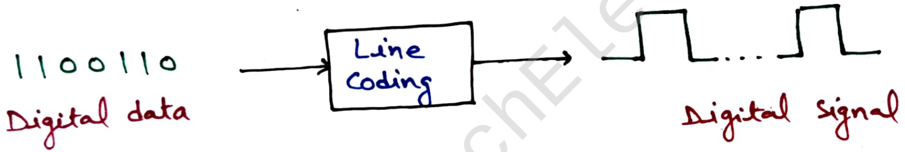
Line Coding:-

→ The binary data such as 1's & 0's produced by a PCM encoder may be represented in various signalling formats for the transmission over a channel.

→ These signalling formats are known as line codes.

→ In otherwords we can say, line coding is the process that converts digital data to digital signal



→ Line coding techniques can be broadly divided into three categories.

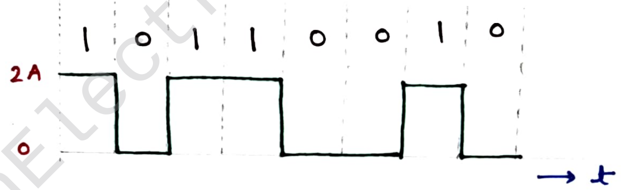
→ Unipolar

→ Polar

→ Bipolar

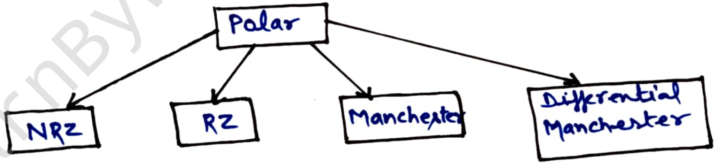
Unipolar:-

- In unipolar encoding technique only two voltage levels are used.
- It uses only one polarity of voltage level as shown in figure
- Unfortunately, DC component present in the encoded signal and there is loss of synchronization for long sequences of 0's and 1's.
- It is simple but obsolete.



Polar:-

- Polar encoding technique uses two voltage levels. one +ve and other one -ve.
- Four different encoding schemes under Polar encoding

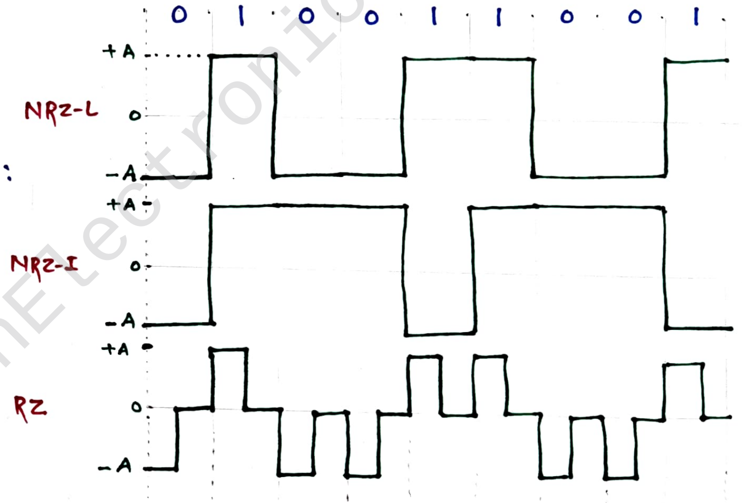


Non Return to zero (NRZ) :-

→ Two different voltage levels are used for two binary digits (i.e 1 and 0)

→ In NRZ encoding the signal level remains Same throughout the bit period.

→ There are two encoding schemes in NRZ : NRZ-L and NRZ-I



NRZ-I

→ For each 1 in the bit sequence, the signal level is inverted

→ A transition from one voltage level to the other represent a 1.

Return to zero (RZ) :-

→ To ensure synchronization there must be a signal transition in each bit.

→ No DC component

→ Three Levels

→ Main limitation is the increase in bandwidth.

Biphase :

→ To overcome the limitations of NRZ encoding biphase encoding techniques (Manchester and differential manchester) are used.

→ In the standard Manchester coding there is a transition at the middle of each bit period.

→ A binary 1 corresponds to low to high transition 

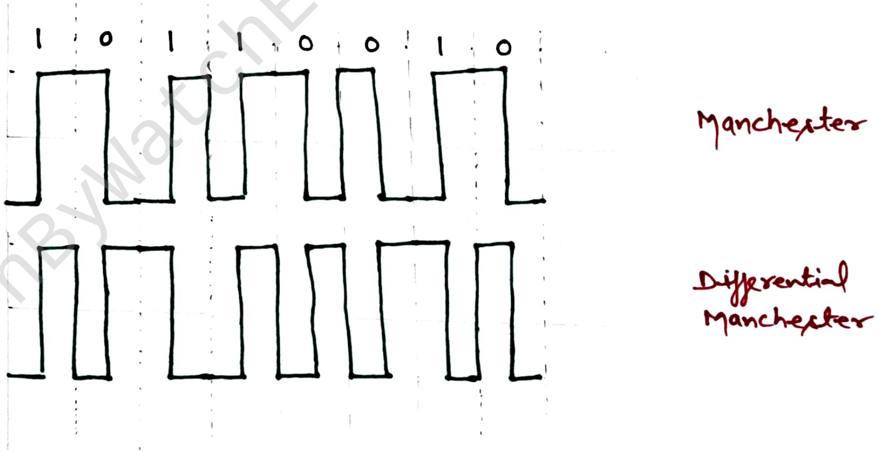
→ A binary 0 corresponds to High to low transition 

Key characteristics

→ Two Levels

→ No DC component

→ Good Synchronization



→ In Differential manchester

0 : Represented by the presence of a transition both at beginning and at the middle of bit period.

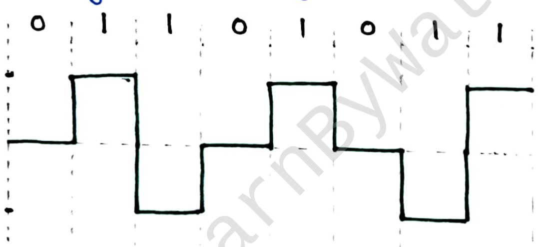
1 : Represented by a transition only in the middle of the bit period.

Bipolar Encoding :-

→ Bipolar AMI uses three voltage levels.

→ Binary 0 is represented by zero level.

→ Binary 1 is represented by alternating positive and negative voltages as shown in figure.



Bipolar AMI

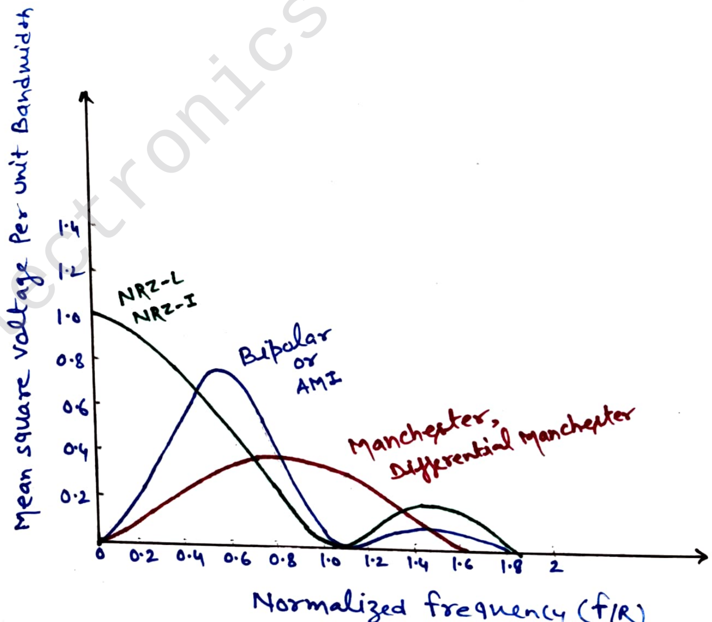


Fig. Frequency spectrum of different encoding schemes

Adaptive delta Modulation :-

→ It is a modified version of delta modulation.

→ To overcome slope overload distortion and Granular noise in delta modulation, Adaptive delta modulation is used.

→ Here the step size Δ of the quantizer is not a constant but varies with time.

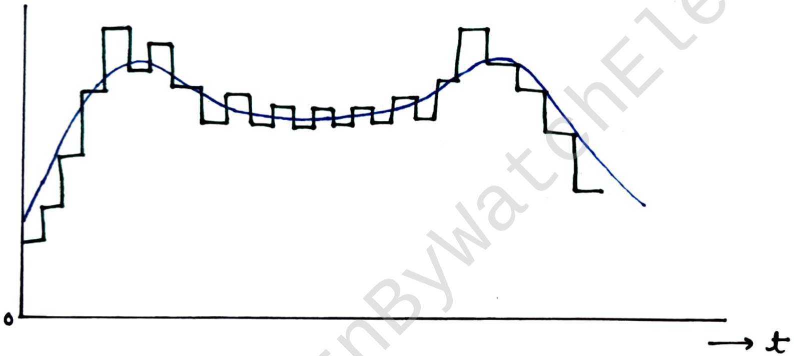


Fig 1. Waveforms illustrative of ADM operation

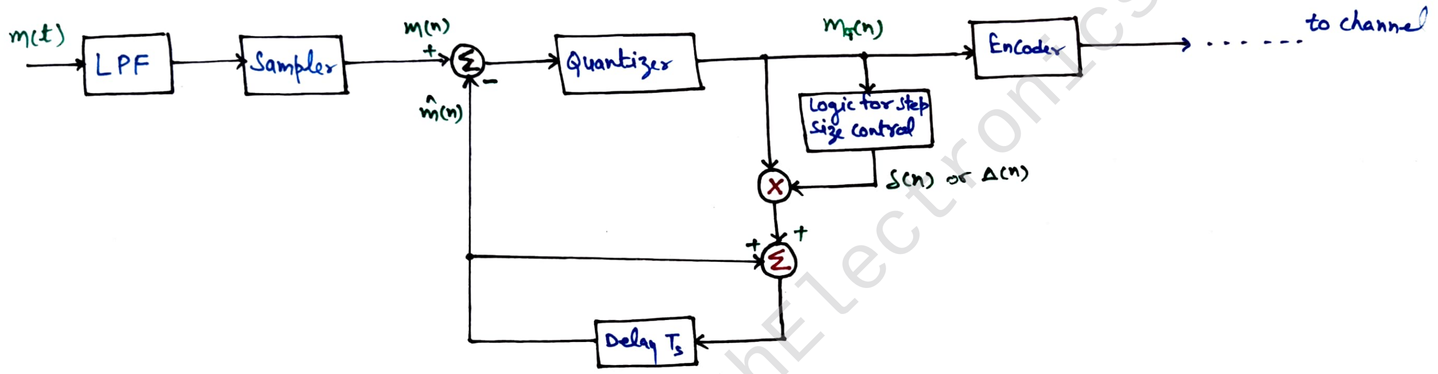


Fig 2. ADM Transmitter

→ Fig 2. shows the block diagram of ADM Transmitter. In practical implementation, the step size $\delta(n)$ is constrained to be between some predetermined minimum and maximum values.

$$\delta_{min} \leq \delta(n) \leq \delta_{max}$$

→ The upper limit δ_{max} controls the amount of slope overload distortion and the lower limit δ_{min} controls the granular noise.

