PULSE MODULATION:-

1.1 Introduction:-

In analog modulation message signal and carrier signal both are analog in nature. In case of Pulse modulation message signal is analog but train of discrete pulses act as carrier. Like analog carrier pulse carrier also have some parameters such that amplitude, width and. position.

Definition:-

"Characteristics or parameters of pulse carrier is varied in accordance with amplitude of analog message signal".

Types of pulse Modulation.

- Analog Pulse Modulation
  - PAM
  - P.T.M
    - PWM
    - PPM
- Digital Pulse Modulation
  - PCM
  - DM
  - ADM
  - DPCM

* In analog pulse modulation, some parameters of pulse carrier is varying with respect to sample value of message signal.

* Transmission of takes place at discrete times.

* In digital pulse modulation, message signal is transmitted in the form of code words.

* Digital pulses are combined together to form a codewords. Thus the transmitted signal in digital pulse communication is a digital signal.

Analog Pulse Modulation:-

* Analog pulse modulation is a technique in which each sample of analog message signal systematically changes the characteristics of a pulse carrier. (Amplitude, Width, Position).

* Analog pulse modulation classified in to three types.

1. Pulse Amplitude Modulation (PAM)
2. Pulse Width Modulation (PWM)
3. Pulse Position Modulation (PPM)

1.2.1 Sampling Theorem:-

"A bandlimited signal having no spectral components above 'f_m' Hz can be determined uniquely by values sampled at uniform intervals of $T_s \leq \frac{1}{2f_m}$ sec (T_s is Sampling time."

* The Nyquist rate of sampling gives the minimum sampling frequency needed to reconstruct the analog signal from Sampled waveform.

$$\boxed{f_s \geq 2f_m}$$

Where, 'f_s' → Sampling frequency

f_m → Maximum frequency of Inpu

* Nyquist interval is Reciprocal of nyquist rate. That is $1/f_s$.

* Time interval Between two adjecent Samples is also said to be Nyquist interval.

1.2 Sampling :-

Formatting an analog signal proceeds in three steps.

1. Discretisation in time, which is known by the name of sampling.
2. Discretisation in amplitude, which is known as quantization.
3. Encoding - encoding the quantised values.

Sampling :-

Sampling discretizes an analog signal in time domain so that instead of a continuous time waveform we get a continuous values of the signal at discrete points of time.

Definition :-

Analog signal is converted in to discrete time signal.

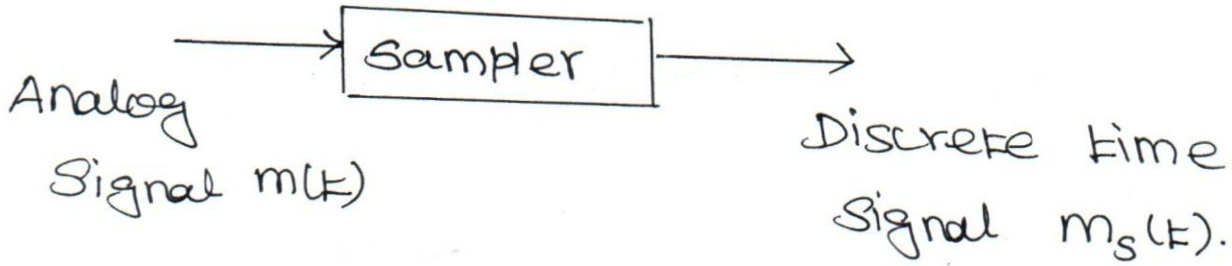


fig: 1 SAMPLER DIAGRAM.

1.2.2 Types of sampling:-

- Instantaneous Sampling
- Flat top Sampling
- Natural Samplir

Instantaneous Sampling:-

* Instantaneous sampling is also called as Ideal sampling or Impulse sampling.

* An analog signal m(t) is sampled by a sequence of unit impulses (or) Dirac delta function.

Natural Sampling:-

* An analog signal m(t) is sampled by flat top rectangular pulses with finite width.

* The top of each pulse in the sampled signal retains the shape of original signal.

Flat top Sampling:-

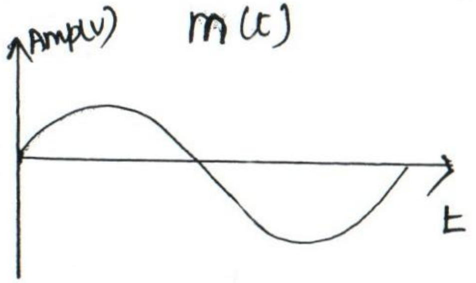
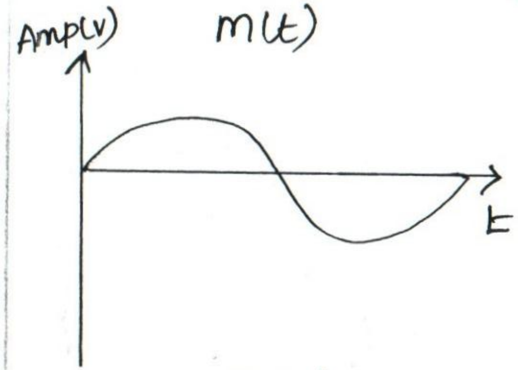
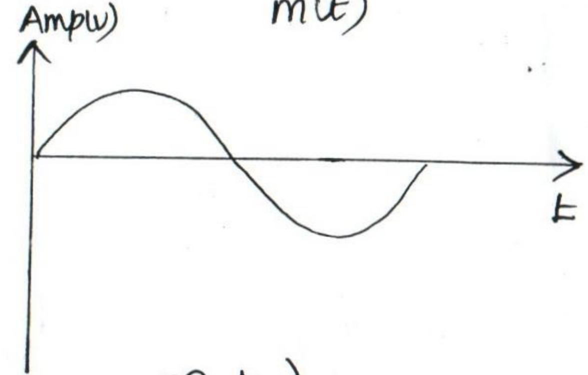
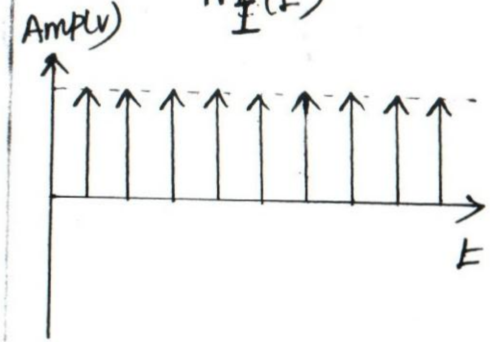
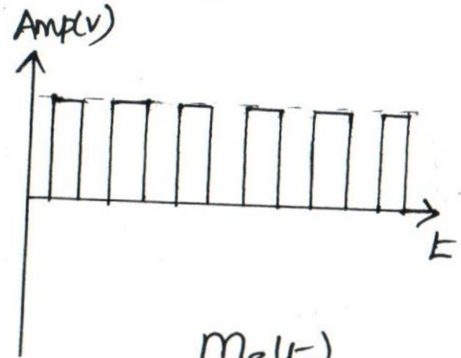
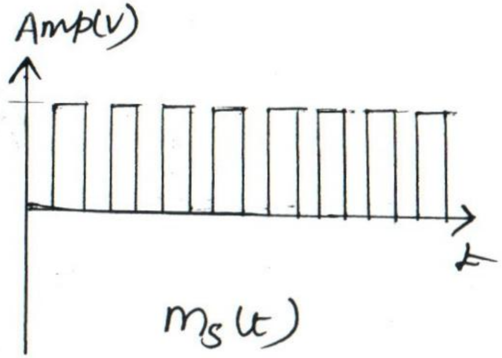
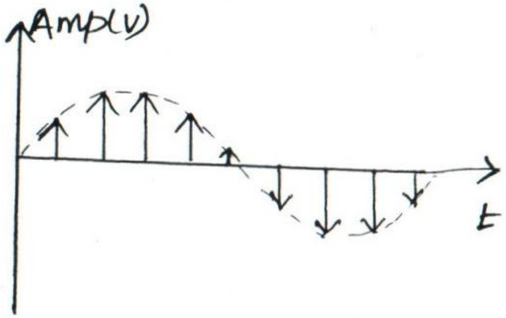
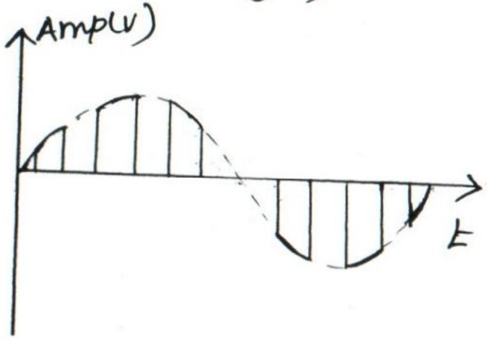
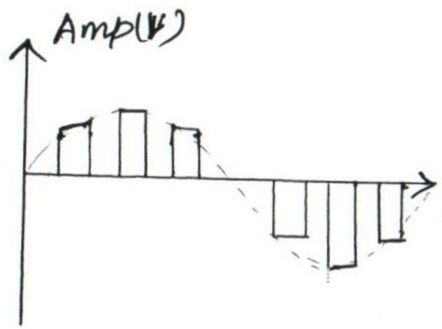
* An analog signal m(t) is sampled by flat top rectangular pulses

* The top of each pulse in the sampled signal is flat.

1.2.3 Comparision of Various Sampling Techniques:

S.NO	Impulse sampling	Natural sampling	Flat top sampling
1.	Unit impulses are used to take the samples of analog signal $m(t)$.	Rectangular pulses are used to take the samples of analog signal $m(t)$.	Rectangular pulses are used to take the samples of analog signal $m(t)$.
2.	Sampling principle is multiplication	Sampling principle of this method is chopping	Sampling principle of this method is sample and hold circuit.
3.	Sampling rate tends to infinity.	Sampling rate satisfies nyquist criteria	Sampling rate satisfies nyquist criteria.
4.	Noise interference is maximum	Noise interference is Maximum	Noise interference is minimum.

Comparision of Various Sampling Techniques.

S.NO:-	Impulse Sampling	Natural Sampling	Flat top Sampling
Input Signal	$m(t)$ 	$m(t)$ 	$m(t)$ 
(F) Carrier Signal	$m_I(t)$ 	$m_p(t)$ 	$m_p(t)$ 
Sampled Signal.	$m_s(t)$ 	$m_s(t)$ 	$m_s(t)$ 

1.2.4 Limitations of Sampling:-

Practically band limiting is not possible to achieve if signal is bandlimited overlapping of signals occurs.

Limitations of sampling is referred as Aliasing.

Aliasing:-

If the sampling frequency "f_s" is less than Nyquist rate ($2f_m$), then a type of distortion referred as aliasing occurs

i-e $\boxed{f_s < 2f_m}$

The interference of high frequency components with low frequency components in the spectrum of sampled version is called Aliasing.

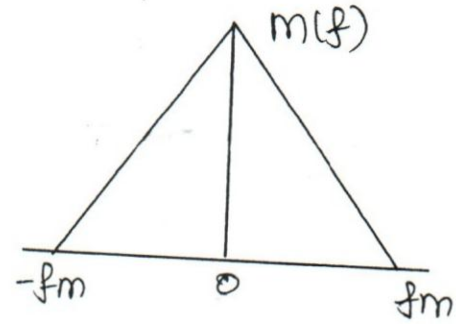
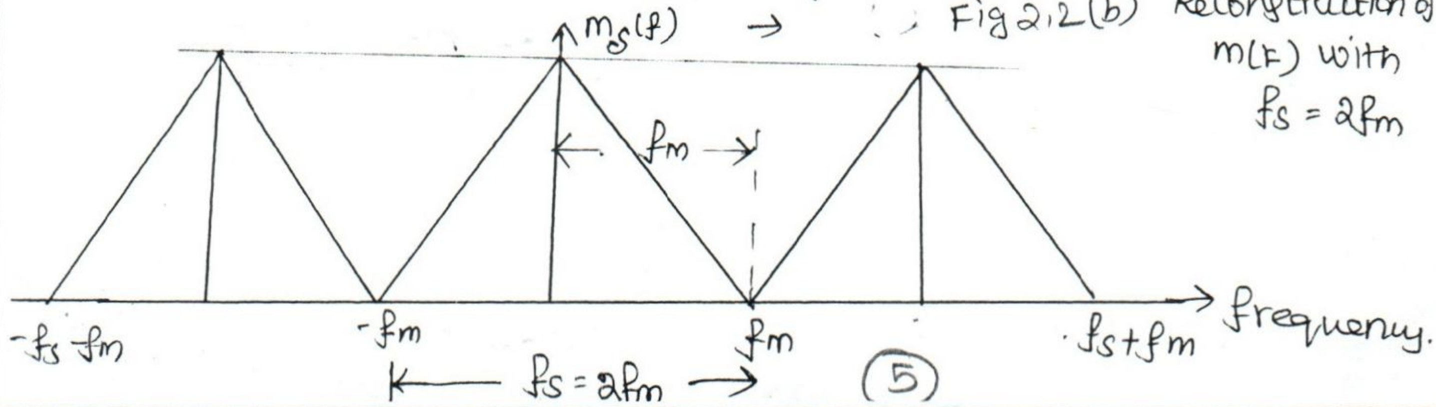


Fig 2.1(a) Spectrum of a signal.

