k_f A_m}{f_m} \sin(\omega_m t)]$$

$$= A_c \cos[\omega_c t + \beta \sin(\omega_m t)]$$

$$\text{Modulation index, } \beta = \frac{k_f A_m}{f_m} = \frac{(\Delta f)_{\text{max}}}{f_m}$$

### Types of FM:

#### 1. Narrowband FM ( $\beta \leq 1$ )

This frequency modulation has a **small bandwidth** when compared to wideband FM. The modulation index  $\beta$  is **small**, i.e., less than 1. Its spectrum consists of the carrier, the upper sideband and the lower sideband. This is used in **mobile communications such as police wireless, ambulances, taxicabs, etc.**

$$s(t) = A_c \cos[\omega_c t + \beta \sin(\omega_m t)]$$

$$\Rightarrow s(t) = A_c \cos(\omega_c t) \cos[\beta \sin(\omega_m t)] - A_c \sin(\omega_c t) \sin[\beta \sin(\omega_m t)]$$

$$\text{where, } \cos[\beta \sin(\omega_m t)] \approx 1$$

$$\sin[\beta \sin(\omega_m t)] \approx \beta \sin(\omega_m t)$$

$$\text{Hence, } s(t) = A_c \cos(\omega_c t) - \beta A_c \sin(\omega_c t) \sin(\omega_m t)$$

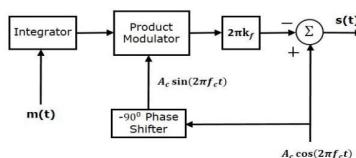
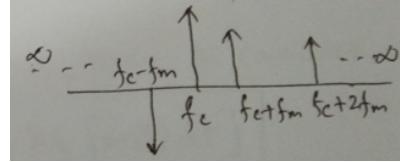


Figure 2.2.2 Generation of Frequency Modulated Signal

#### 2. Wideband FM ( $\beta > 1$ )

Ideal Bandwidth =  $\infty$



To get rid off this infinite bandwidth,

#### Carson's Law :

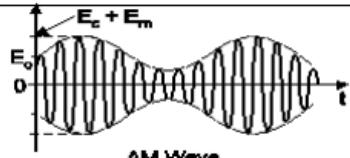
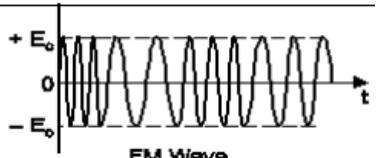
- estimates the transmission bandwidth necessary for efficiently sending an FM signal without excessive bandwidth consumption
- Carson's rule states that the **bandwidth required to transmit an angle modulated wave is twice the sum of the peak frequency deviation and highest modulating signal frequency.**

$$B = 2(\Delta f_{max} + f_m) = 2f_m(\beta + 1)$$

- 90% power transmit

#### 7. Compare NBFM and WBFM in detail.

Sr. No.	Parameter	NBFM	WBFM
1.	Modulation index	Less than or slightly greater than 1	Greater than 1
2.	Maximum deviation	5 kHz	75 kHz
3.	Range of modulating frequency	20 Hz to 3 kHz	20 Hz to 15 kHz
4.	Maximum modulation index	Slightly greater than 1	5 to 2500
5.	Bandwidth	Small approximately same as that of AM $BW = 2f_m$	Large about 15 times greater than that of NBFM, $BW = 2(\delta + f_{mmax})$
6.	Applications	FM mobile communication like police wireless, ambulance, short range ship to shore communication etc.	Entertainment broadcasting (can be used for high quality music transmission)

Parameter	AM	FM
1. Definition	Amplitude of carrier is varied in accordance with amplitude of modulating signal keeping frequency and phase constant.	Frequency of carrier is varied in accordance with the amplitude of modulating signal keeping amplitude and phase constant.
2. Constant parameters	Frequency and phase.	Amplitude and phase.
3. Modulated signal	 <p>AM Wave</p>	 <p>FM Wave</p>
4. Modulation Index	$m =  E_m/E_c $	$m = \delta/f_m$
5. Number of sidebands	Only two	Infinite and depends on $m_f$ .
6. Bandwidth	$BW = 2f_m$	$BW = 2(\delta + f_{m(max)})$
7. Application	MW, SW band broadcasting, video transmission in TV.	Broadcasting FM, audio transmission in TV.