→ The adaptive Rule for step Size $\delta(n)$ can be expressed in the general form

$$\delta(n) = g(n)\,\delta(n-1)$$

Where: The time varying gain $g(n)$ depends on the present output $m_q(n)$ and M Previous Values $m_q(n-1), m_q(n-2) \cdots m_q(n-M)$

→ For simplicity assume $M=1$

→ Successive values of $m_q(n)$ and $m_q(n-1)$ are compared to detect probable slope overload $[m_q(n) = m_q(n-1)]$ or probable granularity $[m_q(n) \neq m_q(n-1)]$

Then $g(n)$ can be written as

$$g(n) = \begin{cases} P, & \text{if } m_q(n) = m_q(n-1) \\ \frac{1}{P}, & \text{if } m_q(n) \neq m_q(n-1) \end{cases}$$

Where: $P > 1$

Typically $P_{opt} = 1.5$ or $\frac{3}{2}$ for speech signal

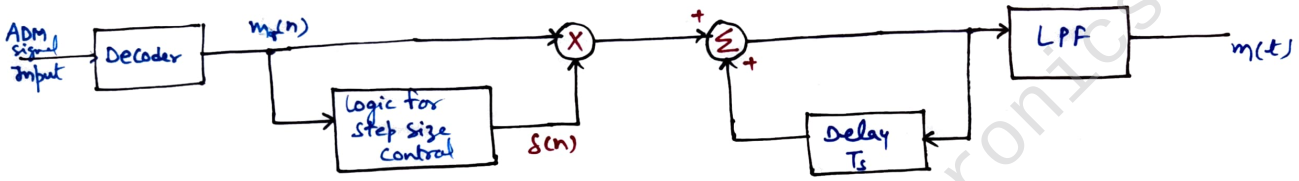


Fig. 3 ADM Receiver

Delta Modulation:-

→ Delta modulation, like DPCM is a predictive waveform coding technique and can be considered as a special case of DPCM.

→ It uses the simplest possible quantizer that is two level (or one bit) quantizer.

→ The simplicity of quantizer is achieved at the cost of increased sampling rate (much higher than the Nyquist rate)

→ In DM, the analog signal is highly over-sampled in order to increase the adjacent sample correlation.

→ The advantage of delta modulation compared to PCM and DPCM is minimum channel Bandwidth requirement.

For delta modulation 1-bit is transmitted for a sample

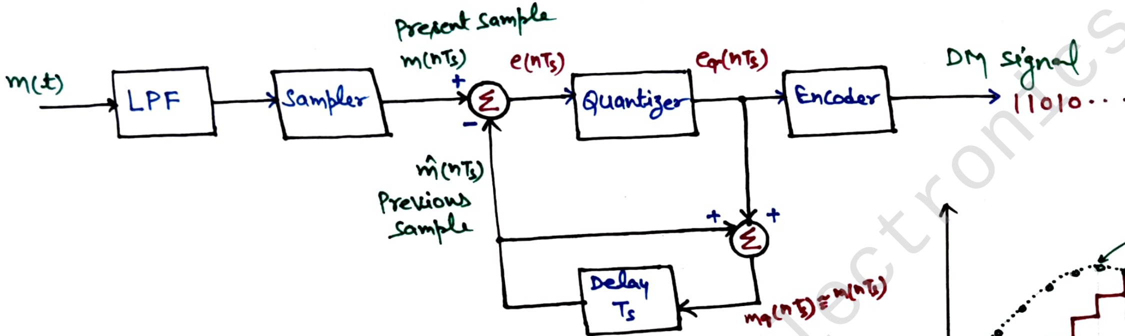
$$\downarrow B.W = \frac{\overset{\downarrow}{n} \cdot f_s}{2}$$

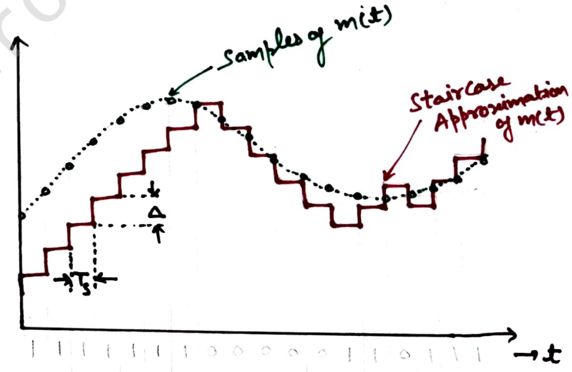
→ For delta modulator two levels are used

$$L = 2 \begin{cases} \rightarrow +\Delta \rightarrow 1 \\ \rightarrow -\Delta \rightarrow 0 \end{cases}$$

Δ: Step Size

DM Transmitter :-

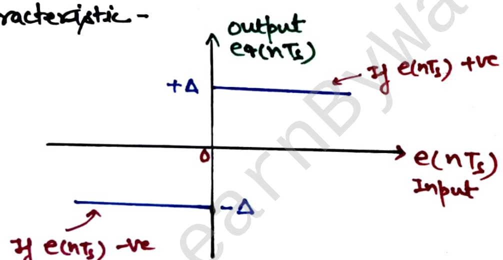




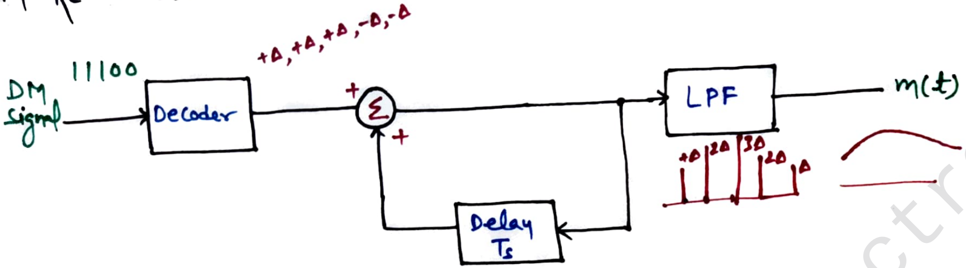
$$(\text{Summer})_{o/p} = e(nT_s) = m(nT_s) - \hat{m}(nT_s)$$

$$= \text{Present sample} - \text{Previous sample}$$

Quantizer characteristic -



DM Receiver:-



Errors in Delta Modulation :-
or
distortions

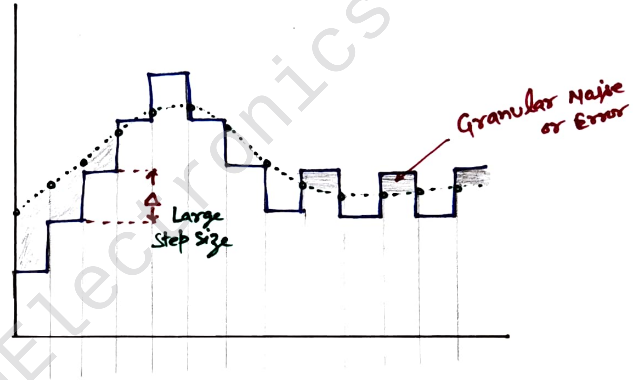
→ The value of Δ for which $m(t)$ can be perfectly reconstructed is called Δ_{opt} or optimum step size.

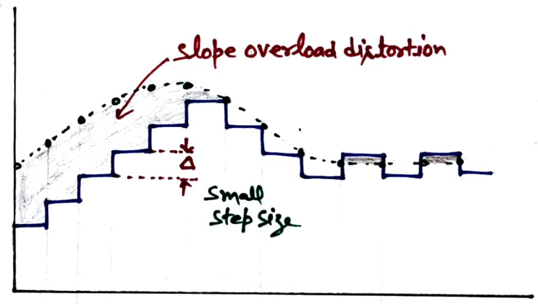
→ If $\Delta < \Delta_{opt}$, slope overload distortion or Error occures.

→ If $\Delta > \Delta_{opt}$, Granular Noise or Error occurs.

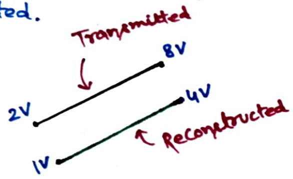
→ To overcome slope overload distortion Δ has to be increased.

→ To overcome Granular Noise, Δ has to be decreased.

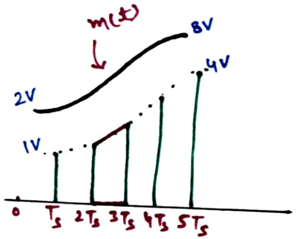




→ If the rate of change (i.e slope) of the reconstructed signal is same as the transmitted signal then transmitted message signal can be perfectly reconstructed.



(i) Perfect Reconstruction –



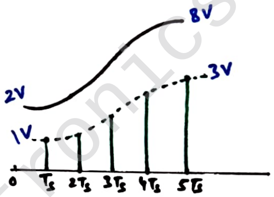
$$\text{Slope} = \frac{\Delta}{T_s}$$

Slope of $m(t)$ = slope of Reconstructed signal

$$\frac{d}{dt} m(t) = \frac{\Delta_{opt}}{T_s}$$

Rate of change of msg signal

(ii) Slope overload distortion

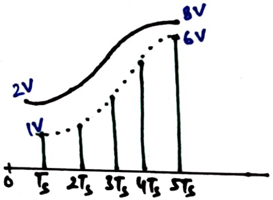


$$\frac{\Delta}{T_s} < \frac{d}{dt} m(t)$$

$$\frac{\Delta}{T_s} < \frac{\Delta_{opt}}{T_s}$$

$$\boxed{\Delta < \Delta_{opt}}$$

(iii) Granular Noise



$$\frac{\Delta}{T_s} > \frac{d}{dt} m(t)$$

$$\frac{\Delta}{T_s} > \frac{\Delta_{opt}}{T_s}$$

$$\boxed{\Delta > \Delta_{opt}}$$

DPCM (Differential Pulse Code Modulation) :-

In DPCM quantization error Qe is decreased without affecting channel Bandwidth Requirment.

$$\downarrow [Q_e]_{max} = \frac{\Delta}{2} \downarrow$$

$$\downarrow \Delta = \frac{V_{max} - V_{min}}{2^n} \downarrow$$

In DPCM Qe is decreased by decreasing the dynamic range of the signal to be quantized.