Fig 2.2(b) Reconstruction of m(f) with $f_s = 2f_m$



Sam-7

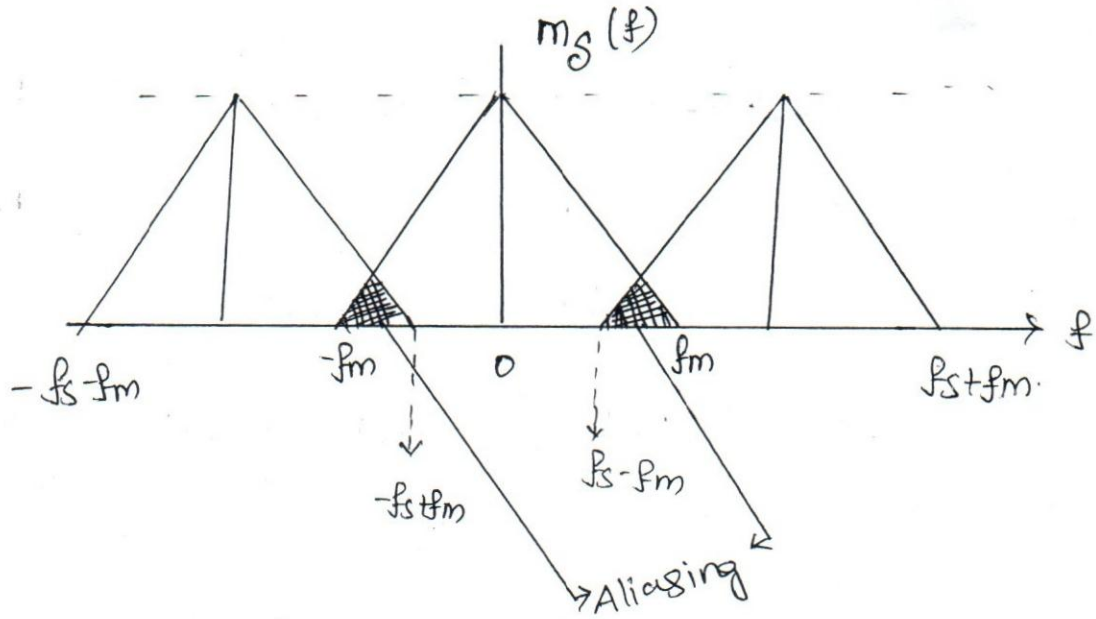


Fig 2.3 (c) Reconstruction of m(t) with $f_s < 2f_m$

* To avoid aliasing, prior to sampler low pass anti aliasing filter is used, it attenuat high frequency components.

* Filtered signal is sampled by selecting a sampling rate slightly greater than '$2f_m$'.

1.3 PULSE AMPLITUDE MODULATION:-

Definition:

Amplitude of the pulse carrier is Varied with respect to the amplitude of the message signal. where as width and position of the pulse carrier remains constant.

1.3.1 Generation of PAM:-

* The message signal m(t) is given to the low pass filter.

Low pass filter:-(LPF)

* LPF is used to band limit the message signal m(t) to the maximum frequency 'fm'.
* (i-e) It attenuate the frequency higher than 'fm'
* LPF also act as anti aliasing filter to avoid aliasing.

Pulse train Generator:-

* It produces a periodic pulse train of frequency 'fs'. where $f_s > 2f_m$ hence Nyquist Criteria is satisfied.

Multiplier:-

Sampling takes place at the block of multiplier and it generates PAM signal.

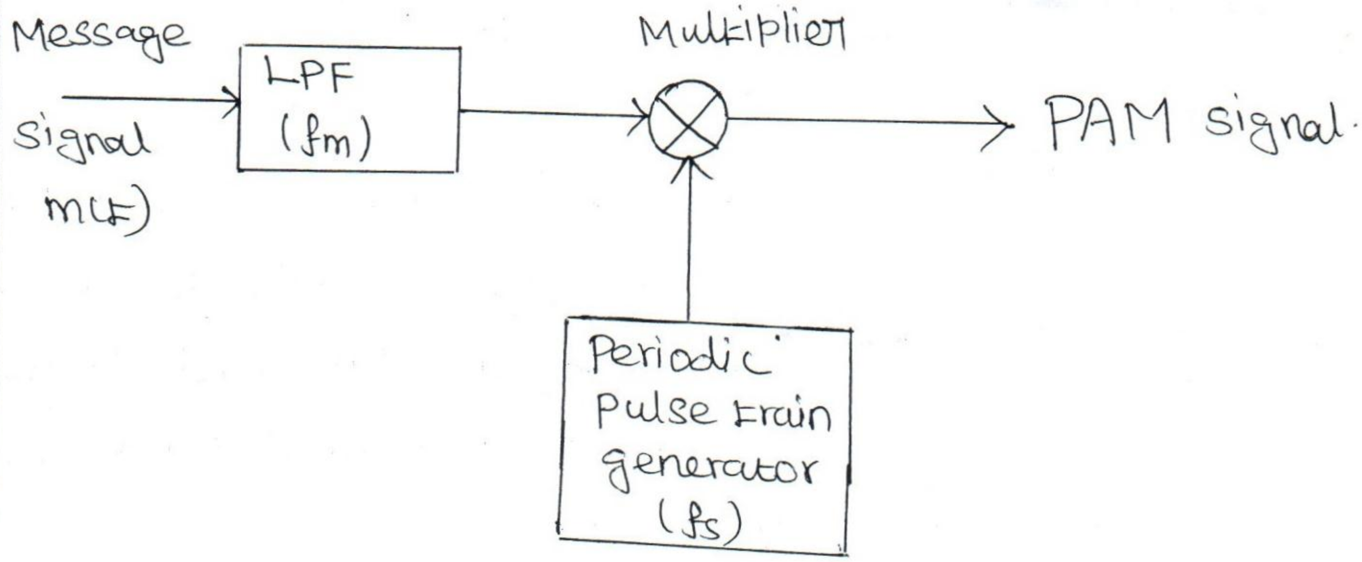


Fig: 3 Block diagram of PAM Generator.

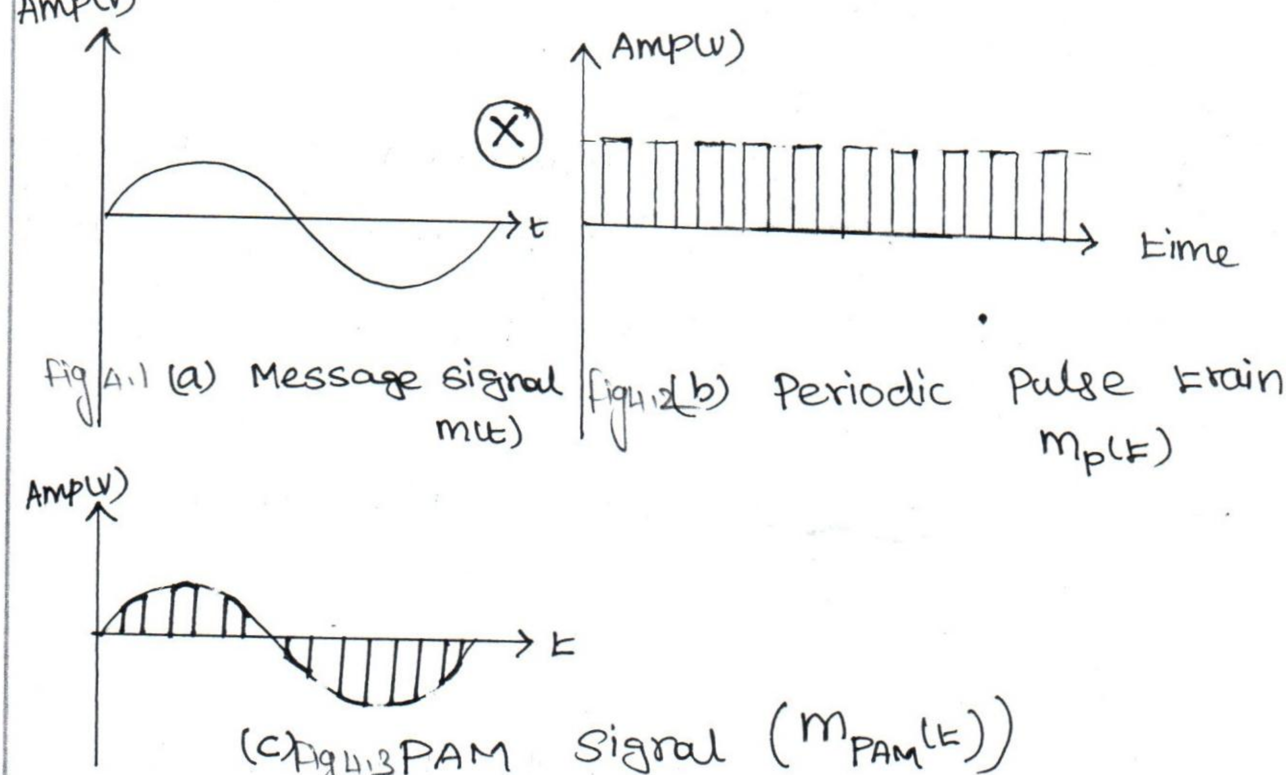


Fig: 4 Waveforms of PAM.

* Message signal i-e modulating signal m(t) is multiplied with pulse train $m_p(t)$ to produce PAM signal [$m_{PAM}(t)$].

1.3.2 Detection of PAM:-

Reconstruction Low pass filter act as a PAM detector. It detects the message & signal from PAM.

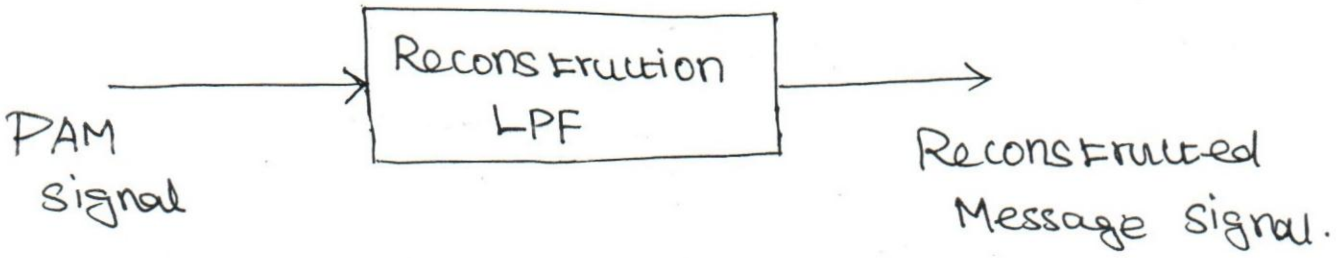


Fig: 5 PAM Detector.

1.3.3 Transmission Bandwidth of PAM (B_T):-

To transmit a PAM signal, bandwidth will be greater or equal to maximum frequency 'f_{max}' is required. (i-e)

$$\boxed{B_T \geq f_{max}} \rightarrow (1)$$

* The pulse duration 'T' of the PAM signal is very small compared to time period 'T_s' between two samples.

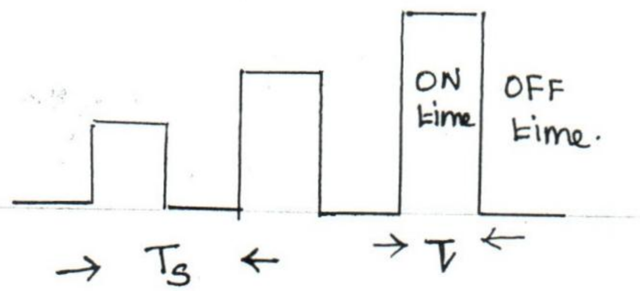


Fig: 6 PAM signal with same ON & OFF time.

* Maximum frequency of PAM signal can be written as

$$f = \frac{1}{T_{ON} + T_{OFF}} = \frac{1}{\tau + \tau} = \frac{1}{2\tau}$$

i-e $f_{max} = \frac{1}{2\tau} \longrightarrow (2)$

Sub (2) in (1)

$$B_T \geq \frac{1}{2\tau} \rightarrow (3)$$

W.K.T, τ is very small compared to 'T_s'

i-e $\tau << T_s$

$\tau << \frac{1}{2f_m} \rightarrow (4)$. since $T_s = 2f_m$

combine (3) & (4) we can write

$$\boxed{B_T \geq \frac{1}{2\tau} >> f_m} \longrightarrow (5)$$

∴ Transmission Bandwidth of PAM is $B_T >> f_m$

Where, $T_s \rightarrow$ Sampling time period

$f_m \rightarrow$ Maximum frequency of message signal (or) modulating signal.