## Phase Modulation

Phase modulation (PM) is a type of angle modulation where the **phase of the carrier signal is varied** in proportion to the the modulating signa  $m(t)$ .

$$\phi \propto m(t) \Rightarrow \phi = k_p m(t)$$

where,  $k_p$  = phase sensitivity of modulation =  $\frac{\text{rad}}{\text{volt}}$

### Expression

$$s(t) = A_c \cos(\omega_c t + \phi) = A_c \cos(\omega_c t + k_p m(t))$$

For single tone,

$$s(t) = A_c \cos[\omega_c t + k_p A_m \cos(\omega_m t)] = A_c \cos[\omega_c t + \beta \cos(\omega_m t)]$$

$$\text{Modulation index, } \beta = k_p A_m = \frac{\Delta f_{max}}{\delta f_m \text{ in}}$$

$$\Delta f = \frac{k_p}{2\pi} \frac{d}{dt} m(t)$$

$$\Delta f_{max} = \frac{k_p}{2\pi} [\frac{d}{dt} m(t)]_{max} = \frac{k_p}{2\pi} A_m \omega_m [(-\sin \omega_m t)]_{max}$$

$$\Delta f_{max} = k_p A_m f_m, [(-\sin \omega_m t)_{max} = 1]$$

$$\Delta f_{min} = -k_p A_m f_m [(-\sin \omega_m t)_{max} = -1]$$

**Phase derivation:** Difference between phase of modulated carrier and phase of unmodulated carrier.

$$\text{Unmodulated carrier} = A_c \cos(\omega_c t + 0^\circ)$$

$$\text{Modulated carrier} = A_c \cos(\omega_c t + k_p m(t))$$

$$\text{Phase derivation, } \Delta\phi = k_p m(t)$$

$$\Delta\phi_{max} = \max(k_p m(t))$$

Single tone,

$$\Delta\phi_{max} = k_p A_m \quad [cos\omega_m t = 1]$$

$$\Delta\phi_{min} = -k_p A_m$$

## Receivers

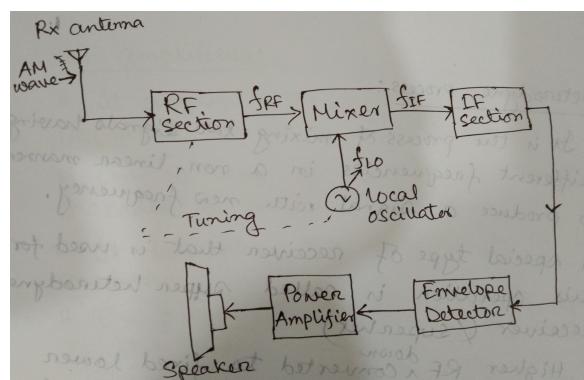
In radio communications, a radio receiver or receiver is an **electronic device** that **receive radio waves** and converts the information carried by them to a **usable** form.

1. TRF : Tuned Radio Frequency
2. SHR : Super Heterodyne Radio Receiver

### Heterodyne Process

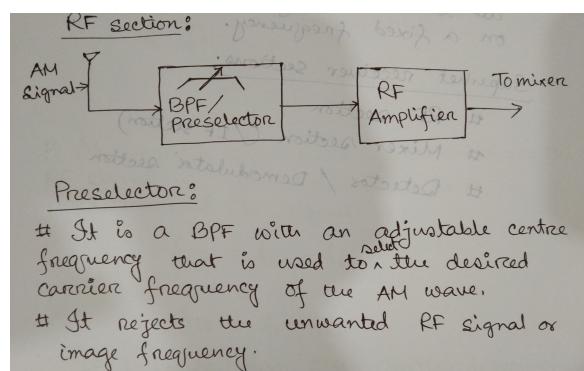
It is a process of **mixing two signals** having different frequencies in a **non-linear manner** to **produce a signal with new frequency**.

A special type of receiver that is used for this operation is called Super heterodyne receiver (Superhet).