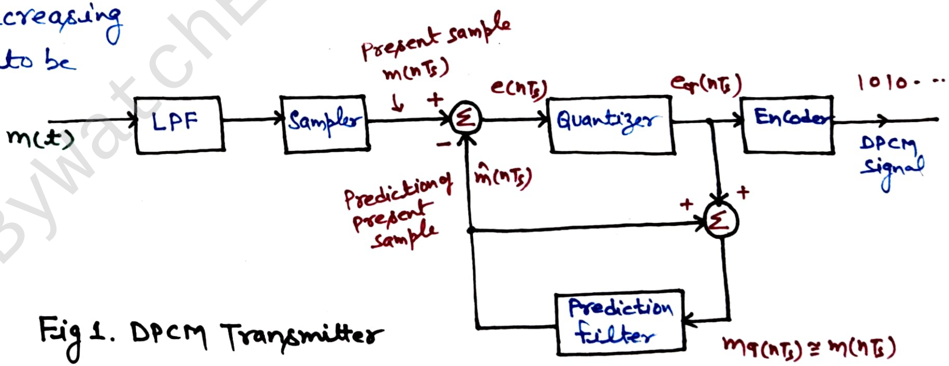


Fig 1. DPCM Transmitter

→ By analysing Past behaviour of the signal to sufficient time extent, prediction filter Predict nearby future values of the signal.

$$(\text{Summer})\,o/p = e(nT_s) = m(nT_s) - \hat{m}(nT_s)$$

→ usually $e(nT_s)$ will be very small.

→ Dynamic range of $e(nT_s)$ will be very small so that corresponding quantization error will also be very small.

Quantization error $q(nT_s) = e(nT_s) - e_q(nT_s)$ [$\because Q_e = S.V - q.V$]

$$\text{Prediction filter input} = \hat{m}(nT_s) + e_q(nT_s)$$
$$= \hat{m}(nT_s) + e(nT_s) - q(nT_s)$$
$$= m(nT_s) - q(nT_s)$$
$$= m_q(nT_s) \cong m(nT_s)$$

→ At a specific time instant previous sample voltages of the message signal will be available to the sufficient time extent. By analysing those previous samples prediction filter predict the next sample.

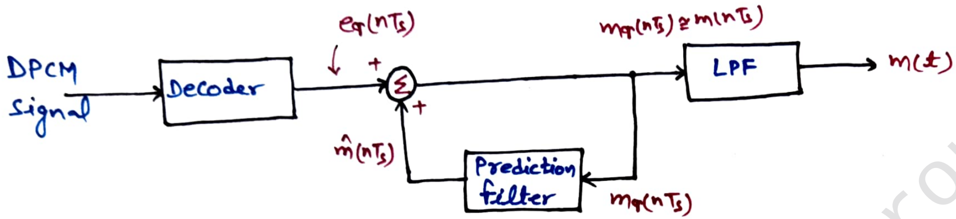


Fig2. DPCM Receiver

$$\text{Low Pass filter input (LPF) i/p} = e_q(nT_s) + \hat{m}(nT_s)$$
$$= \underline{e(nT_s)} - q(nT_s) + \underline{\hat{m}(nT_s)}$$
$$= m(nT_s) - q(nT_s)$$
$$= m_q(nT_s)$$

→ In the reconstructed signal finite amount of quantization error will be retained, which is very small compared to PCM transmission.

→ DPCM is complex than PCM.

Signal to Quantization Noise Ratio (SQNR):-

It is the ratio of signal power to Quantization noise power.

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

Signal Power $S = \frac{A_m^2}{2}$ ——— 1

Quantization noise power (N_q) can be calculated as follows -

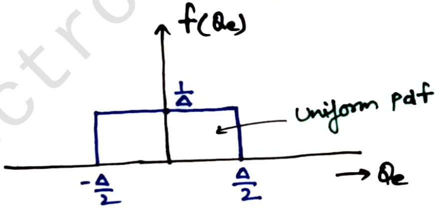
$$N_q = \text{Power of } [Q_e]$$

$$= \text{msq}[Q_e]$$

$$= E[Q_e^2]$$

$$N_q = \int Q_e^2 f(Q_e)\, dQ_e \quad \text{——— } 2$$

Assume Q_e is possessing uniform PDF.



eq^n 2 can be rewritten using uniform PDF

$$N_q = \int_{-\Delta/2}^{\Delta/2} Q_e^2 f(Q_e)\, dQ_e = \int_{-\Delta/2}^{\Delta/2} Q_e^2 \cdot \frac{1}{\Delta} \cdot dQ_e$$

$$= \frac{1}{\Delta}\left[\frac{Q_e^3}{3}\right]_{-\Delta/2}^{\Delta/2} = \frac{1}{3\Delta}\left[\frac{\Delta^3}{8} + \frac{\Delta^3}{8}\right]$$

$$N_q = \frac{\Delta^2}{12} \quad \text{——— } 3$$

for a sinusoidal signal

$$\Delta = \frac{2A_m}{2^n}$$

$$\because \Delta = \frac{V_{max} - V_{min}}{2^n}$$

$$= \frac{A_m - (-A_m)}{2^n}$$

Now eqn (3) can be written as

$$N_q = \frac{1}{12} \cdot \left[\frac{2A_m}{2^n}\right]^2$$

$$= \frac{1}{\cancel{12}_3} \cdot \frac{\cancel{4}A_m^2}{2^{2n}}$$

$$N_q = \frac{1}{3} \cdot \frac{A_m^2}{2^{2n}} \quad \text{—— (4)}$$

From eqn (1) and eqn (4)

$$\frac{S}{N_q} = \frac{\cancel{A_m^2}/2}{\cancel{A_m^2}/3 \cdot 2^{2n}}$$

$$\boxed{\frac{S}{N_q} = \frac{3}{2} \cdot 2^{2n}}$$

OR

$$\boxed{\frac{S}{N_q} = \frac{3}{2} L^2} \qquad \because L = 2^n$$

→ $\frac{S}{N_q}$ should be high

→ SQNR is generally expressed in dB

$$\boxed{\left(\frac{S}{N_q}\right)_{dB} \cong 1.8 + 6n}$$

where:
n: no. of bits per sample

Noise in PCM:-

There are mainly two types of noises that affect the performance of a PCM system.

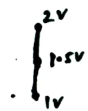
1. Channel Noise
2. Quantization Noise (Q_e) or Quantization Error

→ Channel noise is introduced by the various distrubances available on the channel.

→ Channel noise can be eliminated by using Regenerative Repeaters (RR).

→ Usually no uniform spacing is expected between Regenerative Repeaters over the Channel (It depends on the strength of environmental noises).

Quantization Noise



→ Quantization noise is introduced by the Quantizer.

The maximum quantization error is given by

$$[Q_e]_{max} = \frac{\Delta}{2} \quad \text{--- (1)}$$

where: Δ is the step size

$$\text{Step Size} \downarrow \Delta = \frac{\downarrow V_{max} - V_{min}}{2^{n}\uparrow} \longrightarrow (2)$$

Where: n is the no. of bits per sample used by the Quantizer

→ The Quantization error should be minimum for the proper reconstruction of message signal.

→ To minimize quantization error, step size should be reduced. (see eqⁿ (1))

→ From eqⁿ (2) it is clear that the step size can be reduced either by decreasing dynamic Range ($V_{max} - V_{min}$) or by increasing no. of bits per sample (n).

→ Dynamic Range option is not available in PCM so we have to increase n.

If $n\uparrow \longrightarrow \Delta\downarrow \longrightarrow Q_e\downarrow$

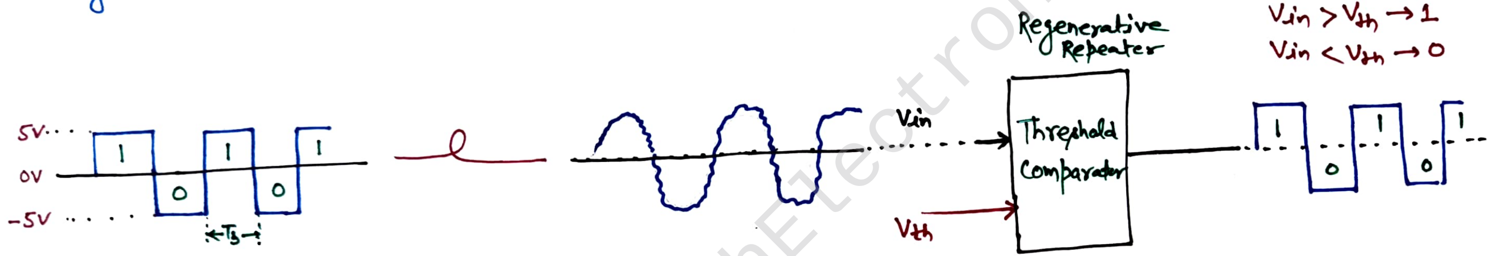
But as $n\uparrow$, Bandwidth Requirment is also increased because

$$\uparrow B.W = \frac{R_b}{2} = \frac{\uparrow n f_s}{2}$$

→ In PCM as Quantization error is decreased Correspondingly channel B.W Requirments are increased this is the major drawback of PCM.

Regenerative Repeater (RR) :-

→ Regenerative Repeater is Responsible for noise free environment in digital communication.



$$V_{th} = \frac{V_{max} + V_{min}}{2} = \frac{5 + (-5)}{2} = 0V$$

→ We can say that if huge amount of noise interefere with transmitted signal, then chances for 1 Received as 0 and 0 Received as 1 can be expected.

Quantization :-

→ A continuous signal such as voice has a continuous range of Amplitudes therefore its samples have a continuous amplitude range.

→ In otherwords we can say, within the finite amplitude range of signal we can find an infinite number of Amplitude levels.

→ It is not necessary to transmit the exact amplitude of the samples because any human sense (the ear or the eye) works as an ultimate Receiver that can detect finite intensity differences.

→ This means that the original continuous signal may be approximated by a signal constructed of discrete amplitudes selected on a minimum error basis from an available set.

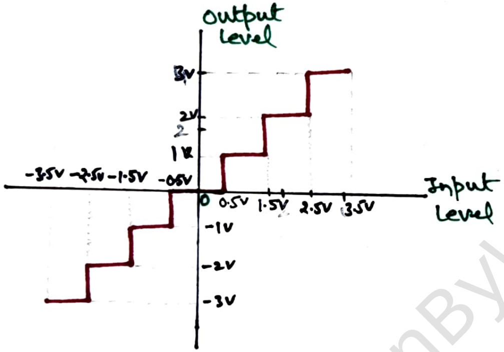
→ Amplitude Quantization is defined as the process of transforming the sample amplitude $m(nT_s)$ of a message signal $m(t)$ at time $t = nT_s$ into a discrete amplitude $m_q(nT_s)$ taken from a finite set of Possible Amplitudes.

→ Quantizer can be of a uniform or non-uniform type.

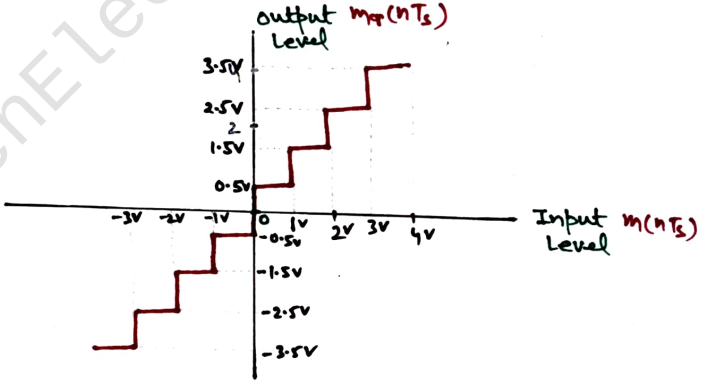
→ In a uniform Quantizer, the representation levels are uniformly spaced otherwise the quantizer is non-uniform.