Advantages:-

* PAM is the basics of other pulse modulation techniques like DM, ADM, PCM.

* Generation of PAM is very easy because a multiplier act as a PAM modulator.

Disadvantages:-

* A kind of distortion Aperture effect occurs.
* It requires huge Bandwidth.
* Poor Noise immunity.
* Interference of noise is maximum since the amplitude of PAM pulse is varied.
* Power requirement is not constant.

1.4 PULSE WIDTH MODULATION:- (PWM|PDM):

Definition:-

Width of the pulse carrier is Varied with respect to amplitude of message (or) modulating signal, where as amplitude and position of pulse remains constant.

* Pulse width modulation is also referred as Pulse duration modulation (PDM).

1.4.1 Generation of PWM:-

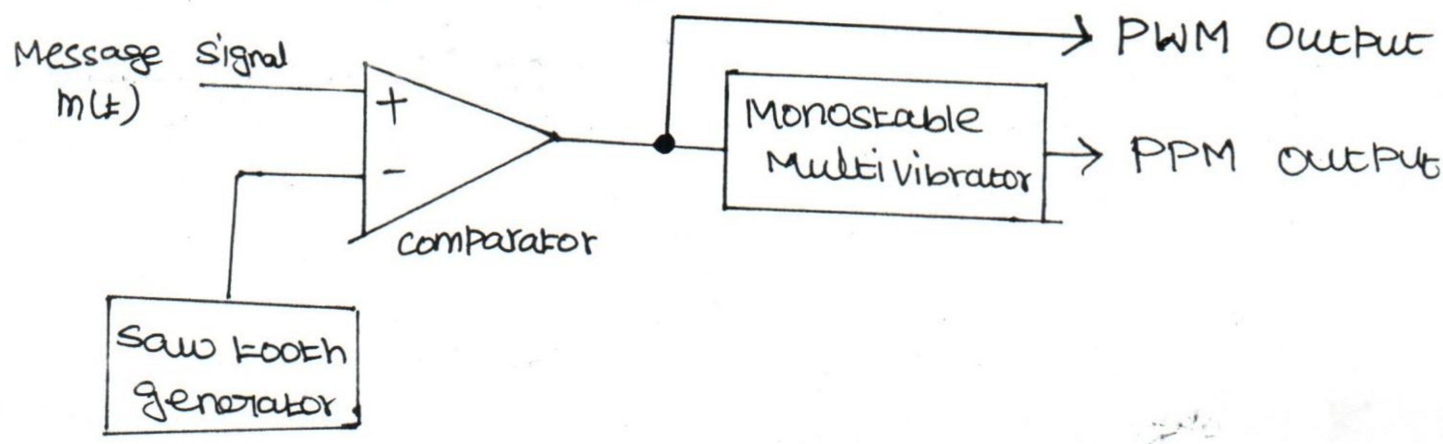


Fig: 7 Block diagram of PWM Generation

* Message signal i-e modulating signal m(t) is applied to non-inverting terminal of comparator.

* Sawtooth generator produces saw tooth signal. It act as a carrier & applied to inverting terminal of a comparator.

* Output of the comparator remain high as long as amplitude of m(t) is higher than that of the saw tooth signal.

* Output of comparator is PWM signal.

* This Block diagram is also used for PPM Generation.

1.4.12 Demodulation of PWM:-

* The combination of product detector and a Low pass filter act as PWM detector.

* Carrier and PWM signal are given to the product detector, and then a sequence of pulses having the width inversely proportional to the width of PWM pulses appears at the output.

* When the 'E_a' signal passes through the Low pass filter;

* At the output of LPF demodulated

signal is obtained.

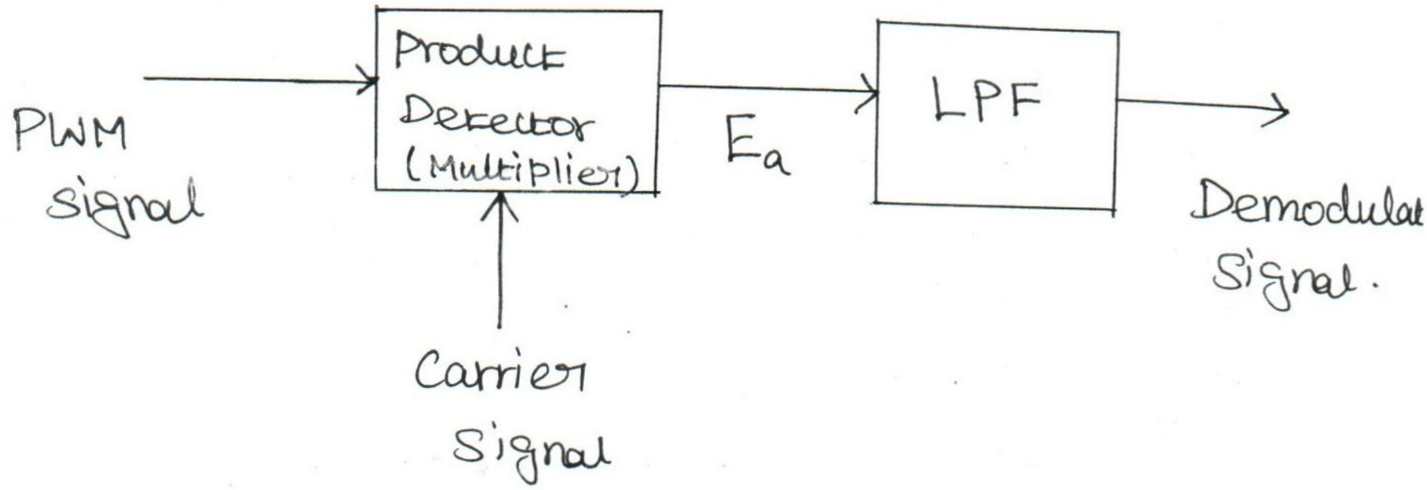


Fig: 8 PWM Detector.

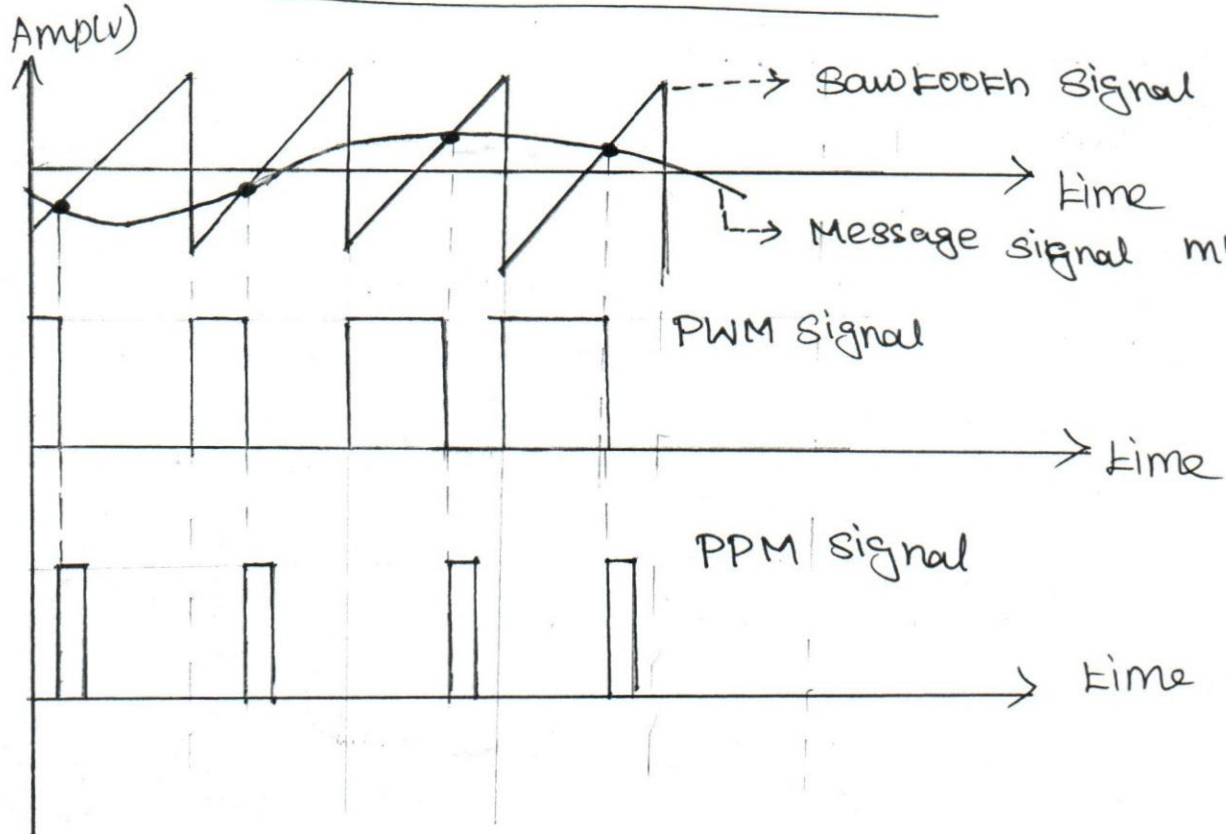


Fig: 9 PWM and PPM waveforms.

* Output of comparator is high when instantaneous value of m(t) is higher than that of sawtooth signal as shown in above figure.
* The leading edge of PWM signal occurs at fixed time period and trailing edge of the

output of comparator depends on the amplitude of message signal m(t).

* When the sawtooth voltage is greater the amplitude of m(t) at that instant, comparator output remains zero.

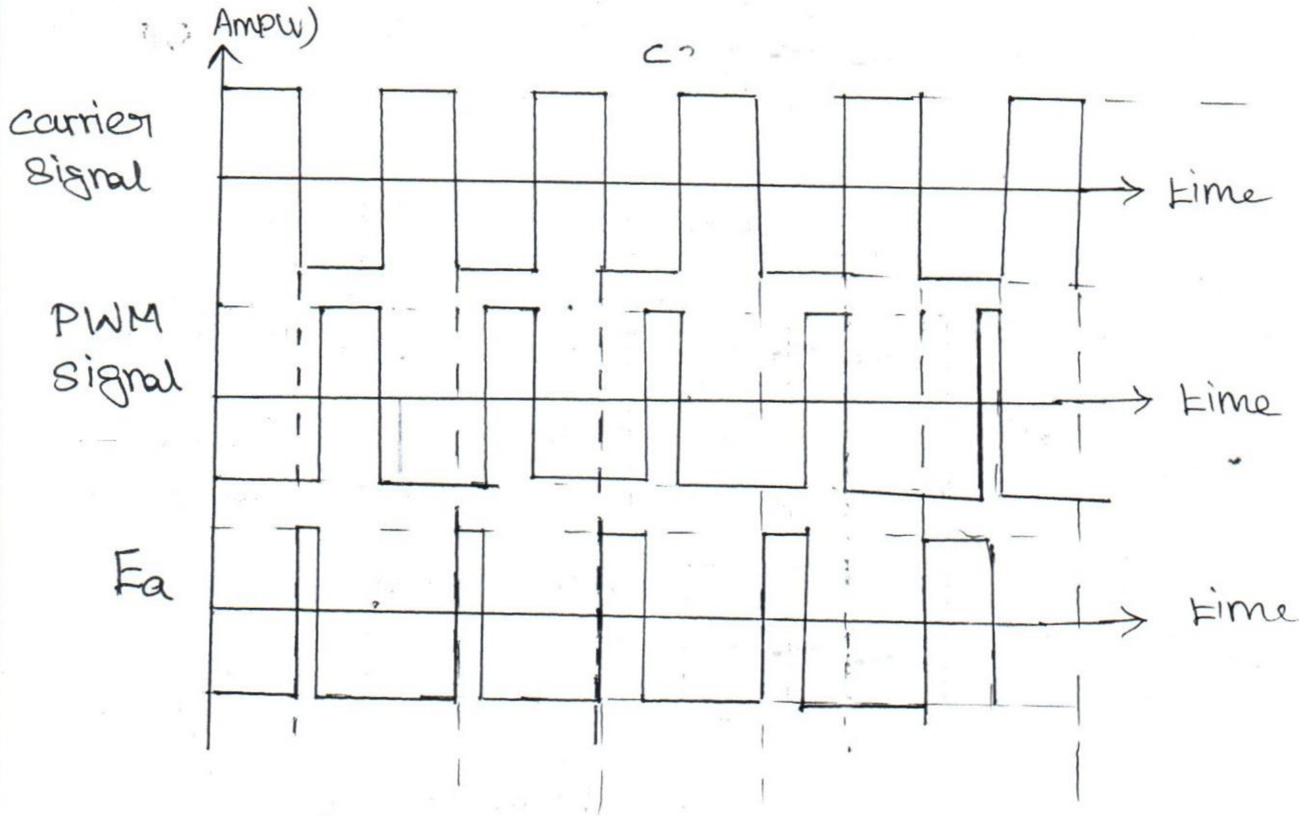


Fig:- PWM Detector output waveform

Advantages:-

* More noise immunity since amplitude kept constant.