### Superhet receiver sections

- RF Section
- Mixer Section/IF section
- Detector/Demodulator Section



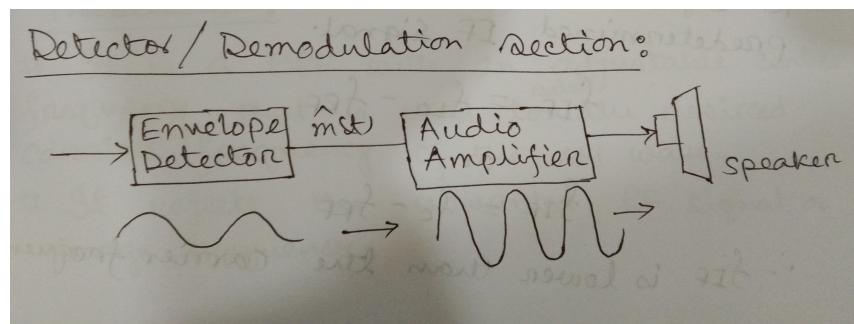
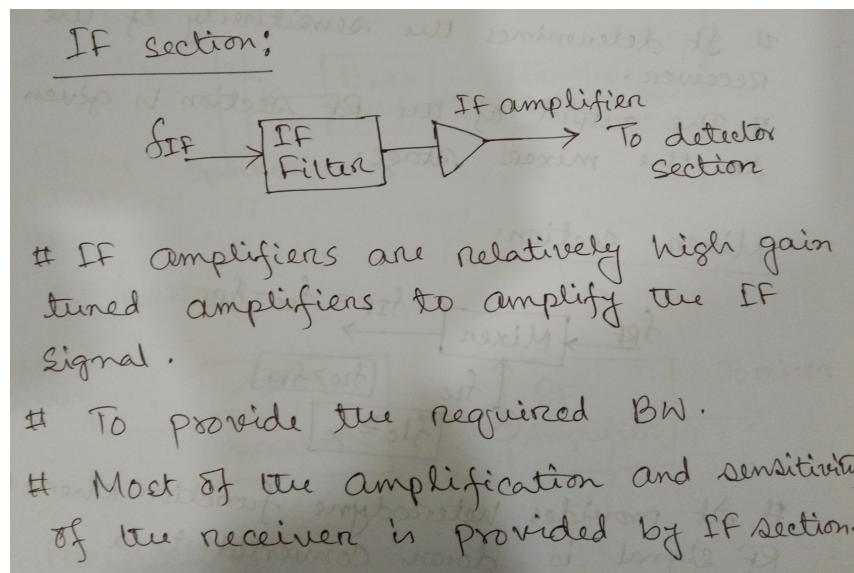
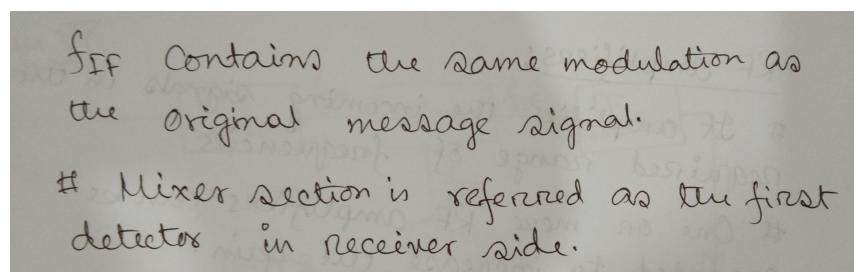
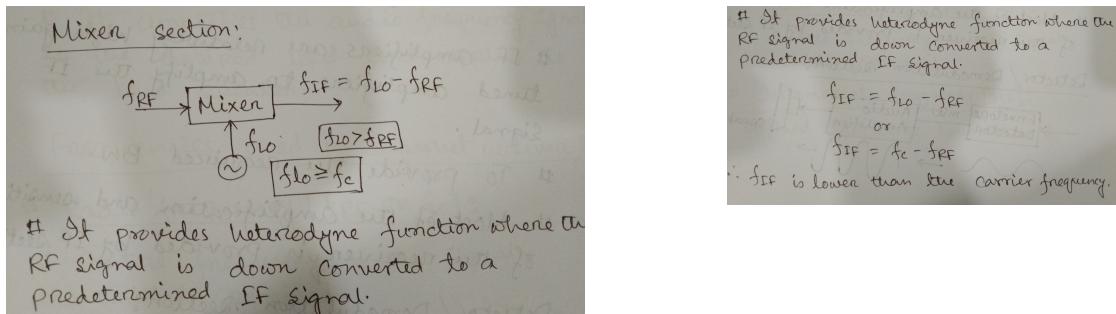
### Preselector

- used to select the desire carrier frequency of AM wave
- It rejects unwanted RF signal or image frequency

### RF Amplifiers

- It amplifies the incoming signals in the required range of frequencies.
- One or more RF amplifiers can be cascaded to increase the gain.
- It determines the sensitivity of the receiver

- The output of the RF section is given to the mixer stage



**Envelope Detector**

**Audio Amplifier**

- Detector **recovers the original information** of the baseband signal from the output of IF amplifier
- The output of detector is **low power audio signal**
- It amplifies the audio frequency signal to the desired level and applies to the loud speaker

### Frequency used in Superhet receiver

1. Radio Frequency ( $f_{RF}$  or  $f_s$ ): Centre frequency of the signal transmitted/received.
2. Intermediate Frequency ( $f_{IF}$ ): Fixed frequency lower than  $f_c$
3. Local Oscillator Frequency ( $f_{LO}$  or  $f_o$ )

$$f_{LO} \geq f_c$$

$$f_{LO} = f_{RF} + f_{IF} = f_s + f_{IF}$$

### Image Frequency ( $f_{si}$ ):

Image signal is the unwanted signal present at  $f_{si}$

$$f_{si} = f_{LO} + f_{IF} = f_{RF} + 2f_{IF}$$

The image frequency must be rejected.