→ The Quantizer characteristic can be of two type -

(a) midtread (b) midrise



(a) midtread



(b) midrise

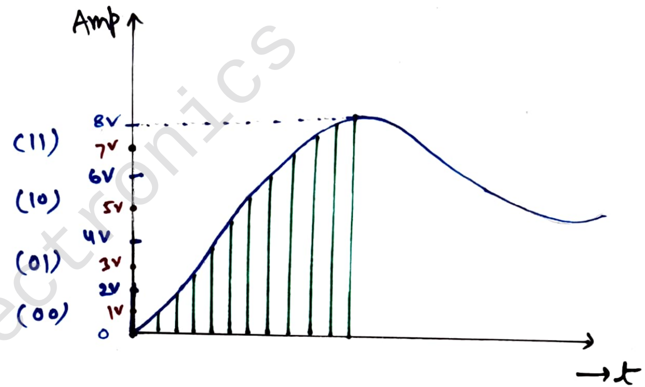
Quantization process:-

→ Total dynamic range of the signal is divided into L equal number of steps.

→ Middle of each step will be selected as quantization voltage

→ Each of voltage corresponding to a step will be rounded off to middle of the step or each of the sample will be rounded off to one of the nearest quantization voltage.

$L = 4$



$\Delta = 2V$

Sample voltage	Q.V	encoded o/p	$Q_e = S.V - Q.V$
0.6 V	1 V	0 0	−0.4 V
1.7 V	1 V	0 0	0.7 V
2.2 V	3 V	0 1	−0.8 V
0 V	1 V	0 0	−1 V → $-\frac{\Delta}{2}$
8 V	7 V	1 1	1 V → $+\frac{\Delta}{2}$

$$[Q_e]_{max} = \pm \frac{\Delta}{2}$$

Sampling theorem :-

A continuous signal bandlimited to f_m Hz can be reconstructed exactly if it is sampled at a rate of atleast twice the maximum frequency component in it.

$$f_s \geq 2f_m \quad \text{or} \quad T_s \leq \frac{1}{2f_m}$$

The minimum required sampled rate $f_s = 2f_m$ is known as Nyquist rate (NR). and $\frac{1}{2f_m}$ is known as Nyquist interval.

→ If $f_s > 2f_m$ or $T_s < \frac{1}{2f_m}$; over sampling

→ If $f_s = 2f_m$ or $T_s = \frac{1}{2f_m}$; critical sampling

→ If $f_s < 2f_m$ or $T_s > \frac{1}{2f_m}$; under sampling

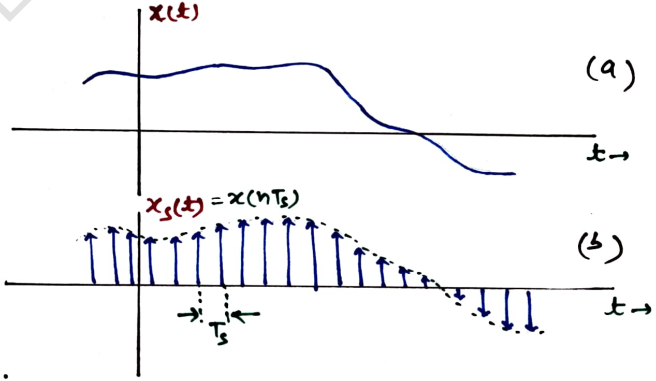
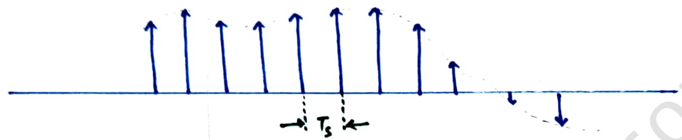
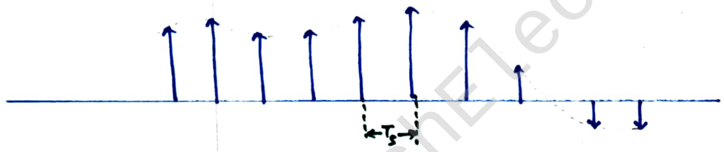
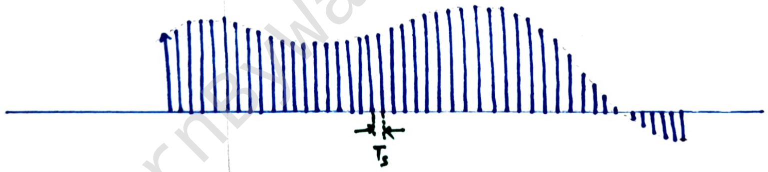


Fig 1.


T_s


T_s


T_s

Proof of sampling theorem –

Consider an arbitrary lowpass signal $x(t)$ shown in fig (a).

Let
$$x_s(t) = x(t)\cdot\left[\sum_{n=-\infty}^{\infty} \delta(t-nT_s)\right]$$

Taking fourier Transform both sides

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

$$\because \; x(t) \otimes \delta(t) = x(t)$$
$$x(t) \otimes \delta(t-t_1) = x(t-t_1)$$

$$\boxed{X_s(f) = f_s \sum_{n=-\infty}^{\infty} X(f-nf_s)}$$

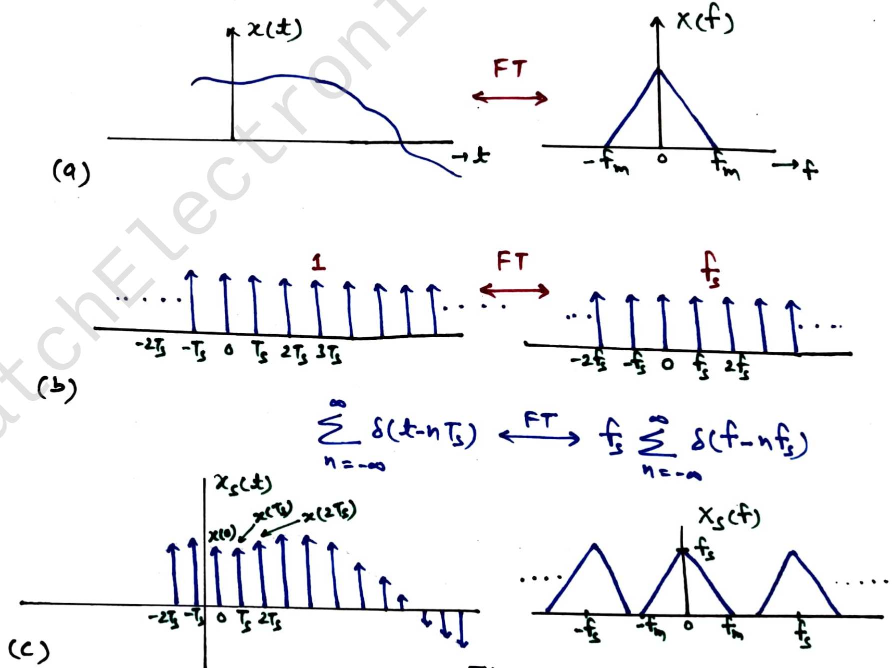
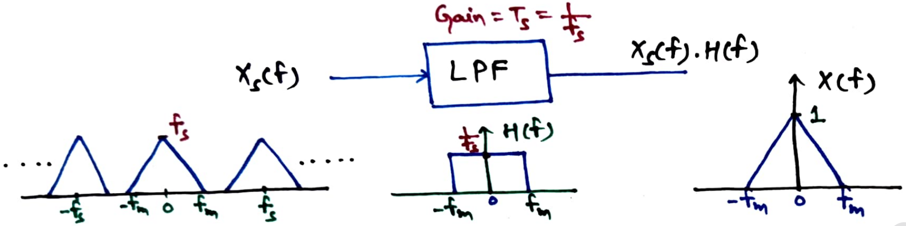
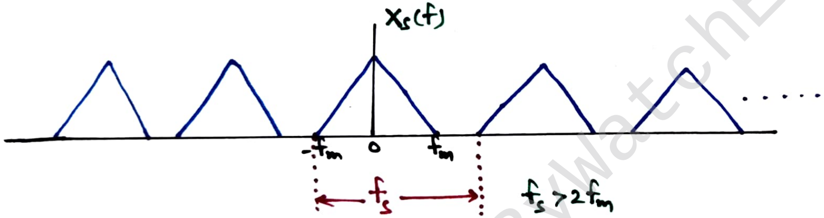


Fig. 2

If $X_s(f)$ is passed through an ideal lowpass filter we can recover $X(f)$ or $x(t)$.

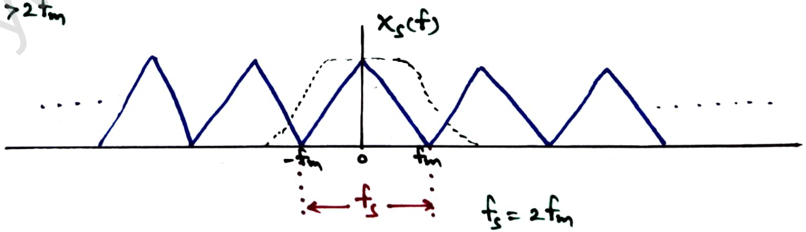


Case (i) $f_s > 2f_m$ or $T_s < \frac{1}{2f_m}$ (over sampling)

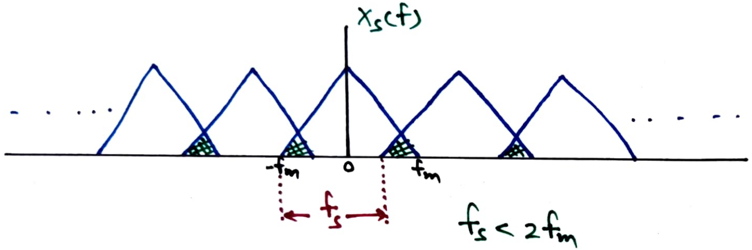


Case (ii) $f_s = 2f_m$ or $T_s = \frac{1}{2f_m}$

(critical sampling)



Case (iii) $f_s < 2f_m$ or $T_s > \frac{1}{2f_m}$ (Under sampling)



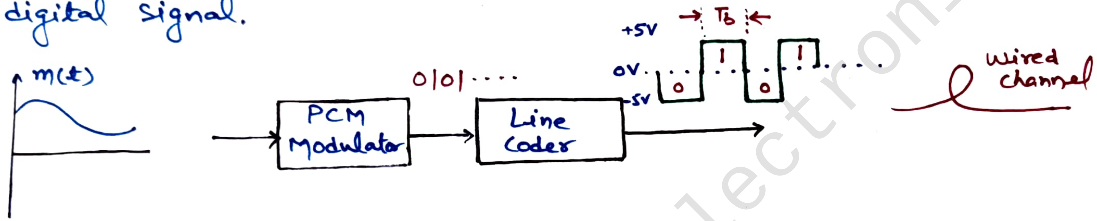
→ Minimum possible sampling rate and maximum sampling interval allowed to avoid aliasing are called as Nyquist rate and Nyquist interval respectively.

$$\text{Nyquist Rate} = 2f_m \text{ samples/sec}$$

$$\text{Nyquist Interval} = \frac{1}{2f_m} \text{ sec}$$

Pulse Digital Modulation Techniques :-

→ Pulse digital modulation Techniques are used to convert a continuous signal to digital signal.