* synchronization is not required.

Disadvantages:-

* Transmission Bandwidth for PWM signal is very high than PAM

* Power requirement is not constant. Variable pulse width causes variable power content.

(1.5) PULSE POSITION MODULATION:-

Definition:-

Position of the pulse carrier varies with respect to amplitude of the message (or) modulating signal. where as amplitude an width of pulse carrier is kept constant.

1.5.1 Generation:-

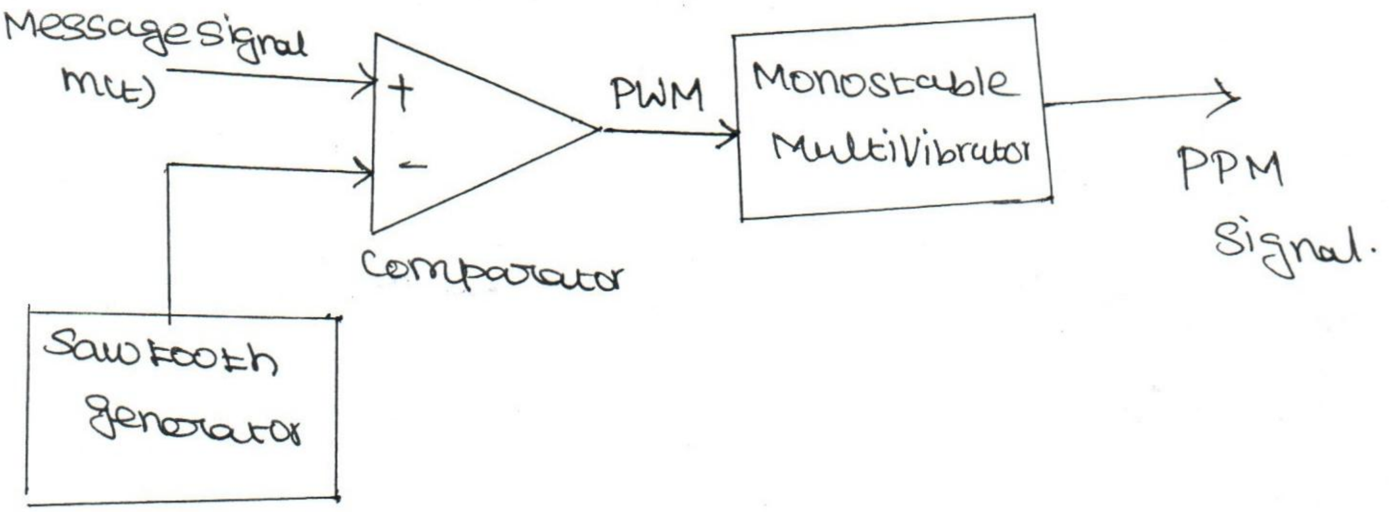


Fig: 10 Generation of PPM.

* Message signal m(t) and sawtooth signal from sawtooth generator is given to the comparator as input signal.

* Comparator produces PWM waveform. This PWM signal act as trigger input to the monostable Multivibrator.

* output of the monostable multivibrator remains zero till the trigger input is given.

* Multivibrator is triggered on the falling edge of PWM pulse and output of multivibrator

switches to positive level. This voltage remains high for fixed duration and it goes low level.

* output waveform is nothing but PPM waveform.

1.5.2 Demodulation of PPM:-

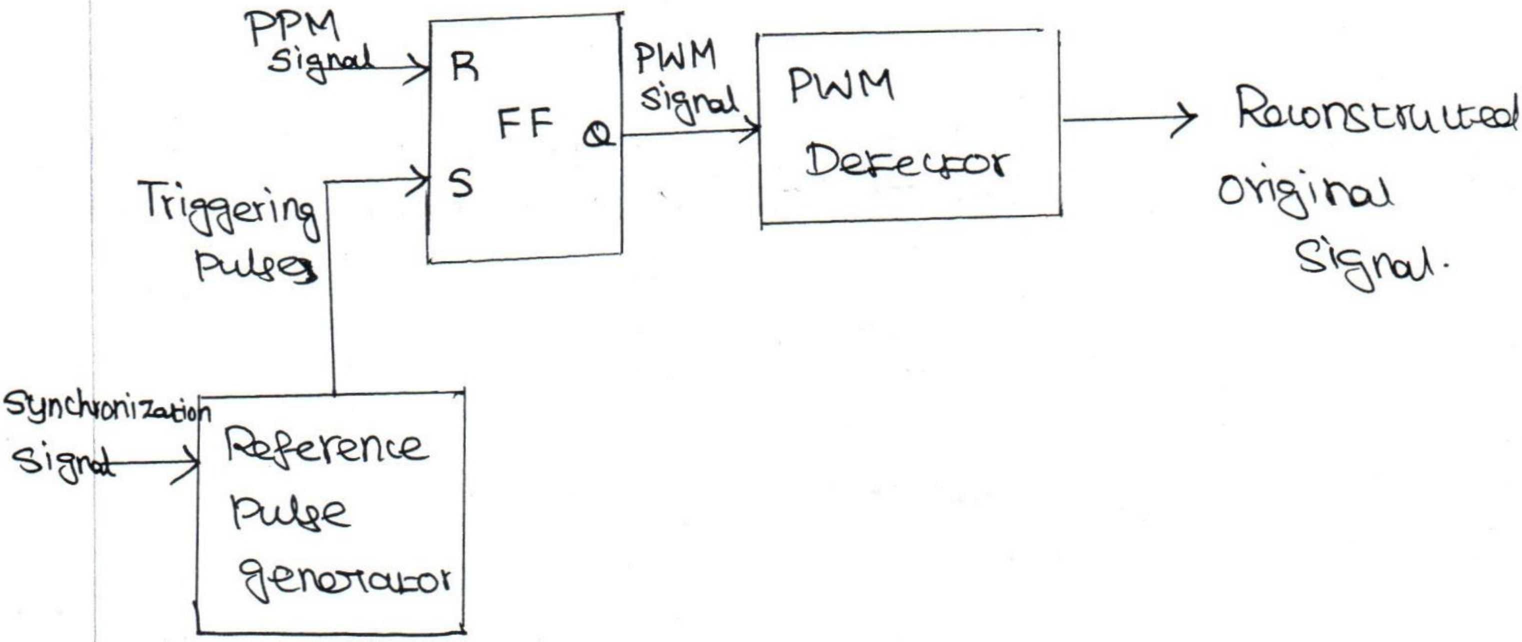


Fig:- 11 Block Diagram of PPM Demodulat

* PPM pulses are converted into PWM pulses by SR Flipflop.

* As shown in Fig the flipflop is set to ON when it receives reference pulse from Reference pulse generator.

* Flip Flop remains in ON state till the leading edge of PPM pulse and turns OFF after finite duration.

* PWM pulses are obtained at the output of FF.

* Then the PWM pulses are demodulated by PWM demodulator to get reconstructed messag signal.

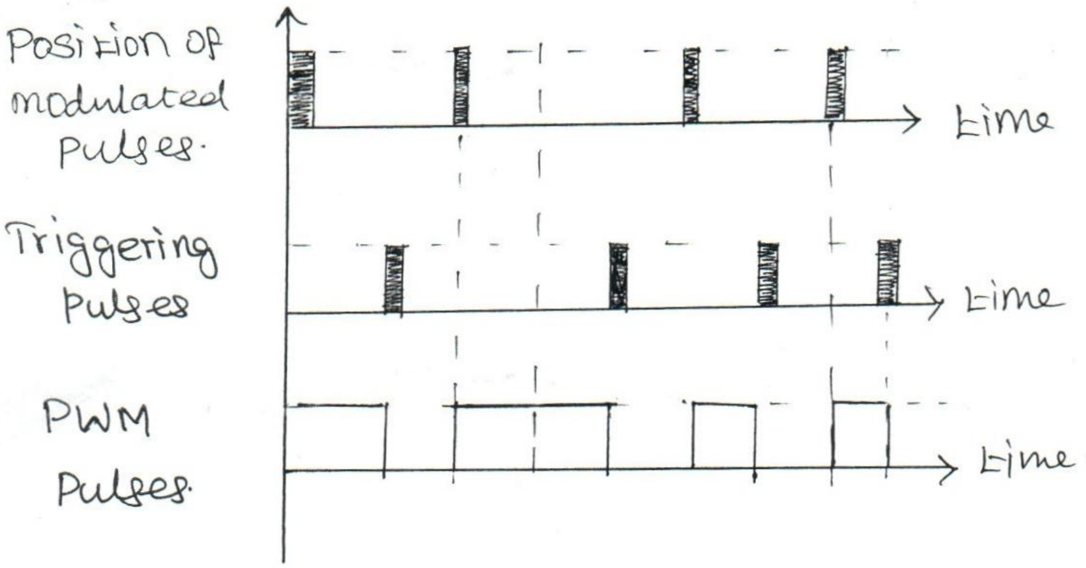


Fig: 12 PPM demodulated waveforms.

1.5.3 Transmission Bandwidth (B_T) of PPM & PWM:-

$$\boxed{B_T \geq \frac{1}{2T_r}}$$

Where,

$B_T \rightarrow$ Transmission Bandwidth.

$T_r \rightarrow$ Rise time.

Advantages:-

* The width of the pulse is constant hence transmitter power remains constant.

* Because of constant amplitude pulses noise interference is less.

Disadvantages:-

* It needs synchronization between transmitter and receiver.

1.6 QUANTIZATION:-

* Discretisation in amplitude is simply defined as Quantization.

Definition:-

Quantization is the process of sampled discrete time signal into discrete amplitude signal.

* Quantized signal is discrete both in time and amplitude.

Sampled discretised in time $\xrightarrow{m}$ Quantizer (q) $\xrightarrow{V}$ Sampled signal discretised in both time and amplitude

Fig: 13 Quantizer diagram

1.6.1 Quantization Error:- (q)

Quantization error is also called as Quantization Noise.

Definition:-

Difference between the instantaneous values of message signal and quantized signal is called as Quantization noise.

Where,

$$\boxed{q = V - m}$$

$m \rightarrow$ Instantaneous value of message signal

$V \longrightarrow$ Instantaneous value of quantized signal

1.6.2 Process of quantization:-

* Message signal m(t) is assumed to have a peak to peak swing of m_L to m_H volts.

* Voltage level from m_L to m_H is divided in to 'L' equal intervals each of size 'P'.

* Where 'P' is step size and it can be defined as

$$\boxed{P = \frac{m_H - m_L}{L}}$$

* At the center of these ranges, quantization levels $r_0, r_1 \dots r_7$ are placed. It's also called as decision thresholds.

* when m(t) is in the range Δ_1, then corresponding to any value of m(t), quantizer output will be equal to r_1.