### Image frequency rejection ration (IFRR):

$$\text{IFRR}, \alpha = \frac{\text{Gain at the signal frequency}}{\text{Gain ar the image frequency}} = \sqrt{1 + Q^2 P^2}$$

$$P = \frac{f_{si}}{f_s} - \frac{f_s}{f_{si}}$$

$Q$  = Quality factor of tuned circuit

The value of IFRR should be high.

### Advantages

- High sensitivity and selectivity
- High adjacent channel rejection
- Stability is improved
- High gain
- Uniform BW is used due to fixed IF

### Disadvantages

- It requires additional LO and RF mixer. This increases cost of overall receiver.
- Filters are also needed to remove any LO Leakage and undesired frequency. This also increases cost and complexity.

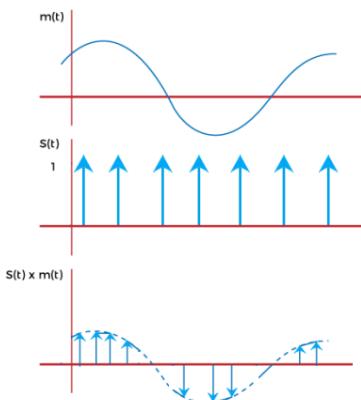
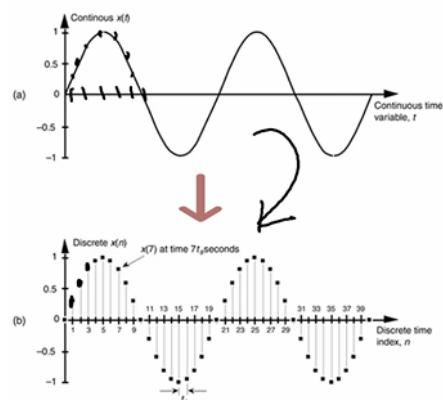
### Applications

All radio and TV receivers operate on the principle of super heterodyne methos.

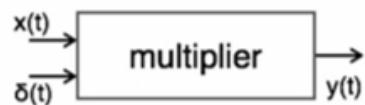
### Sampling

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- Sampling is the reduction of a continuous-time signal to a discrete time signal.
- Sampling is a process performed by a sampler
- Analog signal is samples every  $T_s$  secs, called sampling interval.
- Sampling rate/ frequency =  $f_s$



When the continuous signal  $m(t)$  is sampled at regular intervals multiplied by a periodic pulse train  $s(t)$  / impulse train  $\delta(t)$ , the produced signal is the sampled signal.

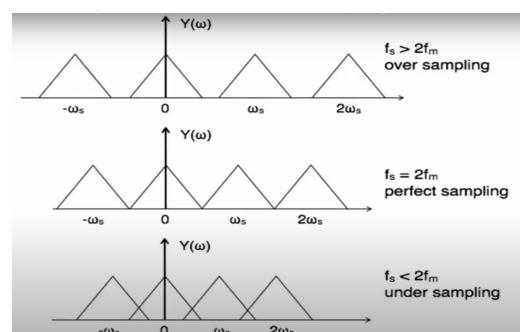


**Sampling Theorem:** A continuous time signal can be represented in its samples and can be recovered back when sampling frequency  $f_s$  is greater than or equal to the twice highest frequency component of message signal.

[ $f_m$  is the maximum frequency of the input or message signal]

Condition:  $f_s \geq 2f_m$

**Sampling rate** is defined as the number of samples taken per second from a continuous signal for a finite set of values.  $f_s = \frac{1}{T_s}$



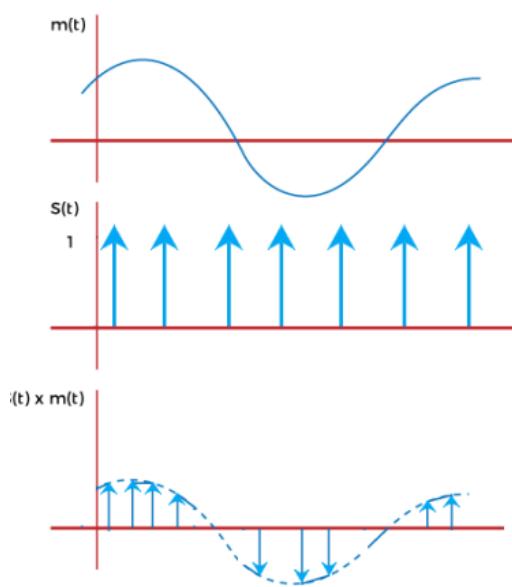
**Nyquist rate:** The theoretical minimum sampling rate at which a signal can be sampled and still can be reconstruct samples without any distortion.