→ PCM modulator convert a continuous signal into a sequence of bits (0's & 1's).

→ These bits can not transmitted directly therefore line coder is used to convert bit sequence in an electrical equivalent signal.

→ To transmit the electrical equivalent of binary signals, wired channel is used.

→ Modem is used to convert a binary signal to electrical signal.

→ If we want transmit digital signals through free space band pass modulation Techniques (ASK, FSK, PSK, ...) are used.

Pulse code modulation (PCM):-

→ The basic form of digital pulse modulation is known as pulse code modulation.

→ In PCM, a message signal is represented by a sequence of coded pulses, which is performed by representing the signal in discrete form in both time and Amplitude.

→ This form of signal representation permits the transmission of the message signal as a sequence of coded binary Pulses, which are less affected by channel noise and can be Regenerate with the help of Regenerative Repeater.

→ The basic operations Performed in the transmitter of a PCM system are Sampling, quantizing and encoding as shown in figure 1.

→ The Low Pass filter prior to sampler is included to prevent aliasing of the message signal so we can say LPF is an anti-aliasing filter.

→ The quantizing and encoding operations are usually performed in the same circuit known as Analog to digital converter.

Pulse Code Modulation (PCM) System:-

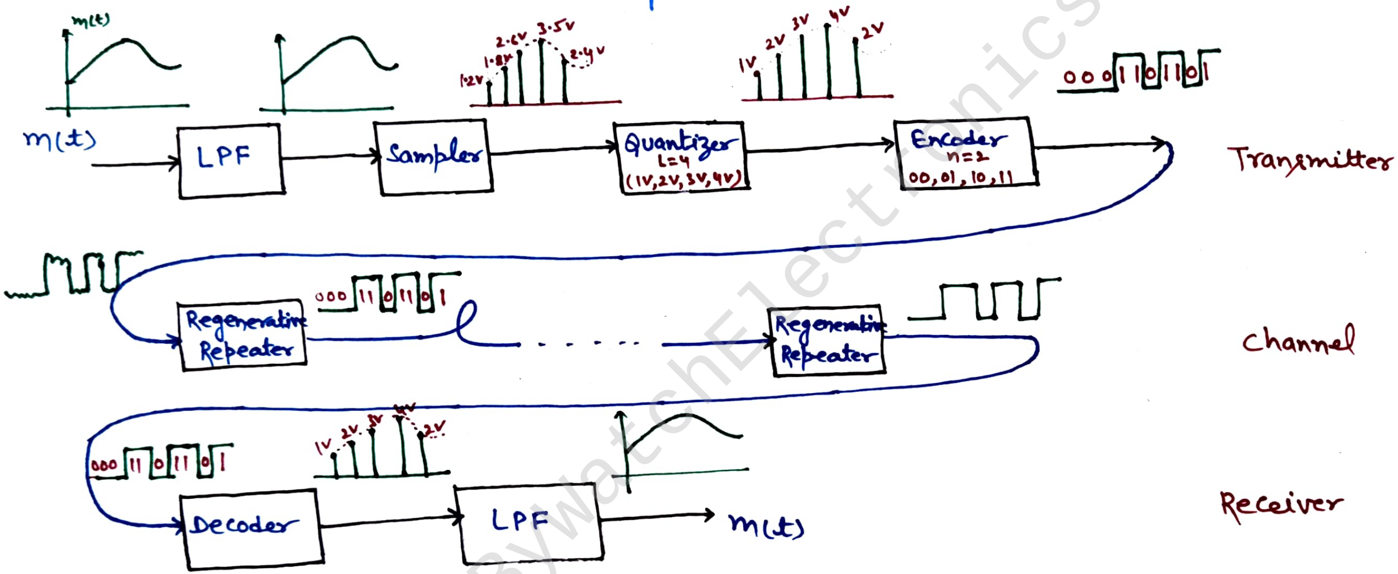


Fig 1. Basic elements of PCM system

→ Quantizer represents each of the sampled voltage by the nearest quantization voltage.

→ Encoder represents each of the quantization voltage by a unique binary code.

→ The no. of quantization levels and binary bit required for encoded follow the relation as -

$$\boxed{L = 2^n}$$

where L : No. of quantization level
n : No. of bits per samples

→ Regenerative Repeaters are used on the channel to eliminate channel noise and regenerate a fresh copy of transmitted signal.

→ At the Receiver side decoder performs the Reverse of encoder.

→ LPF reconstruct continuous signal from its quantized equivalent.

→ At the Receiver, Reverse of quantizer is not possible therefore in the reconstructed signal, finite amount of quantization error will be present permanently.

Pulse Position modulation (PPM)

→ Pulse Position modulation (PPM) is also a pulse analog modulation scheme in which the amplitude and width of the pulses are kept constant, while the position of each pulse, with reference to the position of a reference pulse, is changed according to the instantaneous sampled value of the modulating signal.

→ Thus the transmitter has to send synchronizing pulses to keep the transmitter and Receiver in synchronization.

→ In PPM the Amplitude and width of pulses are constant therefore transmitter handles constant power output, this is an advantage over PWM.

→ Pulse position modulation is obtained from pulse width modulation as shown in fig 1.

→ Each trailing edge of the PWM pulse is a starting point of the pulse in the PPM. Therefore, Position of the pulse is 1:1 proportional to the width of pulse in PWM and hence it is proportional to the instantaneous amplitude of the sampled message signal.

Generation of PPM signal :-

→ Fig 2 shows the block diagram of PPM generator. It consist of PWM generator followed by the monostable multivibrator.

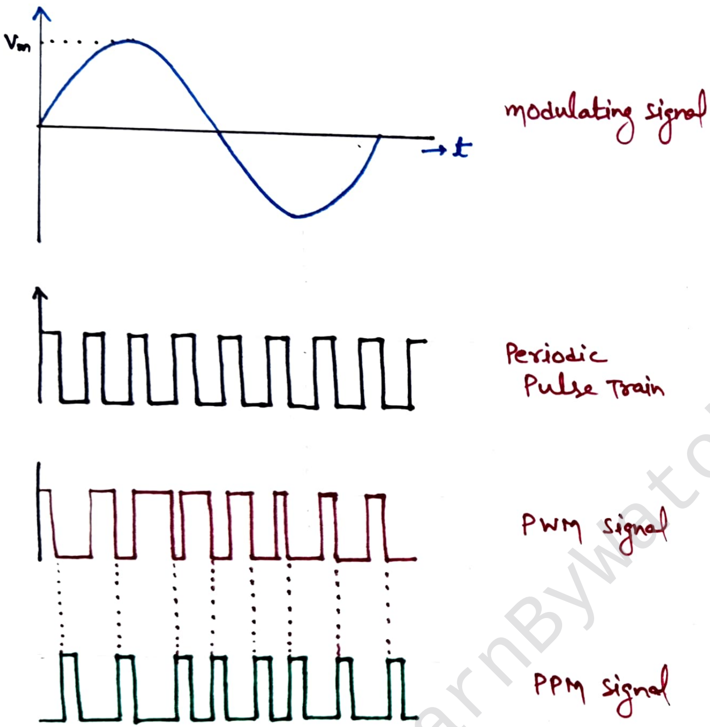


Fig.1.

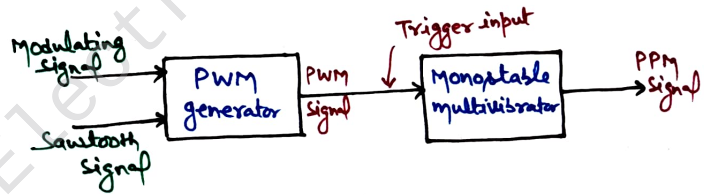


Fig 2. Block diagram of PPM Generator

→ The trailing edge of the PWM signal is used as a trigger input for the monostable multivibrator.

→ The output of monostable multivibrator remains high for fix duration, starting from the trailing edges of the PWM signal.

Demodulation of PPM signal :-

→ For the demodulation of PPM signal, PPM pulses are first converted into corresponding PWM pulses with the help of SR flip-flop as shown in fig3.

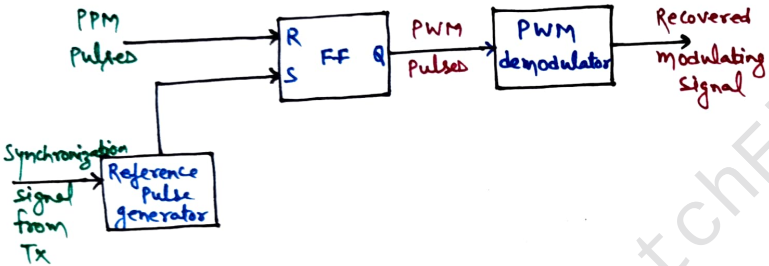


Fig3 PPM demodulator

→ Flip-flop circuit is set or turned ON when the reference pulse arrives.

→ This reference pulse is generated by reference pulse generator of the receiver with the synchronization signal from the transmitter.

→ The flip-flop circuit is reset or turned OFF at the leading edge of the PPM pulse.

→ This repeates and we get PWM pulses at the output of F/F.

→ The PWM pulses are then demodulated by PWM demodulator to get original modulating signal.

Demodulation of PWM signal :-

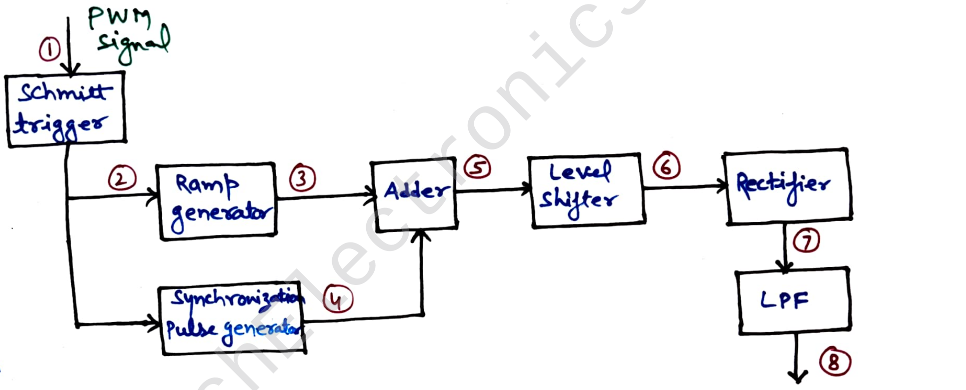


Fig 1. Block diagram of PWM Detector

→ Schmitt trigger circuit removes noise from PWM wave.