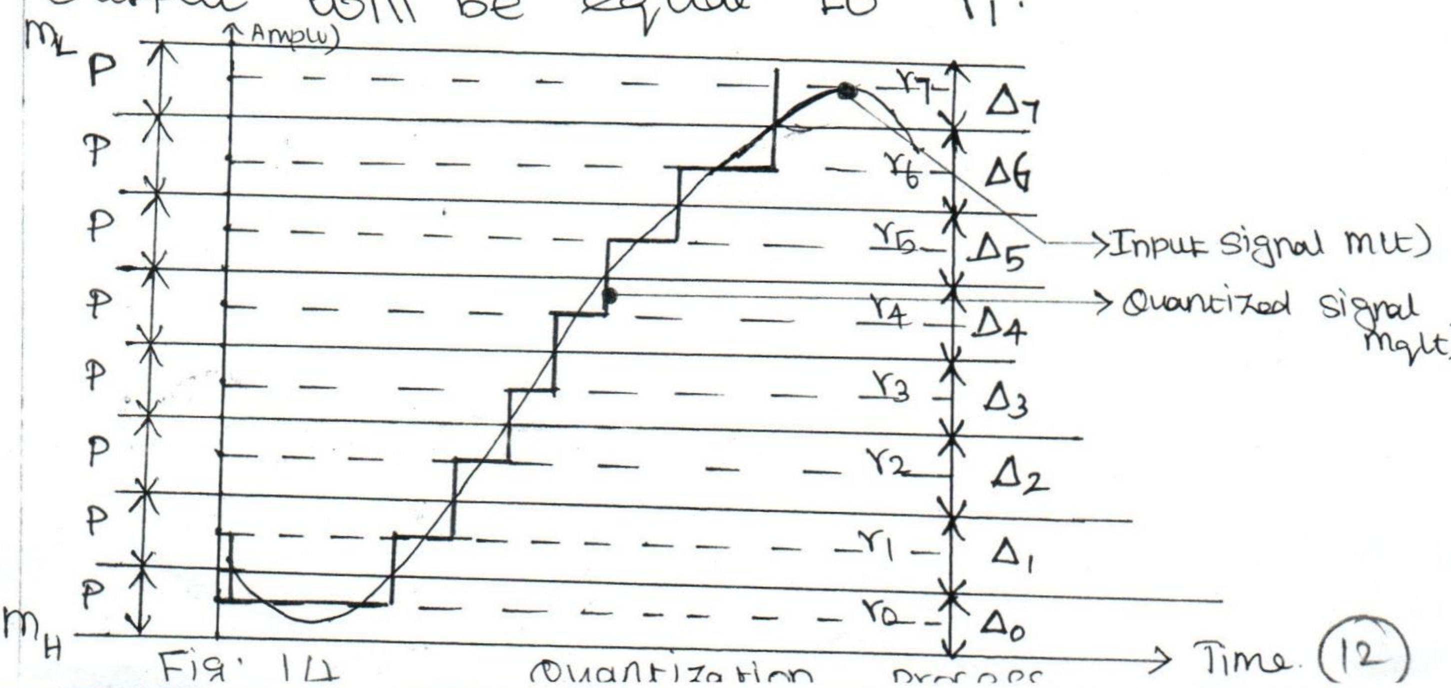


Fig: 14 Quantization process

Signal to quantization Noise Ratio $(SNR)_q$:-

For Sinusoidal signal, Input

$$(SNR)_{dB} = (1.8 + 6R)\,dB.$$

For Non Sinusoidal Input

$$(SNR)_{dB} = (4.8 + 6R)\,dB.$$

where, $R \rightarrow$ Word size i-e Number of bits per sample.

1.6.3 Types of Quantization:-

- Uniform Quantization
  - Midtread Quantizer
  - Midrise Quantizer.
  - Biased.
- Non-Uniform Quantizati

Uniform Quantization:-

Step size between two quantization levels remains constant over the complete amplitude range

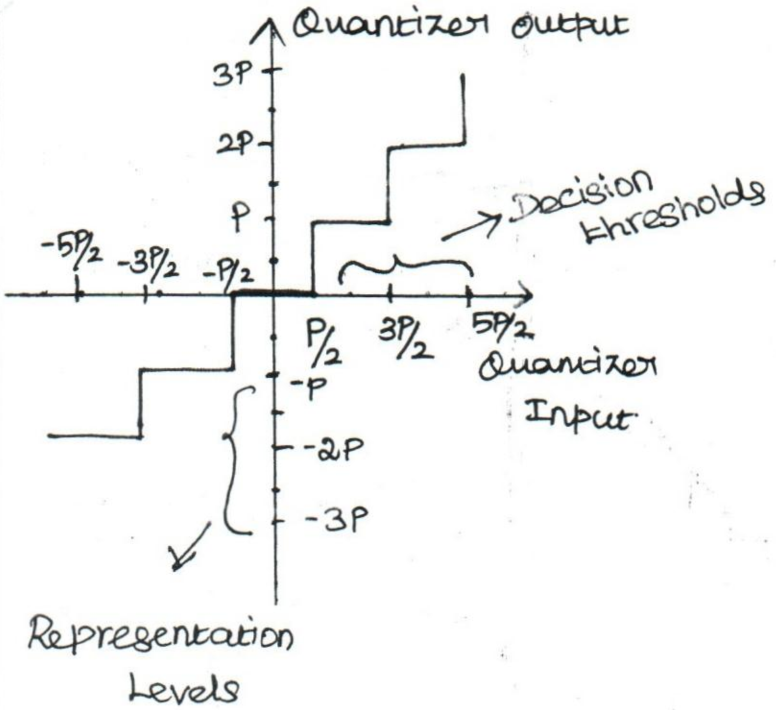
Midtread Quantizer:-

* Origin lies in the middle of a tread of the staircase.
* Quantizer output is zero when input is zero.

Two fold effects of Quantization Process:-

* Peak to peak range of the input is divided in to a finite set of decision levels (or) decision thresholds. and these levels are aligned with the "tread" of the staircase.

* Output is assigned a discrete value selected from a finite set of representatio levels and these levels are aligned with "risers" of the staircase.



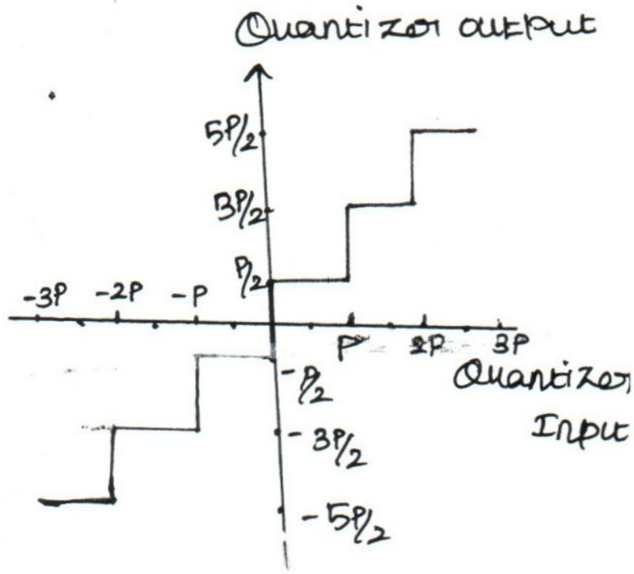


Fig: 15.1 (a) Midtread type

Fig: 15.2 (b) Midriser type

Midriser Type:-

* origin Lies in the middle of riser of the staircase

* Quantizer output is $\pm P/2$ when input is zero, where P is step size.

Biased Quantizer:-

Quantizer output is zero when the input is between '0' to 'P'.

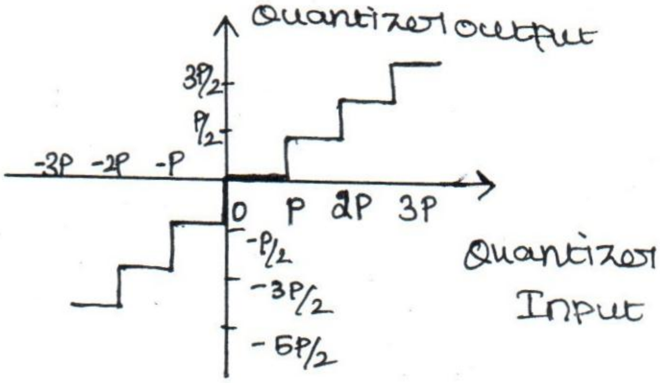


Fig:- Biased Quantizer.

Non-Uniform Quantization:-

* Step size between quantization levels (or) Representation levels are not uniform through out the complete amplitude range.

* In non uniform Quantization step size of the quantizer is varies with input signa

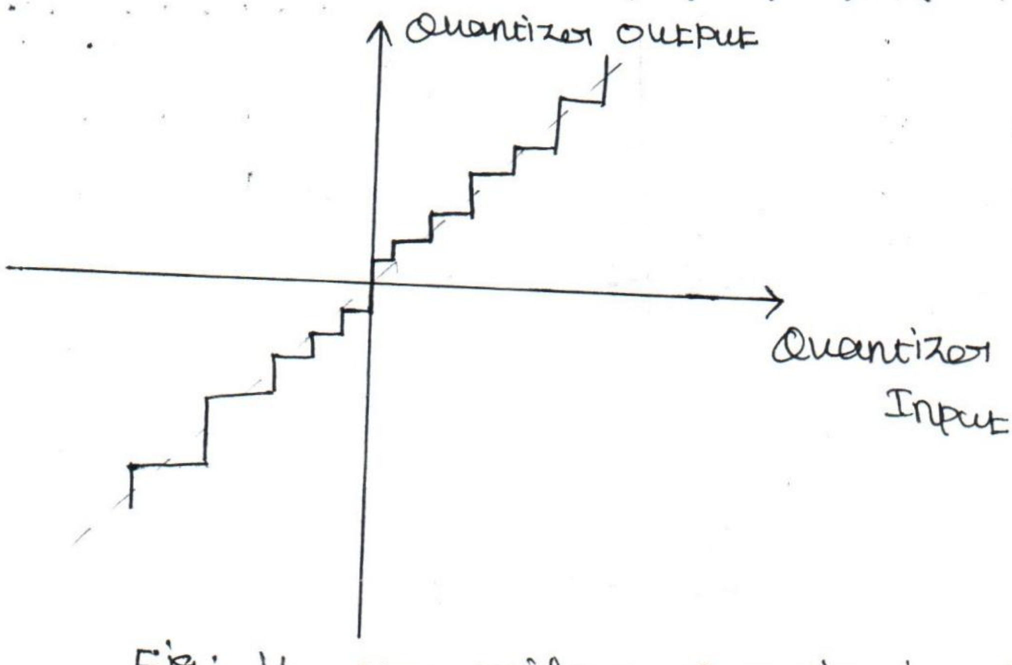


Fig: 16 Non uniform quantization transfer characteristics.

Q-6

* Quantizer with varying step size is difficult to implement

* A more practical approach is to pre distort the signal by a logarithmic compression characteristics. and then put it to an uniform quantizer. This compressed and quantized signal is transmitted through the channel and can be undistorted at the receiver by the same algorithm. This process is called as companding.

1.6.4 Companding:-

Definition:

The processing pair of compression and expansion is collectively called as companding.

* companding is required to improve signal to quantization noise ratio of weak signals.

Types:

M-Law Companding

A-Law Companding

μ-Law companding:-

μ-Law companding used in Digital communication System of North america and japan.

* In this companding, compressor characteristics is continuous for smaller values of input level.

* compressor Characteristics is linear for larger values of input levels. and this Characteristics are logarithmic in nature.

compressor characteristics is expressed as,

$$|V| = \frac{\log_e(1 + \mu|m|)}{\log_e(1+\mu)}$$

Where,

$V \rightarrow$ Output of the compressor

$m \rightarrow$ Input of the compresor

$\mu \rightarrow$ compression factor $= 255$

Reciprocal slope of compression curve can be obtained by

$$\frac{d|m|}{d|V|} = \frac{\log_e(1+\mu)}{\mu}(1 + \mu|m|).$$

A - Law companding:-

Used in european digital communication systems and rest of the world's national systems.

Compressor characteristics is expressed as

$$|v| = \begin{cases} \dfrac{A|m|}{1+\log_e A} ; & 0 \le |m| \le \dfrac{1}{A} \\ \dfrac{1+\log_e (A|m|)}{1+\log_e A} ; & \dfrac{1}{A} \le |m| \le 1 \end{cases}$$

Reciprocal curve of compression curve is

Diff $|v|$ w.r.t $|m|$,

$$\frac{d|m|}{d|v|} = \begin{cases} \dfrac{1+\log A}{A} ; & 0 \le |m| \le \dfrac{1}{A} \\ (1+\log A)|m| ; & \dfrac{1}{A} \le |m| \le 1 \end{cases}$$

A → Compression factor 87.7

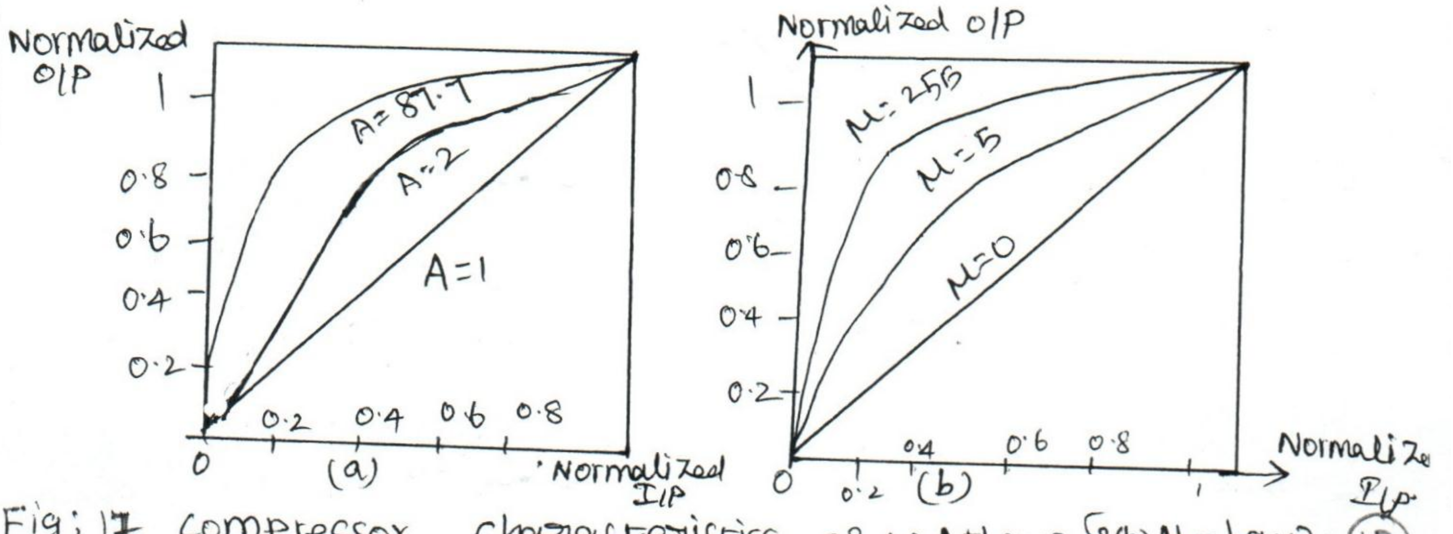


Fig: 17 Compressor characteristics of (a) A-Law & (b) μ - Law.