$$f_s = 2f_m$$

**Nyquist Interval:** Nyquist interval is the reciprocal of the Nyquist rate.  $T_s = \frac{1}{2f_m}$

## Methods of Sampling

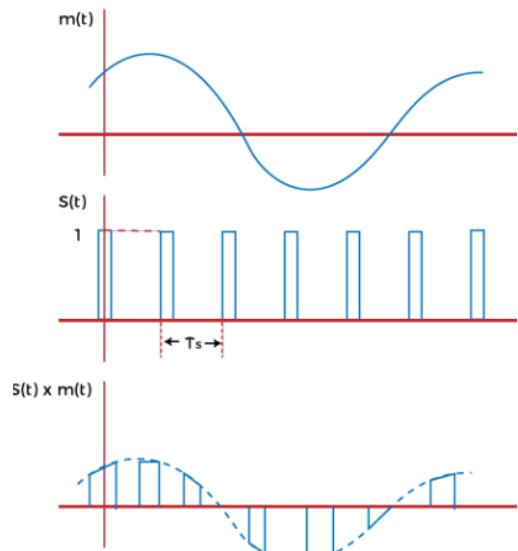
## 1. Ideal Sampling



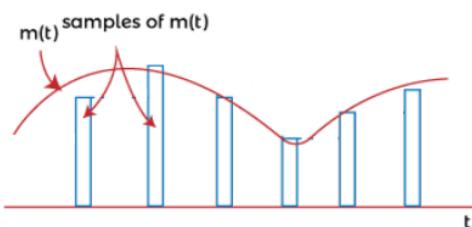
- Known as instantaneous sampling/impulse sampling
- multiplied by a impulse train signal
- Sampling rate in infinite
- Bandwidth is large

## 2. Natural Sampling

- considered as efficient multiplexing method in Pulse Amplitude Modulation
- multiplied by the uniformly spaced rectangular pulses
- pulses do not have flat tops, they are curved
- their tops follow the waveform of input or message signal



## 3. Flat-top Sampling



- easier than natural sampling
  - amplitude of sampled pulses are constant
  - pulses have flat top
  - Aperture effect: The loss of high frequency content because of constant amplitude.
- Example:



### Advantages/Why

The advantages of the sampling process are due to conversation of the transmission of digital form, which have various advantages.

**Applications:** PAM, PCM, TDM

- Low cost
- High Accuracy
- Easy to implement
- Less time consuming
- Low signal loss
- High Scope

### Quantization

Quantization is the process in which continuous amplitude (analog) signals is converted into discrete amplitude (digital) signals.

Given a real number  $x_m$  we denote the quantized value of  $x$  as,

$\hat{x} = Q(x) = x + \epsilon$ , where  $\epsilon$  is the quantization error.

There are two main types of quantization

#### 1. Truncation

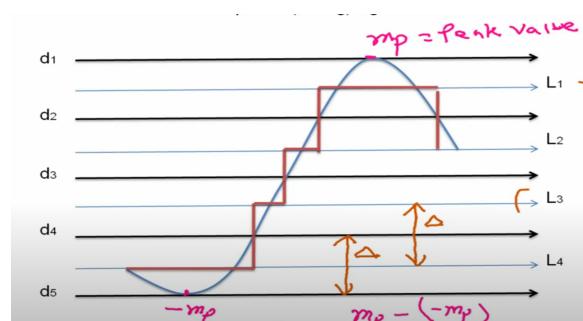
Just discard the least significant bits

$$Q(1/\sqrt{2}) = [0_{\Delta} 10]_{110 \dots} = 0.50$$

#### 2. Rounding

just choose the closet value

$$Q(1/\sqrt{2}) = [0_{\Delta} 11]_{110 \dots} = 0.75$$



Peak to peak value =  $2m_p$

- Dividing by decision boundaries (Here 5)
- Quantization level ( $L$ ) : Decision Boundaries - 1
- Step size,  $\Delta = \frac{V_H - V_L}{L} = \frac{2m_p}{L}$
- Number of bits to represent each sample (Level),  $n = \log_2^L$

**Quantization Error:** The difference between an input and output of a quantizer is known as a quantization error.

Quantization Error,  $q = x(t) - y(t)$

Range of quantization error :  $-\frac{\Delta}{2} \leq q \leq \frac{\Delta}{2}$

### Signal-to-noise Ratio

$$P = \frac{V^2}{R} = \frac{A^2}{R}$$

$$SNR = \frac{\text{Wanted Component}}{\text{Unwanted Component}} = \frac{P_{signal}}{P_{noise}} = \left(\frac{A_{signal}}{A_{noise}}\right)^2,$$