→ The ramp generator produces ramp for the duration of pulses such that height of ramps are proportional to the width of PWM pulses.

→ The maximum ramp voltage is retained till the next pulse

→ Synchronization pulse generator produces reference pulses with constant Amplitude and pulse width. These pulses are delayed by specific amount of delay as shown in figure.

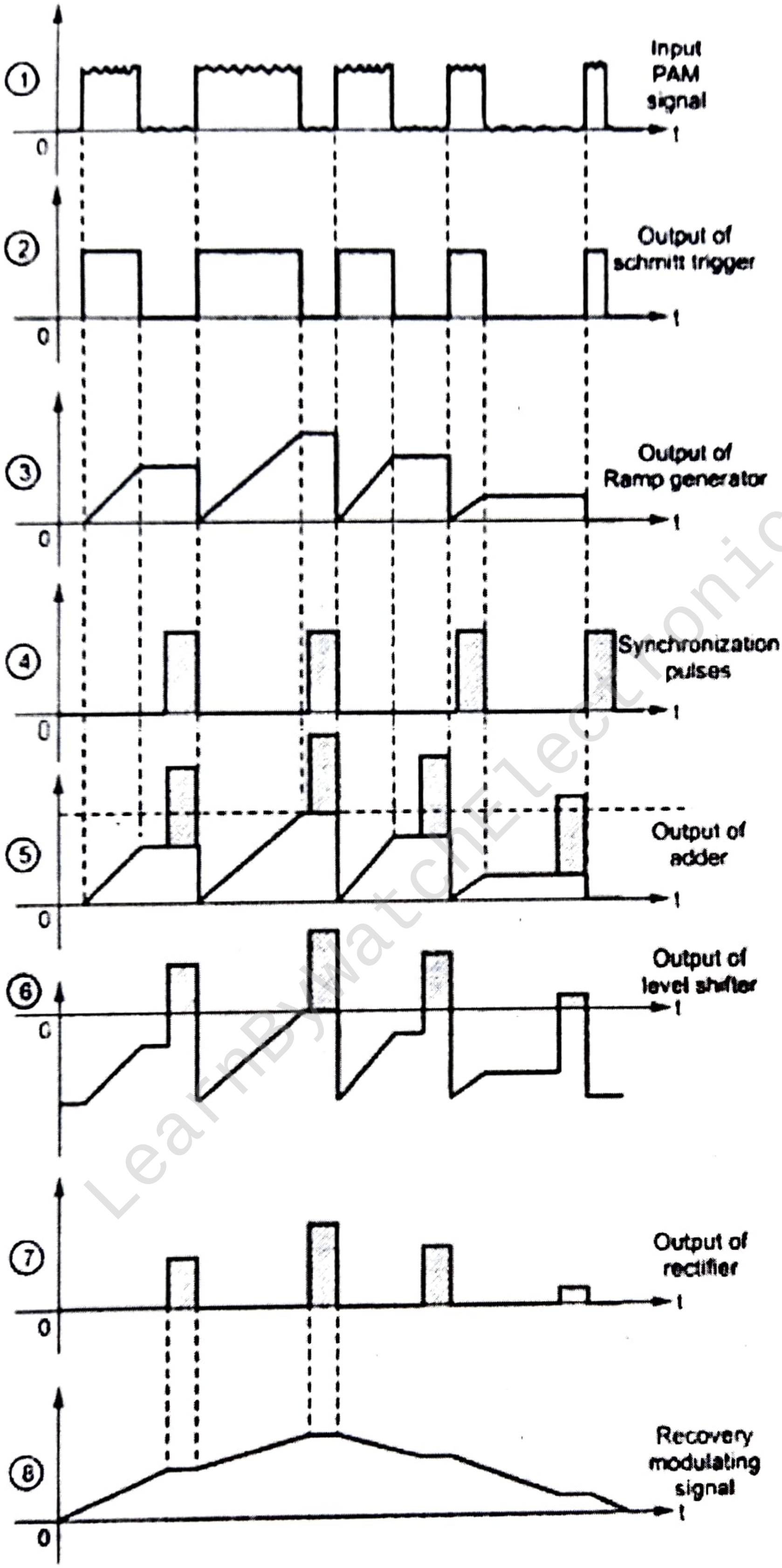

Input PAM signal
Output of schmitt trigger
Output of Ramp generator
Synchronization pulses
Output of adder
Output of level shifter
Output of rectifier
Recovery modulating signal
LearnByWatchElectronics

Fig.2

→ The delayed reference pulses and the output of Ramp generator is added with the help of adder.

→ Level shifter off-set the DC level.

→ Rectifier clipped negative offset waveform.

→ Finally the output of Rectifier is passed through Low Pass filter (LPF) to recover the modulating signal as shown in figure.

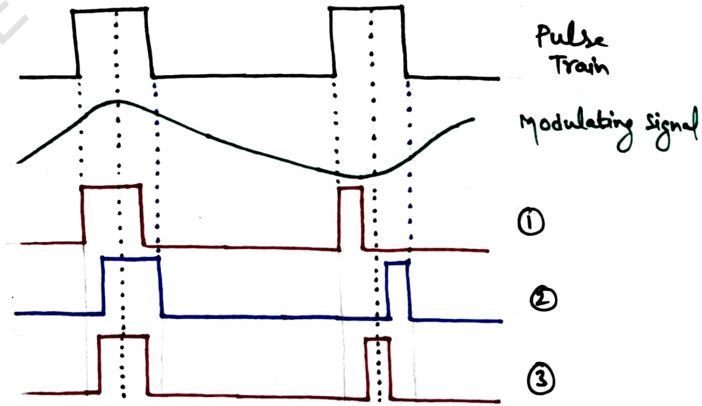
Pulse width Modulation (PWM)

→ As we know PAM signals are affected by Noise, this problem can be solved by using other pulse analog modulation schemes like PWM and PPM.

→ In these schemes the pulse Amplitude is not changed rather some other Parameters are changed.

→ If the duration of pulse or pulse width is changed as per the samples of message signal the resulting modulation is known as pulse duration modulation (PDM) or pulse width modulation (PWM).

→ Three variations of PWM are possible
1. Tail edge of pulse kept constant
2. Leading edge of pulse kept constant
3. Center of pulse kept constant



Generation of PWM :-

- → PWM generator can be designed by using a sawtooth generator and comparator.
- → The sawtooth generator generates a sawtooth signal which is used as a sampling signal
- → The comparator compares the Amplitude of modulating signal and the amplitude of sampling signal i.e sawtooth signal.
- → The output of comparator is high as long as Amplitude of message signal is greater than that of the sawtooth signal.
- → Thus the duration for which the comparator output remains high is directly proportional to the amplitude of the modulating signal.
- → As a result the comparator output is a PWM signal as shown in figure.

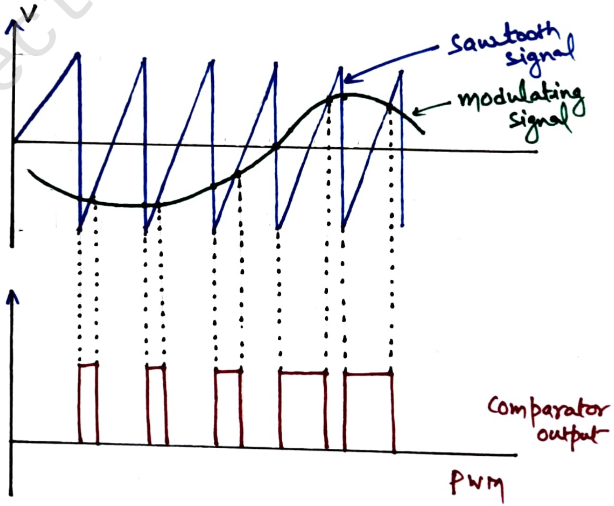
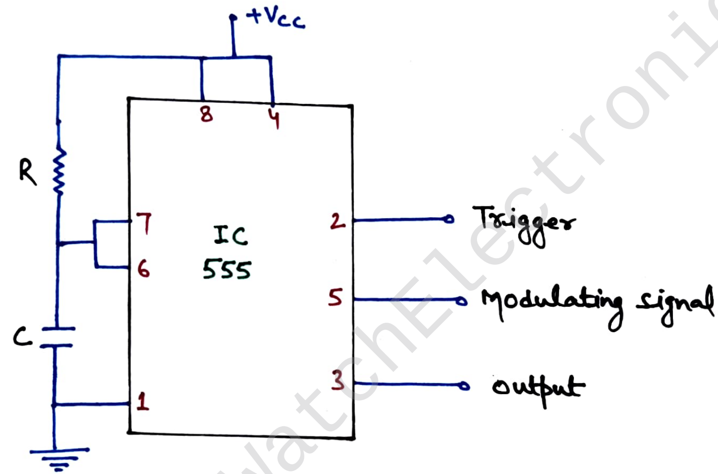


Fig. Waveform of PWM Generator

→ A Practical PWM Generator can be designed with the help of 555-Timer IC as shown below.



→ Basically it is a monostable multivibrator with a modulating input signal applied at the control voltage input.

Pulse Amplitude Modulation (PAM)

→ Pulse Amplitude modulation is a pulse Analog modulation scheme in which the amplitudes of a train of carrier pulses are varied according to the amplitude variations of message signal.

→ In pulse analog modulation pulse by pulse transmission will occured. Each of the pulse to be transmitted corresponds to baseband signal and can be directly transmitted through baseband channel only.

PAM Generation :-

PAM signal can be generated either using natural sampling or using flat-top sampling as shown in fig 1.

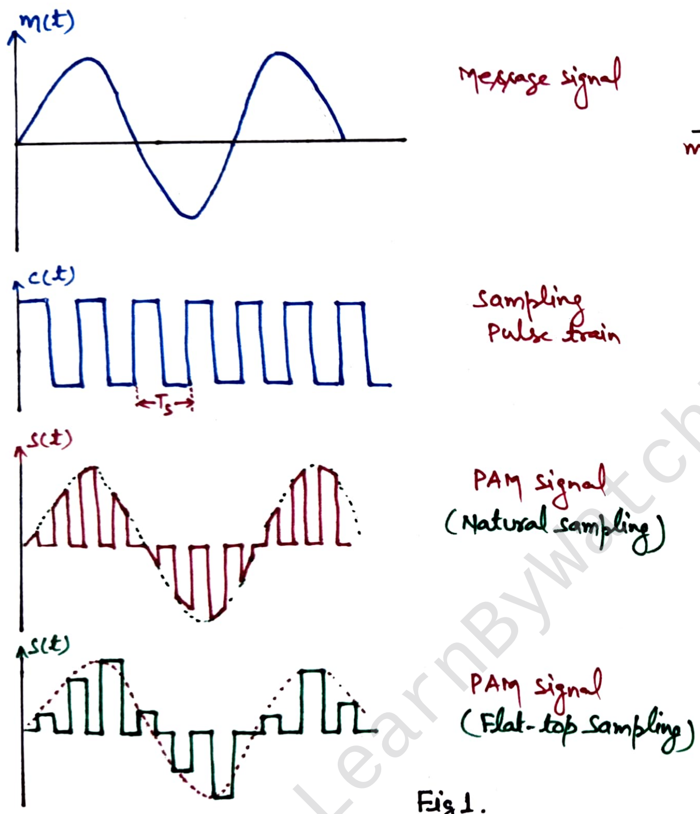


Fig 1.