(2) ENCODING:-

~~Definition~~ (2.1) INTRODUCTION:

* Encoding is a process, it convert a Quantized discrete signal to a digital signal. (binary bits).

* In a binary code each symbol may be either of two different values denoted as '0' & '1'.

* The input message does not depend on the Characteristis of pulse (amplitude, width, & position). It depends on the presen and absence of pulse.

* The Presence or absence of a pulse is a Symbol

* Encoding process increse the immunity of the signal. It directly affects the two basic parameters of digital signal

(i) Transmission Bandwidth

(ii) Bit error rate.

* Analog Encoders are used for encoding speech signal and images. They are Classified in to three types.

Q-10.

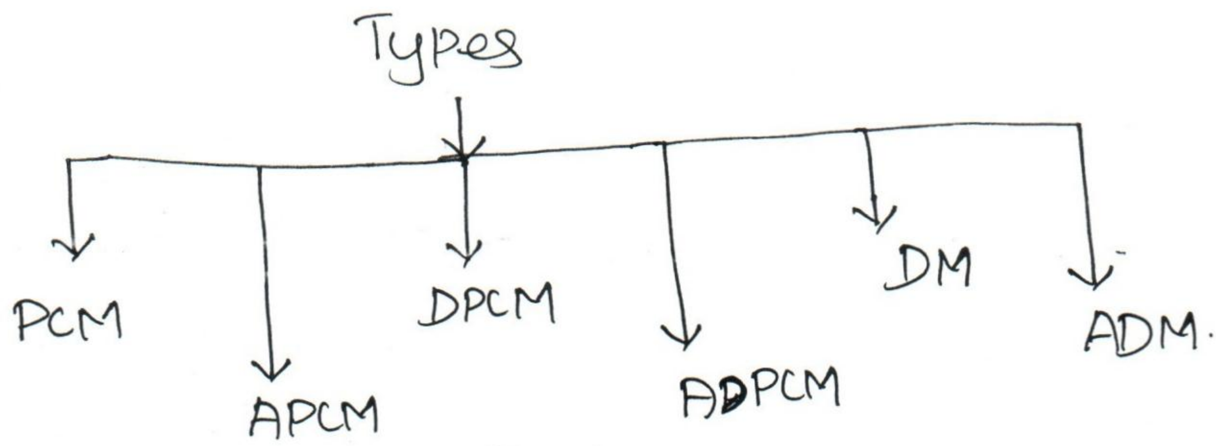
Types:-

1. Temporal waveform Encoding.
2. Spectral Waveform Encoding.
3. Model Based Encoding.

Temporal Waveform Encoding:-

Encoder captures the temporal characteristics of source waveform (i-e) Speech (or) music signal.

* Time domain waveform is encoded and bit rate is high compared to Signal Bandwidth.



PCM → Pulse code Modulation

APCM → Adaptive Pulse code Modulation

DPCM → Differential Pulse code Modulation

ADPCM → Adaptive Differential Pulse Code Modulation

DM → Delta Modulation

ADM → Adaptive Delta Modulation.

PULSE CODE MODULATION (PCM):

PCM is essentially an analog to digital conversion process, where the information contain in the instantaneous sample of analog signal ar represented by digital codes and are transmitte as a serial bit stream.

In PCM the message signal is represented in the form of coded pulses by representing the signal in discrete time and amplitude.

Definition:-

In PCM, PAM signal is quantized and converted to a digital code this signal Called as pulse code modulated signal this process is called as pulse code modulation.

Basic operations involved in PCM:-

The basic operations performed in transmitte is Sampling, Quantizing, Encoding.

The basic operations in the receiver is Regeneration, Decoding & Filtering.

2.2.1 Basic Elements of PCM:-

PCM consists three major blocks

1. Transmitter
2. Transmission Path
3. Receiver.

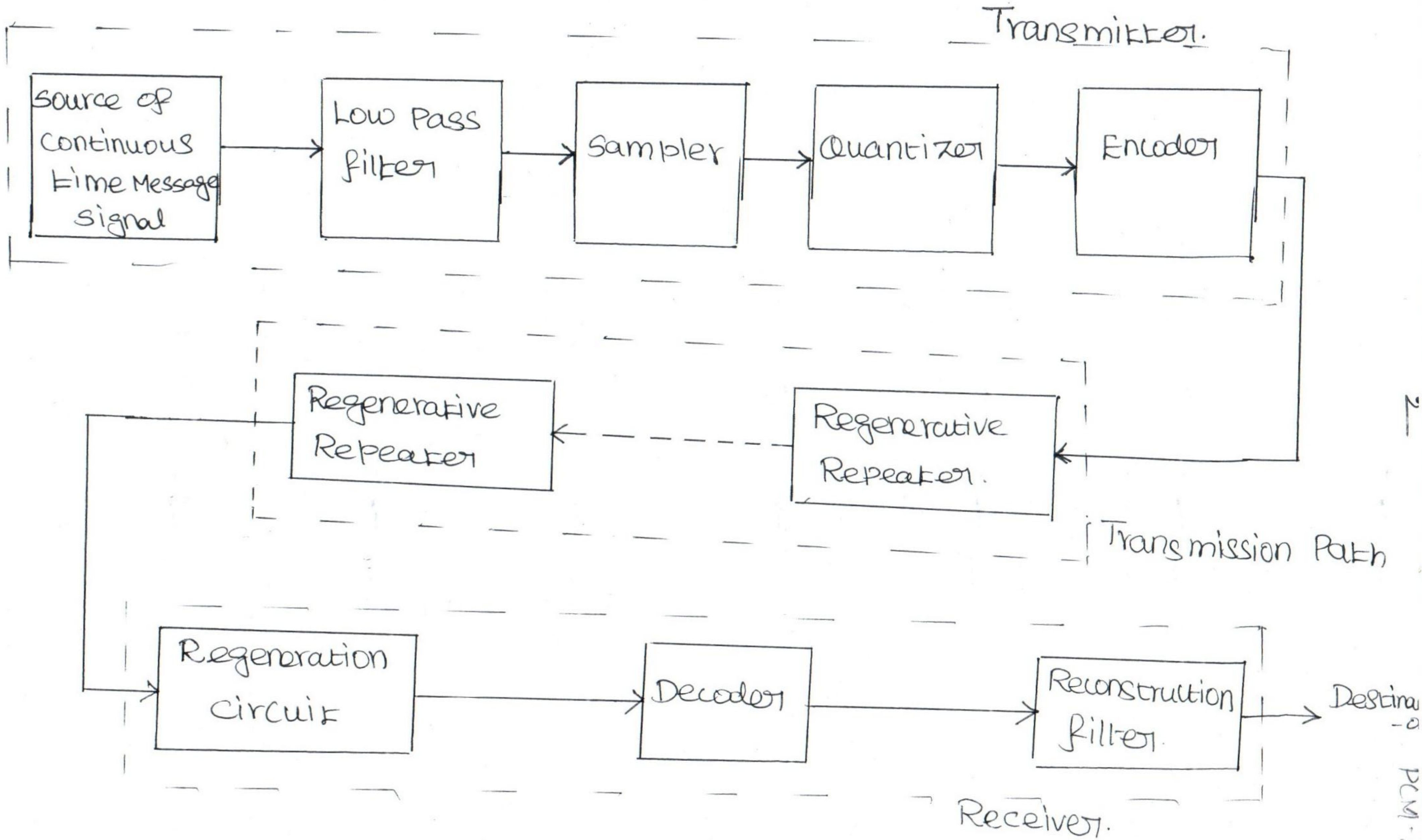


Fig: 1 Basic Elements of PCM.

Transmitter:-

* Source of continuous time message signal produces continuously varying message signal with time.
* Low pass filter act as anti aliasing filter attenuate high frequency components.

SAMPLING:- * Emitter follower circuit is used as sampler. Filtered signal is sampled with the help of rectangular pulses.

Quantization:-

Sampled signal is quantized so that resultant signal is discrete in both time and amplitude.

Types

Uniform Quantization   Non Uniform Quantization

Uniform Quantization:-

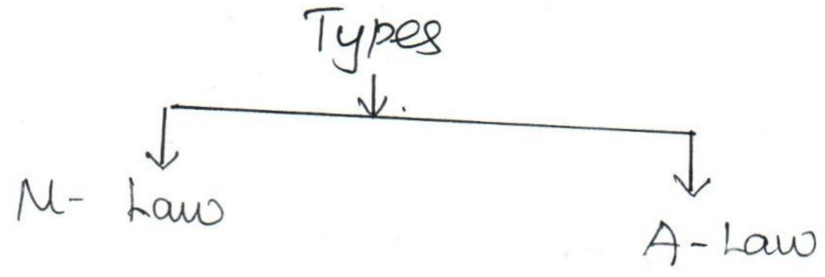
The step size between two quantization level remains constant over the complete amplitude range.

Types:-

Mid-tread   Mid-rise

Non-Uniform Quantization:-

Step size between two quantization levels are non-uniform throughout the complete amplitude range.

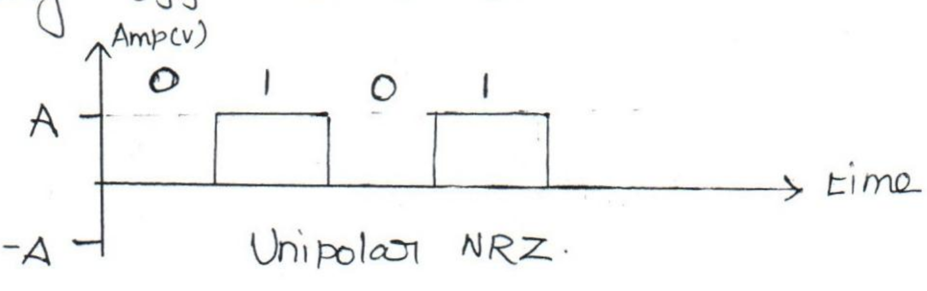


Encoding:-

* Discrete samples are converted to coded pulses
* Presence (or) absence of a pulse is a symbol.
* Some well-known line codes that can be used for the electrical representation of a binary data stream, are: (a) Unipolar NRZ signaling (b) Polar NRZ signaling (c) Unipolar RZ signaling (d) Bipolar RZ signaling (e) Split-phase (or) Manchester code.

Unipolar NRZ:

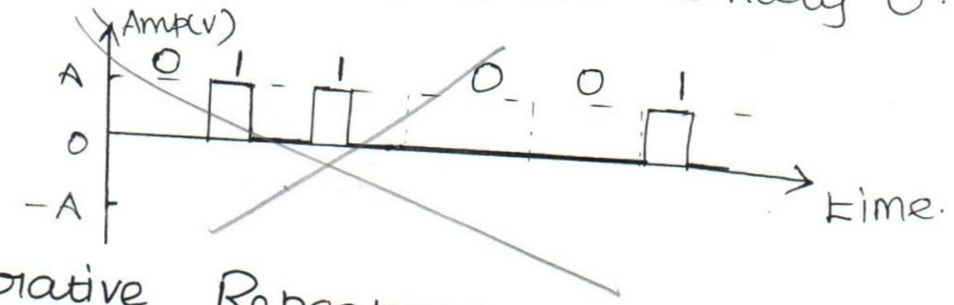
* This line code is also known as on-off signaling.
* Transmitting a pulse of amplitude +A for the duration of the symbol represents binary '1'. Switching off the pulse represents binary '0'.



Unipolar NRZ.

Polar NRZ :-

* Transmitting a pulse of amplitude '+A' represents binary '1'. Transmitting a pulse of amplitude '-A' represents binary '0'.



Regenerative Repeater:-

A regenerative repeater consists of an equalizer, a timing circuit and a decision making device. Regenerative repeaters are used to reconstruct the PCM signal.