Unit : Watt or dB, 1 dB =  $10 \log_{10}^P$

$$SNR_{dB} = 10 \log_{10}^{SNR} = 10 \log_{10}\left(\frac{P_{signal}}{P_{noise}}\right) = 20 \log_{10}\left(\frac{V_{signal}}{v_{noise}}\right)$$

$$\text{Signal Power, } P_{signal} = \frac{m_p^2}{2}$$

Quantization noise power/ Average Quantization power / Mean square value of quantization error ,

$$Nq = \frac{1}{\Delta} \int_{-\frac{\Delta}{2}}^{\frac{\Delta}{2}} q^2 dq = \frac{\Delta^2}{12}$$

$$\text{Noise power, } P_{noise} \text{ or } Nq = \frac{\Delta^2}{12} = \frac{4m_p^2}{12L^2} \text{ [Derivation below]}$$

$$SNR = \frac{3}{2}L^2$$

$$SNR_{dB} = 10 \log_{10} \frac{3}{2}L^2 = 1.76 + 10 \log_{10} L^2 = 1.76 + 6n$$

[ $n$  = Number of bits to represent each sample or level ]

**Bit Rate:** The total bits transmitted in one unit time. It means the total bits that travel per second.

$$R_b = \frac{\text{bits}}{\text{samples}} \times \frac{\text{samples}}{\text{sec}} = n \times f_s, \text{ Unit : bits/sec}^{-1}, \text{ bits per second}$$

**Shannon Capacity or Channel Capacity:** Shannon's Channel Capacity theorem states that the maximum rate at which information can be transmitted over a communication channel of a specified bandwidth in the presence of noise.

$$C = B \log_2(1 + SNR)$$

$C$  = Channel Capacity, Unit : bits/sec<sup>-1</sup>

$B$  = Bandwidth

It tells us the theoretical highest limit in which any channel can transmit signals with the presence of noise. [\[Ref\]](#)

**Baud Rate:** The total number of signal units transmitted in one second.

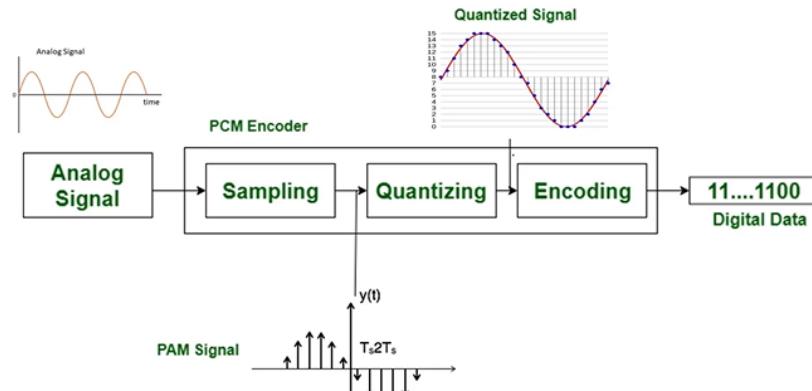
$$\text{Baud Rate} = \frac{R_b}{n}$$

Parameters	Baud Rate	Bit Rate
Basics	The Baud rate refers to the total number of signal units transmitted in one second.	The Bit rate refers to the total Bits transmitted in one unit time.
Meaning	Baud rate indicates the total number of times the overall state of a given signal changes/ alters.	Bit rate indicates the total bits that travel per second.
Determination of Bandwidth	The Baud rate can easily determine the overall bandwidth that one might require to send a signal.	The bit rate cannot determine the overall signal bandwidth.
Generally Used	It mainly concerns the transmission of data over a given channel.	It mainly focuses on the efficiency of a computer.
Equation	Baud Rate = Rb/n	Bit Rate = fs * n

[\[Ref\]](#)

## Pulse Code Modulation (PCM)

Information is transmitted in the form of "code words". PCM output is in coded digital form.



**Block Diagram Of PCM**

#### 1. LPF

- Bandlimits
- Eliminates possibility of aliasing

#### 4. Encoder

- Analog to digital converted
- Each quantized level convert into N bit digital word