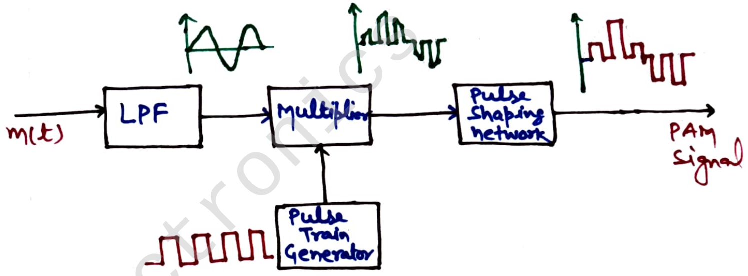


Fig 2. Block diagram of PAM Generator

→ Fig 2 shows the block diagram of PAM Generator.

→ It consist of LPF, a multiplier and a pulse generator.

→ Initially the modulating signal m(t) is passed through LPF to bandlimit the message signal.

→ The band limiting is necessary to avoid aliasing effect in the sampling process.

→ Pulse shaping network does the shaping work to give flat tops.

→ If the generated pulses are narrow, PAM signals require little Power for transmission and are suitable for Time Division multiplexing (TDM).

→ Flat topped Pulses are easily regenerated by Repeaters and can be used for transmission over long distances.

→ One disadvantage of PAM signals is these are affected by noise as much as analog signals.

Reconstruction of message signal:-

→ The original message signal can be detected from PAM signal by Passing PAM signal through a low Pass reconstruction filter with cut-off frequency slightly higher than the maximum frequency in message signal.

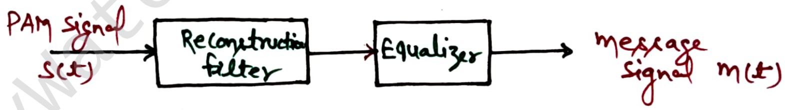


Fig 3. System for Recovering message signal m(t) from PAM signal s(t)

→ The equalizer compensate the aperture effect and attenuation provided by Low Pass Reconstruction filter.

FOM of FM Receiver :-

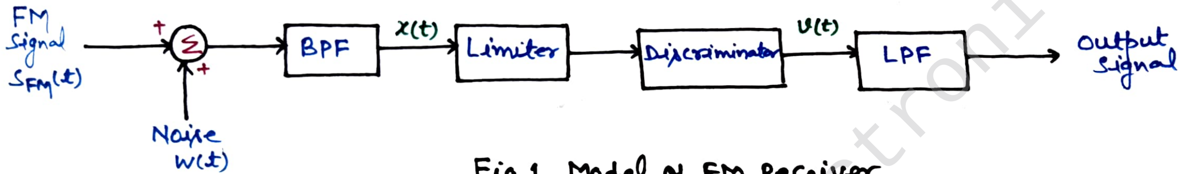


Fig 1. Model of FM Receiver

Figure 1. shows the model of FM Receiver. The noise w(t) is modeled as white gaussian noise of zero mean and power spectral density $\frac{N_0}{2}$.

→ Due to noise, Amplitude of frequency modulated signal may vary that may cause problem for FM demodulator therefore before demodulation, the Amplitude limiter is used which clipped the Amplitude variations.

The incoming FM signal $S_{FM}(t)$ is defined as

$$S_{FM}(t) = A_c \cos\left[2\pi f_c t + \underbrace{2\pi K_f \int_0^t m(t)\,dt}_{\phi(t)}\right]$$

The filtered noise $n(t)$ is expressed as

$$n(t) = n_I(t)\cos 2\pi f_c t - n_Q(t)\sin 2\pi f_c t$$

we may express $n(t)$ in terms of its envelope and Phase as -

$$n(t) = r(t)\cos\left[2\pi f_c t + \psi(t)\right]$$

where

envelope is $r(t) = \sqrt{n_I^2(t) + n_Q^2(t)}$

Phase $\psi(t) = \tan^{-1}\left[\frac{n_Q(t)}{n_I(t)}\right]$

$$x(t) = A_c\cos\left[2\pi f_c t + \phi(t)\right] + r(t)\cos\left[2\pi f_c t + \psi(t)\right]$$

After lot of Assumptions and Calculation the input SNR and output SNR for FM Receiver can be written as

$$\frac{S_i}{N_i} = \frac{A_c^2}{2WN_0}$$

$$\frac{S_o}{N_o} = \frac{3A_c^2 k_f^2 P}{2N_0 W^3}$$

$$FOM = \frac{(SNR)_{O/P}}{(SNR)_{I/P}}$$

$$= \frac{3A_c^2 k_f^2 P}{2N_0 W^3} \Big/ \frac{A_c^2}{2WN_0}$$

$$FOM = \frac{3k_f^2 P}{W^2}$$

where:

$P \rightarrow$ message signal Power

$W \rightarrow$ message signal BW

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

$$P = \frac{A_m^2}{2} \quad , \quad W = f_m \text{ for single tone FM}$$

$$FOM = \frac{3 k_f^2 \cdot \frac{A_m^2}{2}}{f_m^2} = \frac{3}{2}\left(\frac{k_f A_m}{f_m}\right)^2$$

$$\boxed{FOM = \frac{3}{2}\beta^2}$$

NBFM

$\beta \leq 1$ (small)

FOM is small

WBFM

$\beta > 1$

$FOM > 1$

As $\beta \uparrow \longrightarrow FOM \uparrow$

But $\beta \uparrow \longrightarrow B.W \uparrow = 2(\beta+1) f_m$

Generally $\beta_{max} = 10$

$$FOM = \frac{3}{2} \times 100 = 150$$

→ Wideband FM preffered over NBFM because of its high figure of merit (FOM).

FOM of AM Receiver:-

In case of AM signal, both sidebands and the carrier wave are transmitted

$$S_{AM}(t) = A_c[1 + k_a m(t)]\cos 2\pi f_c t$$

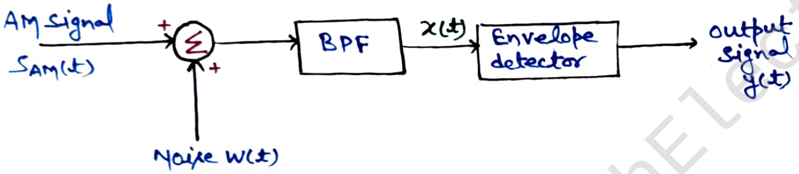


Fig 1. Model of AM Receiver

$$S_{AM}(t) = A_c \cos 2\pi f_c t + A_c k_a m(t)\cos 2\pi f_c t$$

The average Power of AM signal $S_{AM}(t)$ will be

$$S_i = \frac{A_c^2}{2} + \frac{A_c^2 k_a^2 \overline{m^2(t)}}{2} = \frac{A_c^2}{2}[1 + k_a^2 P]$$

where $P = \overline{m^2(t)}$ = Power of $m(t)$

Input Noise Power $N_i = N_0 W$ (white noise power affecting message signal)

Input SNR will be

$$\frac{S_i}{N_i} = \frac{\frac{A_c^2}{2}(1+k_a^2 P)}{N_0 W} = \frac{A_c^2}{2N_0 W}(1+k_a^2 P) \quad \text{--- (1)}$$

From fig 1. the envelope detector input is represented by $x(t)$

$$x(t) = S_{AM}(t) + n(t)$$

$$= A_c \cos 2\pi f_c t + A_c k_a m(t) \cos 2\pi f_c t + n_I(t) \cos 2\pi f_c t - n_Q(t) \sin 2\pi f_c t$$

$$= \underbrace{[A_c + A_c k_a m(t) + n_I(t)]}_{A} \cos 2\pi f_c t - \underbrace{n_Q(t)}_{B} \sin 2\pi f_c t$$

For an envelope detector

$$A \cos 2\pi f_0 t - B \sin 2\pi f_0 t \xrightarrow{\text{Envelope detector}} \sqrt{A^2+B^2}$$

$$(ED)_{o/p} = \sqrt{(A_c + A_c k_a m(t) + n_I(t))^2 + n_Q^2(t)}$$

$$(ED)_{o/p} \cong \underbrace{A_c k_a m(t)}_{\text{signal}} + \underbrace{n_I(t)}_{\text{Noise}}$$

Output signal Power S_o = Power of $[A_c k_a m(t)]$

$$= A_c^2 k_a^2 \overline{m^2(t)}$$

$$= A_c^2 k_a^2 P$$

Output Noise Power N_o = Power of $[n_I(t)]$

$$= \overline{n_I^2(t)} = 2 N_o W \text{ watt}$$

$$\frac{S_o}{N_o} = \frac{A_c^2 k_a^2 P}{2 N_o W} \quad \text{——— } (2)$$

From eqn (1) and eqn (2)

$$FOM = \frac{S_o/N_o}{S_i/N_i} = \frac{\cancel{A_c^2} k_a^2 P / \cancel{2 N_o W}}{\cancel{A_c^2}(1+k_a^2 P)/\cancel{2 N_o W}} = \frac{k_a^2 P}{1+k_a^2 P}$$

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

$$P = \frac{A_m^2}{2}$$

$$FOM = \frac{k_a^2 \times \frac{A_m^2}{2}}{1 + k_a^2 \times \frac{A_m^2}{2}} = \frac{(k_a A_m)^2}{2 + (k_a A_m)^2}$$

$$\boxed{FOM = \frac{\mu^2}{2+\mu^2}}$$

As $\mu \uparrow$, FOM $\uparrow$

$$[FOM]_{max} = \frac{1}{3} \quad \text{for } \mu = 1$$

$$\frac{S_o}{N_o} = \frac{1}{3}\frac{S_i}{N_i}$$

→ The performance of envelope detector towards channel noise is not good.

FOM of DSB-SC Receiver:-

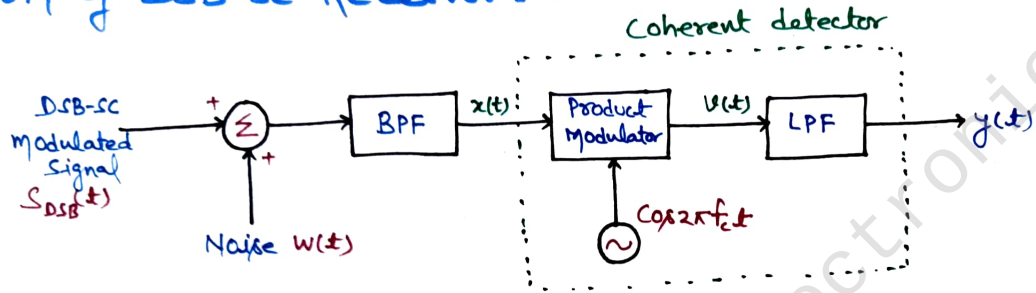


Fig 1. DSB-SC Receiver model

Consider DSB-SC Receiver model using Coherent detector as shown in fig 1.