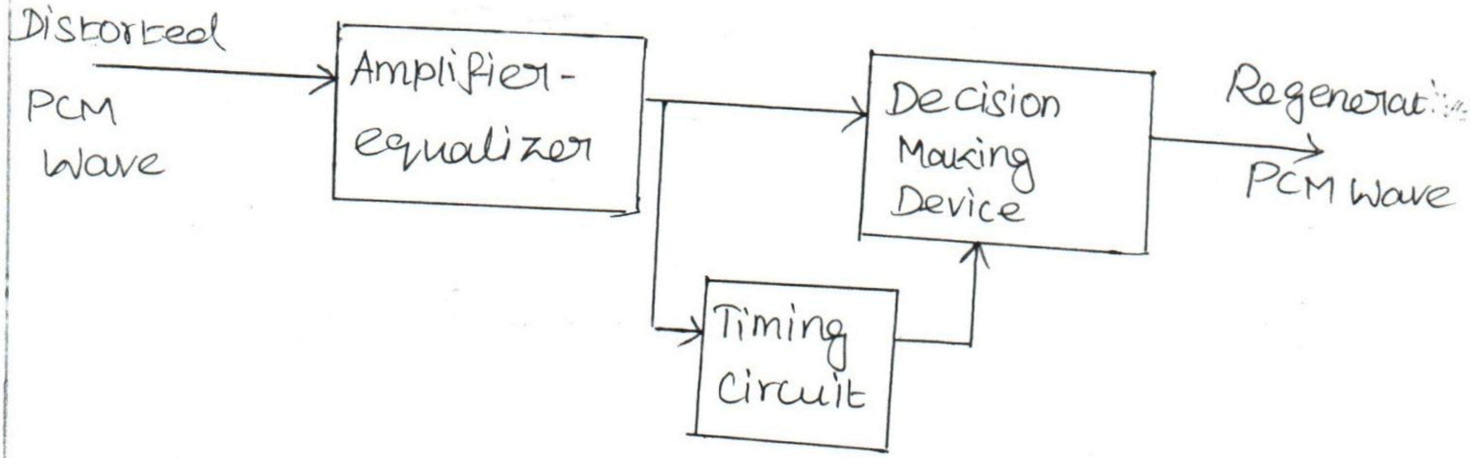


Fig: 2 Regenerative Repeater.

* Equalizer is used to shapes the received pulses.

* Timing circuit provides a periodic pulse train for sampling

* The function of the decision-making device is to detect the different pulses based on

some threshold information. Each sample is compared with a threshold value. If the threshold is exceeded, '1' is transmitted otherwi '0' is transmitted thereby removing distortion anc noise.

Decoding:-

* After regeneration i.e. reshape and clean up, the clean pulses are regrouped in to code words and decoded in to a quantized PAM signal.

* The decoding process generates a pulse, the amplitude of which is the linear sum of all pulses in the code word.

Filtering:-

The message signal is recovered by passing the decoder output to Low pass reconstruction filter whose cut-off frequency is equal to message bandwidth 'W'.

PCM BANDWIDTH:-

Transmission Bandwidth of PCM for

Sinc pulse $\boxed{B_T \geq \frac{R_s}{2}} \longrightarrow$ 1

Where, $R_s \rightarrow$ Symbol rate $= lf_s \longrightarrow$ 2

$l \rightarrow$ PCM word size

$f_s \rightarrow$ Sampling frequency $= 2B \rightarrow 3$

Substitute 3 in 2

$$R_s = l2B \rightarrow 4$$

Substitute 4 in 1

$$\therefore B_T \geq \frac{l2B}{2}$$

Transmission Bandwidth of PCM (Sinc pulses) $\boxed{B_T \geq lB} \rightarrow$

For Rectangular pulses $B_T = R_s \rightarrow 6$

Substitute 4 in 6

$$\boxed{B_T = 2lB} \rightarrow 7$$

PCM WORD SIZE:-(l)

Number of bits per sample.

$$\boxed{l \geq \log_2\left(\frac{1}{2p}\right) \text{ bits.}} \rightarrow 8$$

where,

$p \rightarrow$ Tolerance in quantization Error.

(or)

Fraction of the peak to peak analog voltage V_{pp}

SNR OF A PCM SYSTEM:

Signal to Noise ratio (SNR) of a PCM system when only quantization noise is present:

The quantized signal power can be approximated to the second moment of the clean signal m(t)

$$S_q = \overline{m^2(t)} \rightarrow 9$$

Quantization noise power $(N_q) = \overline{e_q^2} = \frac{m_p^2}{3L^2} \rightarrow 10$

where, $L \rightarrow$ Number of levels of the quantizer

$m_p^2 \rightarrow$ Peak quantization level of uniform quantizer.

$$\frac{(9)}{(10)} = \boxed{(SNR)_q = \frac{S_q}{N_q} = \frac{3L^2\,\overline{m^2(t)}}{m_p^2}} \rightarrow 11$$

For Sinusoidal Input $(SNR)_q = (1.8 + 6N)\,dB$

For Non sinusoidal Input $(SNR)_q = (4.8 + 6N)\,dB$

Bandwidth Expansion factor (b): $\rightarrow \boxed{b = \frac{B_T}{B}} \rightarrow 12$

Where, $B_T \rightarrow$ Transmission Bandwidth.

$B \rightarrow$ Message Signal Bandwidth.
(or)
Maximum frequency of Message signal.

2.2.2 NOISE CONSIDERATIONS IN PCM SYSTEMS:

Noises present in PCM system are.

i) Aliasing Noise ii) Quantization Noise
iii) Channel Noise (iv) Intersymbol Interference

Aliasing Noise:-

* Aliasing Noise occures due to improper Sampling.

Quantization Noise:-

* Completely Known from the Specification of quantizer. Quantization noise is introduced in the transmitter and carried all the way along the receiver output.

Channel Noise:-

* Channel Noise intruduces bit error in the transmission path.

* To minimize bit error rate, Additive white gaussian channel is used.

Intersymbol Interference:-

Communication Channel is band limited hence it always spreads a pulse waveform passing through it.

Case:- 1 Channel Bandwidth > Pulse Bandwidth

* Spreading of pulse is small in this case.

Case:- 2 Channel Bandwidth = Pulse Bandwidth

* Spreading will exceed a symbol duration and cause signal pulse to overlap. This overlapping is known as ISI (Inter Symbol Interference).

ADVANTAGES:-

* Multiplexing of various PCM signals is easily possible.

* PCM systems have high noise immunity compared to other digital techniqus.

* In long distance digital telephone systems, PCM technique uses repeaters, which regenerates a clean PCM waveform at the output by removing the distortion and noise.

* Due to the digital nature of PCM signal, it can be easily stored.

* It is possible to use various coding techniques, so that only the desired person can decode the received signal.

* PCM signal can be stored easily.

Disadvantages:-

* PCM requires large bandwidth.
* Due to encoding & decoding the PCM systems are complex.

Applications:-

* In long distance digital telephone systems.
* In space communication only PCM systems can be used.

Modifications of PCM:

* It can be modified to delta modulation.
* With the help of data compression, redundancy can be removed. This is used in DPCM.

2.3 DELTA MODULATION :-

Delta modulation (DM) is a DPCM scheme in which the difference signal $\Delta(t)$ is encoded into just a single bit. The single bit; providing for just two possibilities, is used to increse or decrease the estimate $\hat{m}(t)$.

Definition:

One bit (or) Two level version of DPCM. Delta modulation provides a staircase approximation to the oversampled version of message signal. Difference between Input and approximation is quantized into two levels namely $\pm\Delta(t)$.

2.3.1 DM TRANSMITTER:-

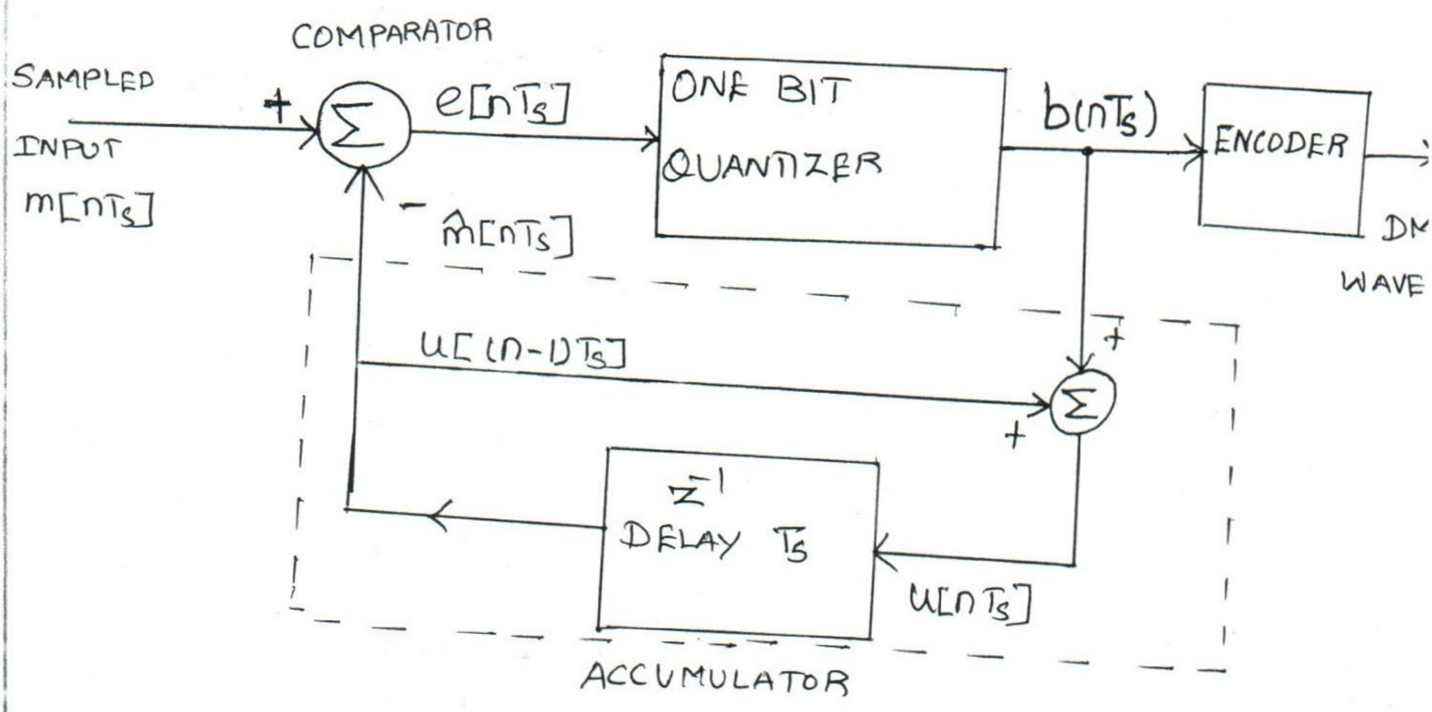


Fig : 3 DM TRANSMITTER

Transmitter consists:

1. One bit Quantizer
2. Comparator
3. one bit delay Unit
4. Accumulator.

One bit Quantizer:

Used to find the difference between Input and staircase approximation is quantized in to two levels that is $\pm\Delta$

Case: 1 If the approximation falls below the Input signal, quantizer Input is $+\Delta$ & quantizer output is '1' where step size is Incresed.

Case: 2 If the approximation lies above the Input signal, quantizer Input is $-\Delta$ & quantizer output is '0' where step size is decresed.

Accumulator:

* Generates staircase approximation delayed by one sample period T_s & It is also add... the quantized output as a stream.
* If the output of quantizer is '1' one step '$+\Delta$' is added with previous delayed staircase approximation.

* If the output of quantizer is '0', one step '$-\Delta$' is subtracted with previous delayed staircase approximation signal.

* $m(t)$ is the message signal, $m_q(t)$ is staircase approximation.

$$m[t] = m[nT_s] \quad ; n = 0, \pm 1, \pm 2 \dots$$

Where, $T_s \rightarrow$ sampling period

$m[nT_s] \rightarrow$ Sample of the signal $m[t]$ taken at time $t = nT_s$.

* Error signal is computed with the difference between last approximated sample and sampled value of $m(t)$,

$$\boxed{e[nT_s] = m[nT_s] - \hat{m}[nT_s]} \longrightarrow (1)$$

where,

$e[nT_s] \rightarrow$ Error at present sample

$\hat{m}[nT_s] \rightarrow$ Last approximated sample of the staircase waveform.

$$\boxed{\hat{m}[nT_s] = u[(n-1)T_s]} \longrightarrow (2)$$

where,

$u[(n-1)T_s] \longrightarrow$ Staircase approximation of last sample.