#### 2. Sample & Hold Circuit

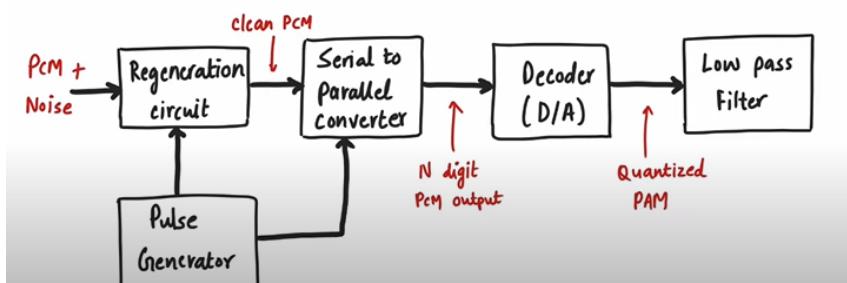
- Convert to flat top PAM

#### 3. Quantizer

- Quantized PAM

#### 5. Parallel to Serial Converter

### Decoder



### Advantages

- Very high noise immunity
- Due to digital nature of the signal, repeaters can be placed between transmitter and receiver. It reduce the effect of noise and regenerate the received PCM signal.
- It is possible to store PCM.
- It is possible to use various coding techniques so that the desired person can decode the

### Disadvantages

- The encoding, decoding and quantizing **circuitry of PCM is complex**.
- Requires a **large bandwidth**

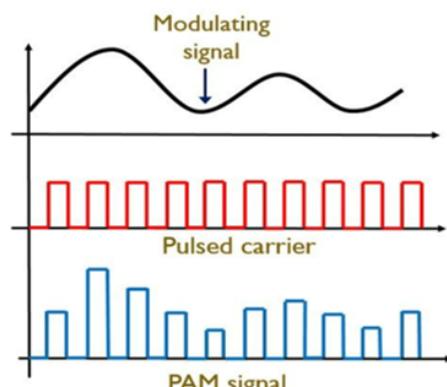
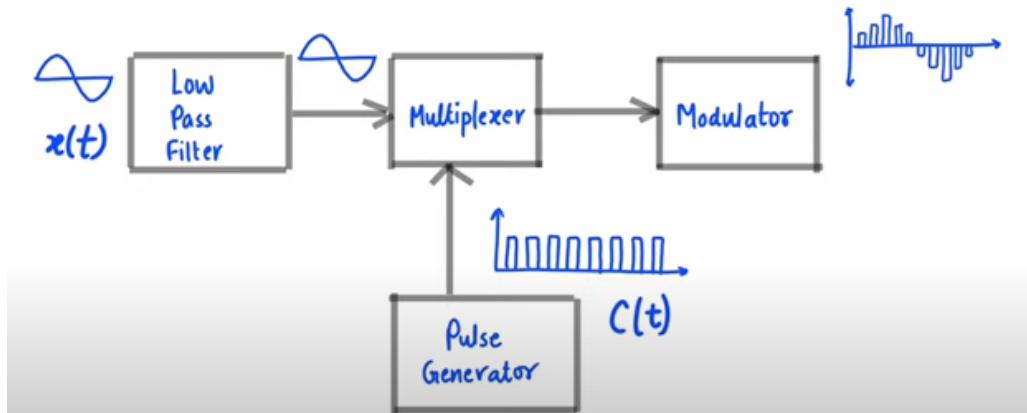
### Applications:

- Telephony
- Space Communication (because of high noise immunity)

received signal.

- Integration with other form of digital data is possible

## Pulse Amplitude Modulation



- amplitude of the pulse carrier signal is changed according to the amplitude of the message signal.
- Sampling Method: Flat Top PAM, Natural PAM

**Applications:** Ethernet, Photo biology, Driver for LED lighting, Micro controller

## Demodulation



### Reconstruction Filter:

- Cut off frequency  $f_c > f_m$

### Equalizer:

- Reduce aperture effect and attenuation

## Advantages

- Simple modulation and demodulation
- No complex circuitry for transmission and reception

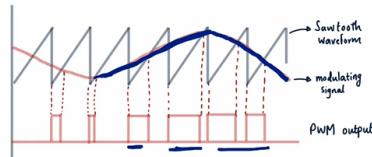
## Disadvantages

- Bandwidth will be large. Due to Nyquist criteria also high bandwidth is required.

- PAM can generate other pulse modulation
- Can carry message or information at the same time

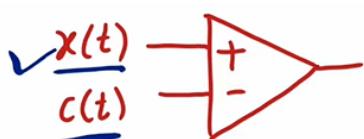
- Amplitude varies according to modulating signal  $\rightarrow$  interference  $\rightarrow$  more noise
- Pulse amplitude signal varies  $\rightarrow$  more power required

## Pulse Width Modulation (PWM)



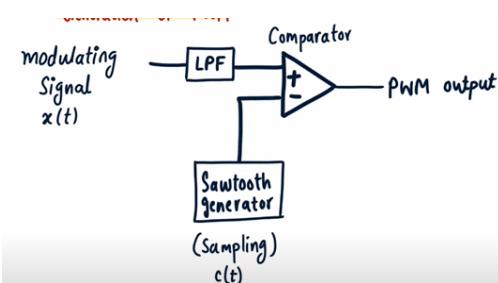
- Amplitude of  $x(t) \uparrow \rightarrow$  Width of PWM Signal  $\uparrow$
- Amplitude of  $x(t) \downarrow \rightarrow$  Width of PWM Signal  $\downarrow$
- Leading edges of PWM waveform coincide with the falling edges of ramp signal