The output of Band Pass filter is

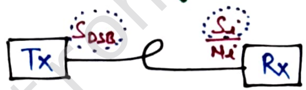
$$x(t) = S_{DSB}(t) + n(t)$$

The DSB-SC component of the filtered signal $x(t)$ expressed as

$$S_{DSB}(t) = A_c\, m(t) \cos 2\pi f_c t$$

Input signal Power $\quad S_i = \dfrac{A_c^2\, \overline{m^2(t)}}{2} = \dfrac{A_c^2 P}{2}$

Where: $\overline{m^2(t)}$ = Power of $m(t) = P$



Input Noise Power

N_i = white Noise Power affecting message signal

$N_i = N_0 W$ watt

Input SNR Can be Calculate using S_i & N_i

$$\frac{S_i}{N_i} = \frac{\frac{A_c^2 P}{2}}{N_0 W} = \frac{A_c^2 P}{2 N_0 W} \quad \text{——— (1)}$$

From fig 1. the output of product modulator given by

$$v(t) = \left(S_{DSB}(t) + n(t)\right) \cos 2\pi f_c t$$

$$= \left(A_c m(t) \cos 2\pi f_c t + n_I(t) \cos 2\pi f_c t - n_Q(t) \sin 2\pi f_c t\right) \cos 2\pi f_c t$$

$$= A_c m(t) \cos^2 2\pi f_c t + n_I(t) \cos^2 2\pi f_c t - \frac{n_Q(t)}{2} \sin 4\pi f_c t$$

$$v(t) = \frac{A_c m(t)}{2}(1+\cos 4\pi f_c t) + \frac{n_I(t)}{2}(1+\cos 4\pi f_c t) - \frac{n_Q(t)}{2}\sin 4\pi f_c t$$

$$(LPF)_{O/P} = \underbrace{\frac{A_c m(t)}{2}}_{\text{signal}} + \underbrace{\frac{n_I(t)}{2}}_{\text{noise}} \qquad \text{——} \quad (2)$$

Equation (2) indicates that -

→ The message signal $m(t)$ and In-phase noise component $n_I(t)$ of the filtered noise $n(t)$ appear additively at the Receiver output.

→ The quadrature component $n_Q(t)$ of the filtered noise $n(t)$ is completely rejected by the coherent detector

output signal Power S_o = Power of $\left[\frac{A_c m(t)}{2}\right] = \frac{A_c^2\, \overline{m^2(t)}}{4} = \frac{A_c^2 P}{4}$

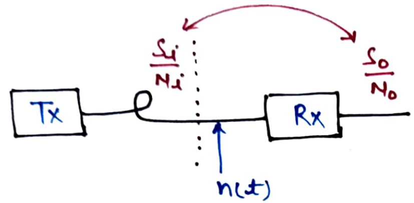
output Noise Power N_o = Power of $\left[\frac{n_I(t)}{2}\right] = \frac{\overline{n_I^2(t)}}{4} = \frac{2N_oW}{4} = \frac{N_oW}{2}$

output SNR

$$\frac{S_o}{N_o} = \frac{\frac{A_c^2 P}{4}}{N_oW/2} = \frac{A_c^2 P}{2N_oW} \qquad \text{——} \quad (3)$$

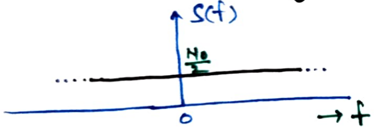
$$FOM = \frac{S_o/N_o}{S_i/N_i} = 1$$

$$\frac{S_o}{N_o} = \frac{S_i}{N_i}$$

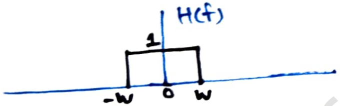


Receiver is cancelling the effect of channel noise.

PSD of white noise affecting message signal :-

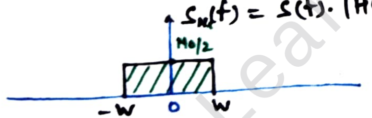


White Noise PSD



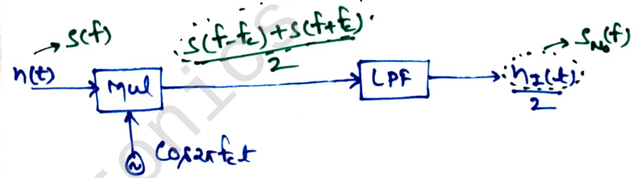
LPF

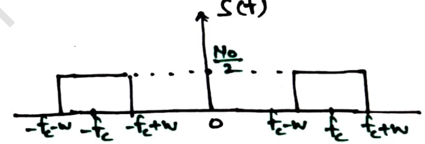
$$S_{Ni}(f) = S(f) \cdot |H(f)|^2$$

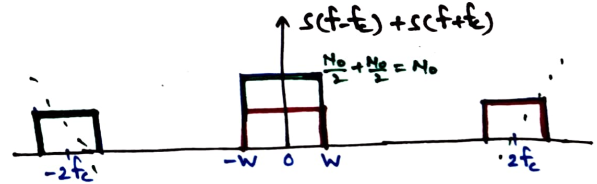


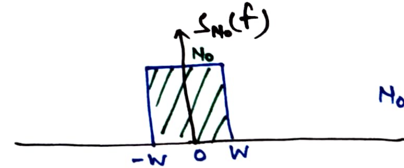
$$N_i = \frac{N_0}{2} \times 2W = N_0 W$$

PSD of $n_I(t)$ affecting AM & DSB-SC :









$$N_{out} = N_0 \times 2W = 2N_0 W$$

Noise :-

Noise is an unwanted electrical or electromagnetic energy that interferes with the transmitted message and degrade the quality of message signal.

Noise

- Internal Noise
  - ① Thermal Noise or White Noise or Johnson noise
  - ② Shot Noise
  - ③ Transit time noise
  - ④ Miscellaneous internal Noise : flicker Noise, Partition Noise etc.
- External Noise
  - ① Atmospheric Noises
  - ② Extraterrestrial Noises
  - ③ Man made Noises or Industrial Noises

→ Internal noises generated within the Receiver or Communication system.

→ External Noises generated by the external sources.

Thermal Noise :-

This type of Noise is generated by all resistances (e.g a resistor, Semiconductor, the resistance of resonant circuit i.e the Real Part of impedance, cable etc.)

Due to thermal agitation, the molecules in the electrical component gain energy, moves in random fashion and collide each other therefore Produces heat and this heat produced is corresponds to the thermal Noise.

Thermal noise increases with temperature and Resistance value.

Thermal Noise Power $\boxed{N = \overset{\text{Watt/Hz}}{KT} \cdot \overset{\text{Hz}}{B}}$ watt

where

K: Boltzman Constant 1.38×10^{-23} Joules/K

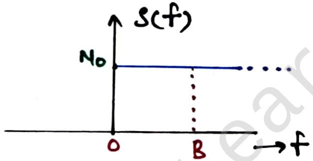
T: Absolute Temperature

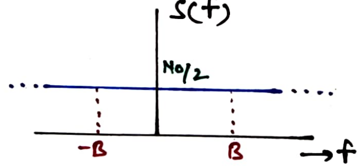
B: Bandwidth

$KT = N_0 =$ Power spectral density

Thermal Noise $N = N_0 B$ watt

One-sided PSD





Double Side PSD

Shot Noise :-

Shot noise is produced by the random movement of electrons or holes across a PN Junction.

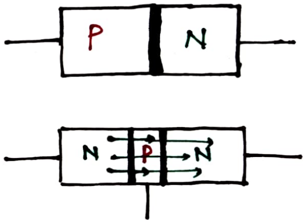
→ The electrons or holes enter the junction region from one side, drift or are accelerated across the Junction and are collected on the other side.

→ The average Junction current determines the average interval that elapses between the times when two successive carriers enter the junction.

– However the exact interval that elapses is subject to ~~radom~~ random statistical fluctuations.

– This randomness gives rise to a type of noise which is referred to as shot noise.

→ shot noise is also encountered as a result of the randomness of emission of electrons from a heated surface.



Transit Time Noise:-

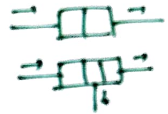
- This noise occurs in the transistors.
- Transit time is the time duration that is taken by current carrier such as e^- or hole to move from the input to the output.
- At low frequencies this time is negligible. But when the frequency of operation is high, then problem arises.
- The transit time shows up as a kind of random noise within the device, and this is directly proportional to the frequency of operation.

Flicker Noise:-

- Flicker Noise is also known as modulation noise or Pink noise.
- This noise is inversely proportional to the frequency.
- Hence for the frequencies above about 500Hz, this noise does not creat serious problem.

Partition Noise :-

Partition noise occurs whenever current has to divide between two or more paths, and results from the random fluctuations in the division.



→ Due to this noise diode would be less noisy than a transistor.

External Noises :-

Atmospheric Noise :-

Atmospheric noise is caused by lightning discharges in thunderstorms and other natural electrical disturbances occuring in the atmosphere.

→ These electrical impulses are random in nature Hence the energy is spread over the complete frequency spectrum used for radio communication.

→ Atmospheric noise consist of spurious radio signals with components spread over a wide frequency range.

→ These spurious wave propagated over earth in the same fashion as the desired wave propagated therefore picked up by the Receiving antenna.

→ Large Atmospheric noise is generated in low and medium frequency bands, while very little noise is generated in VHF and UHF bands Therefore, the atmospheric noise becomes less severe at frequencies above 30 MHz.

Extraterrestrial Noise :-

It is divided in two Groups – ① Solar Noise
② Cosmic Noise

(i) Solar Noise –

→ This is the electrical noise emanating from the sun.

→ The sun is a large body at a very high temperature (exceeding 6000°c on the surface) and radiates electrical energy in the form of noise over a very wide frequency spectrum used for radio communication.

→ The intensity of Noise produced by sun Varies with time.

→ Sun has a repeating 11-year noise cycle.

→ During peak of noise cycle, high amount of noise is produced that causes tremendous radio signal interference, and many frequencies becomes unusable for communication.

Cosmic noise :- This noise is generated by distant stars having high temperature.

→ The noise received from distant stars is thermal noise and is distributed almost uniformly over entire sky.

Man-made Noise or Industrial Noise :-

Industrial noise is an electrical noise produced by the sources such as- automobiles and aircraft ignition, electrical motors and switch gears, leakage from high voltage lines etc.

→ Such noises are produced by the arc discharge taking place during the operation of heavy electrical machines.

→ Man-made Noise is most intensive in industrial and densely populated areas.