### Comparator



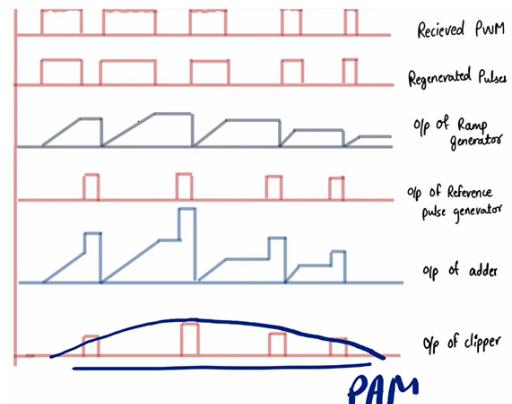
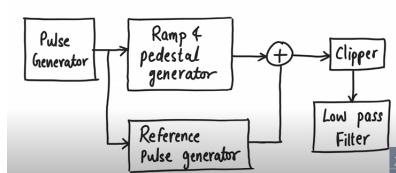
- Output High,  $x(t) > c(t)$
- Output Low,  $x(t) < c(t)$

### Generation of PWM



- non-inverting  $x(t)$
- inverting  $c(t)$

### Demodulation



## Advantages

- Amplitude constant → Noise interference less
- Signal and noise can be separated very easily at demodulation
- Synchronization between transmitter and receiver is not required

## Disadvantages

- Power will be variable → width of the pulse varies
- Bandwidth is large compared to PAM

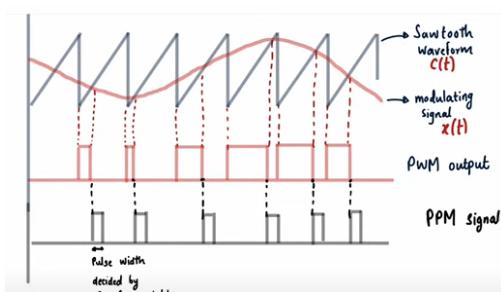
**Applications:** Telecommunication, robotics, audio effects and amplifications, control the amount of power, control the speed of the robot

## Pulse Position Modulation (PPM)

Position of pulse varies according to modulating signal.

PPM can be derived from PWM.

### Generation

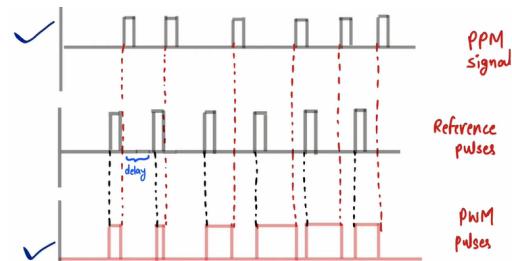
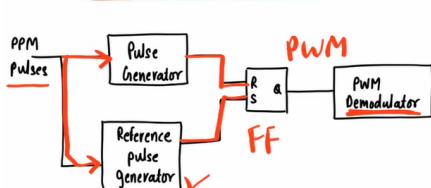


### Monostable:

- One stable state (Low)
- Give Trigger pulse → output = High (for some time)
- "Some time" depends on Register and Capacitor (RC) component

- Width and Amplitude Constant
- Position varies

### Demodulation



## Advantages

- Low noise interference compared to PAM
- Noise removal and separation is very easy
- Power usage is very low

## Disadvantages

- Synchronization between transmitter and receiver is required.
- Large bandwidth

**Applications:** Non coherent detection, RF communication, contactless smart card, RFID

Parameters	ASK	FSK	PSK
Variable Characteristics	Amplitude	Frequency	Phase
<b>Bandwidth</b>	Minimum bandwidth requirement for bpsk is $N_b$ ; $N_b$ is the baud rate;	Here bandwidth requirement is $(f_{c2} - f_{c1}) + N_b$ ; $f_{c2}$ & $f_{c1}$ are the carrier frequencies for bpsk	the maximum bandwidth efficiency of BPSK is $1/b/s/HZ$
<b>Noise Immunity</b>	As noise is very sensitive to amplitude, here noise immunity is low	Here noise immunity is high.	Here also noise immunity is high.

<b>Complexity</b>	Simple	Moderately complex	Very complex
<b>Error probability</b>	High	low	Low
<b>Performance in presence of noise</b>	Poor	Better than ask	Better than fsk
<b>Bit rate</b>	Suitable up to 100 bits/sec	Suitable up to 1200 bits/sec	Suitable for higher bit rate