Sub (2) in (1)

$$\therefore \boxed{e[nT_s] = m[nT_s] - u[(n-1)T_s]} \rightarrow (3)$$

⇒ Input of delay unit $u[nT_s]$,

$$\boxed{u[nT_s] = \hat{m}[nT_s] + b[nT_s]} \rightarrow (4)$$

Sub (2) in (4)

$$\boxed{u[nT_s] = u[(n-1)T_s] + b[nT_s]} \rightarrow (5)$$

Where,

$b[nT_s] \rightarrow$ Quantized version of $e[nT_s]$.

$$\boxed{b[nT_s] = \Delta \operatorname{sgn} [e[nT_s])} \rightarrow (6)$$

⇒ Output of the one bit quantizer depends on the sign of $e[nT_s]$,

(i) If step size is $+\Delta$, binary '1' is transmitted. i-e,

$$b[nT_s] = +\Delta; \quad \text{if } m[nT_s] \geq \hat{m}[nT_s]$$

(ii) If step size is $-\Delta$, binary '0' is transmitted i-e,

$$b[nT_s] = -\Delta; \quad \text{if } m[nT_s] < \hat{m}[nT_s]$$

Hence (5) can also be written as

$$\boxed{u[nT_s] = u[(n-1)T_s] + [\pm\Delta]} \rightarrow (7)$$

2.3.2 DM RECEIVER:

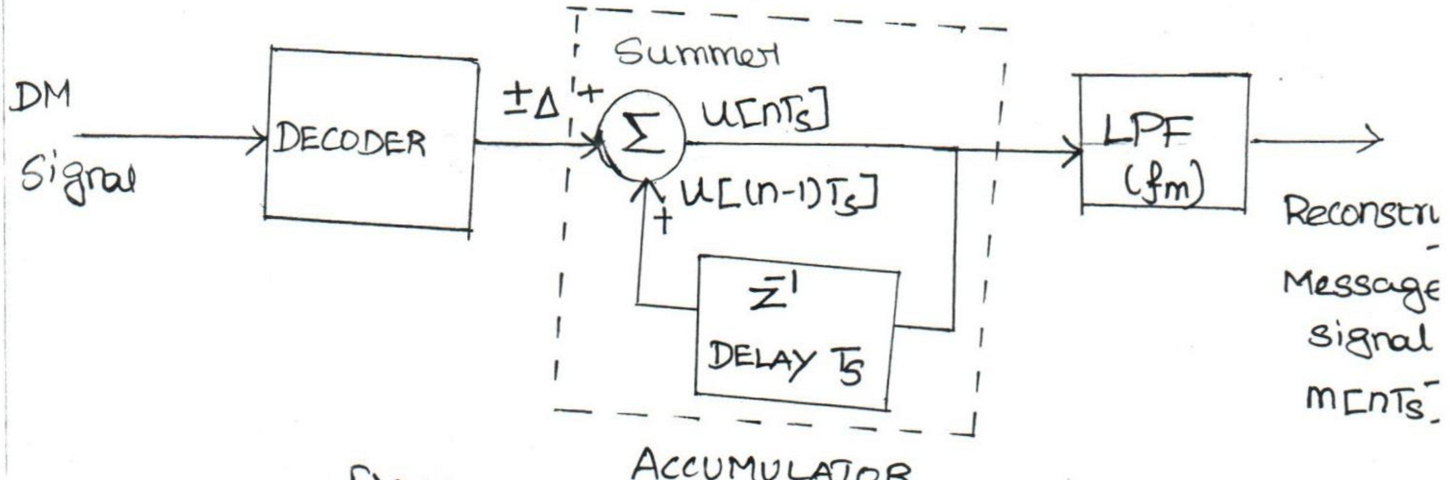


fig: 4 DM RECEIVER.

⇒ Receiver consists Accumulator, Decoder, LPF. LPF is also called as Reconstruction filter.

⇒ Accumulator is used to generate approximat staircase with one period delay (T_s).

⇒ Summer adds quantizer output with ($\pm\Delta$) the previous sample approximation. It gives present approximation,

$$u[nT_s] = u[(n-1)T_s] + [\pm\Delta] \quad \text{—} (8)$$

$$u[nT_s] = u[(n-1)T_s] + b[nT_s] \quad \text{—} (9)$$

Then, the accumulator,

(i) If the input is binary '1', then '$+\Delta$' step i add with the previous output.

(ii) If the input is binary '0', then '$-\Delta$' ste is add with the previous output.

⇒ LPF is used to reconstruct the original message signal $m[nT_s]$.

Waveform of Deltamodulation:

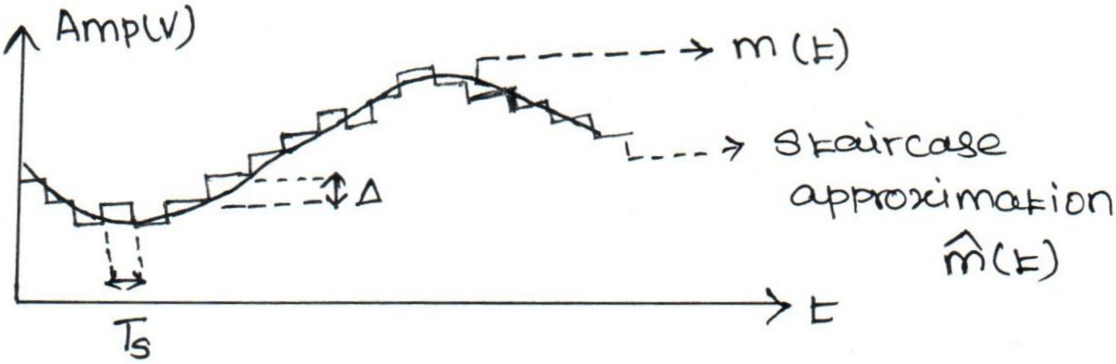


Fig : 5 Waveform of Delta modulation

Binary Sequence at
Modulator output 0 0 1 0 1 1 1 1 1 1 0 1 1 1 0 0 0 0

2.3.3 QUANTIZATION ERRORS IN DELTA MODULATION :-
(Or)
QUANTIZATION NOISE IN DELTA MODULATION :-

DM is subject to two types of quantization error.

1. Slope overload distortion
2. Granular noise.

Slope overload Distortion:

Definition: The excessive disparity between m(t) & $\hat{m}(t)$ is described as a slope overload error or distortion. & occurs whenever m(t) has a slope larger than the slope $\hat{m}(t)$.

=> If the step size of staircase approximation is too small, It cannot follow a steep

Segment of Input signal hence slope of message signal (m(t)) is higher than slope of staircase approximation $\hat{m}(t)$.

Mathematically speaking, slope overload can be avoided If

$$\boxed{\frac{\Delta}{T_s} > \left|\frac{dm(t)}{dt}\right|_{max}}$$

Granular Noise:

⇒ If the stepsize (Δ) of staircase approximation is too large it cannot follow flat segment of message signal (m(t)).

⇒ When the input signal is almost flat staircase approximation keeps oscillating between ±Δ.

⇒ This kind of error signal is called granular noise.

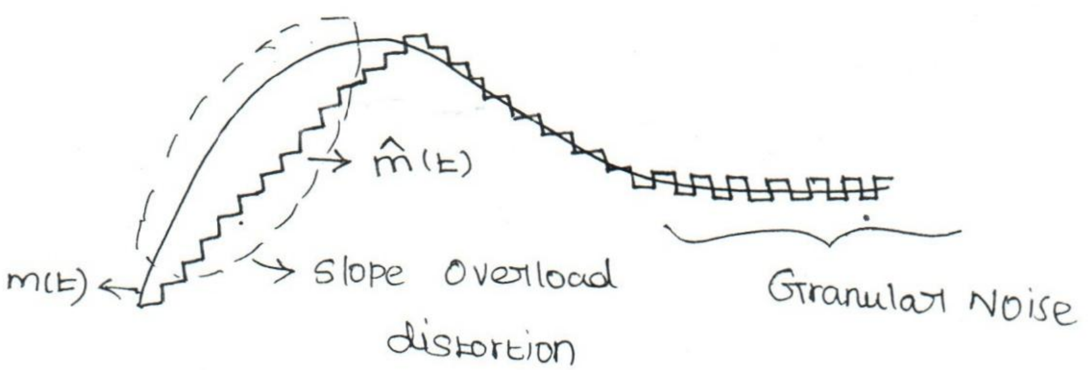


Fig: 6 Slope overload distortion and Granular Noise in DM.

⇒ The small step sizes are required for reducin granular noise whereas large step sizes are preferred for minimizing the slope overload error

⇒ If the step size is variable then the slope overload distortion and granular noise bo can be controlled. A system with a variable stepsize is known as the adaptive delta modu -a (ADM).

2.3.4 SNR Calculation of DM:

In DM, the amplitude of modulator outpu changes by $\pm\Delta$ only. So the maximum amplit is $A_m = \frac{\Delta f_s}{2\pi f_m}$, when there is no slope overlo distortion.

where, $A_m \rightarrow$ Peak amplitude of the message signal.

$\Delta \rightarrow$ step size

$f_m \rightarrow$ Message signal frequency

$T_s \rightarrow$ sampling period.

$$A_m = \frac{\Delta f_s}{2\pi f_m} \quad \text{——— (10)}$$

$$SNR = \frac{S_p}{N_p} \quad \text{——— (11)}$$

Where, $S_p \rightarrow$ signal power

$N_p \rightarrow$ output noise power.

To find Signal Power:-

Signal power is given as

$P = V^2/R$ where V is the rms value of the signal.

Substituting $V = \frac{A_m}{\sqrt{2}}$, we get

$$P = \frac{A_m^2}{2} \quad \text{—— (12)} \quad \because R=1$$

$\therefore$ Signal Power (Substituting (11) in (12))

$$\boxed{S_p = \frac{\Delta^2 f_s^2}{8\pi^2 f_m^2}} \quad \text{—— (13)}$$

To find Noise Power:-

In DM, the maximum quantization error is equal to step size 'Δ'.

quantization error be uniformly distributed over an interval $(-\Delta, \Delta)$

The PDF of quantization error can be expressed as

$$f_Q(q) = \frac{1}{2\Delta} \quad \text{for } -\Delta < q < \Delta \quad \to (14)$$

Quantization Noise power $(\sigma_q^2) = N_q = \int_{-\Delta}^{\Delta} q^2 f_Q(q)\, dq \quad \to (15)$

Substituting (14) in (15)

$$\therefore N_q = \int_{-\Delta}^{\Delta} q^2 \frac{1}{2\Delta}\, dq$$

$$N_q = \frac{\Delta^2}{3} \quad \text{--- (16)}$$

The lowpass reconstruction filter with cut off frequeny W, present at the delta modulator receiver, passes part of the noise power uniforml distributed over $-f_s$ to f_s range, to the output

$\therefore$ Output Noise Power $= \frac{W}{f_s} \times$ Noise Power (N_q) → (17)

Sustituting (16) in (17)

$$\therefore \text{Output Noise Power } (N_p) = \frac{W}{f_s} \frac{\Delta^2}{3} \quad \text{--- (18)}$$

$$\therefore \text{SNR} = \frac{\text{Normalized Signal Power}}{\text{Output Noise power}}$$

$$\text{SNR} = \frac{S_p}{N_p} = \frac{(13)}{(18)} = \frac{\dfrac{\Delta^2 f_s^2}{8\pi^2 f_m^2}}{\dfrac{W\Delta^2}{3 f_s}}$$

$$\boxed{\text{SNR} = \frac{3 f_s^3}{8\pi^2 W f_m^2} \text{ or } \frac{3}{8\pi^2 W f_m^2 T_s^3}} \quad \text{--- (19)}$$

Where, $f_s \rightarrow$ Sampling frequency

$T_s \rightarrow$ Sampling Period

$f_m \rightarrow$ Message signal frequeny

Minimum Transmission bandwidth

$$B_{T\,min} = \frac{f_s}{2};$$

Bandwidth expansion factor $b = \frac{\text{Transmission b.}}{\text{Signal}}$

$$b = \frac{B_T}{W} = \frac{f_s}{2W}.$$

2.3.5 ADVANTAGES AND DISADVANTAGES OF DM:-

Advantages:

1. Low signaling rate and low transmission channel bandwidth, because here only one bit is transmitted per sample.

2. DM Transmitter and receiver are less complicated to implement as compared to PCM. There is no need for synchronization circuits.

Disadvantages:

1. Slope overload error and granular noise are present.
2. Practically the signaling rate with no slope overload error will be much higher than that of PCM.

(2.4) ADAPTIVE DELTA MODULATION:

A modification of DM called adaptive DM in which step size is not kept fixed. Rather ~~ss~~ step size is varying with respect to changes in the input (or) message signal.

* Step size of the modulator is adapted with the level of Input signal such that, step size is increased during a steep segment of the input signal and step size is reduced when input signal having flat segment.

* To minimize slope overload noise while holding the granular noise at a resonable value ADM is used.

Operation:-

In ADM we have two modes of operation.

(i) Granular Mode

(ii) Slope overload Mode.

Granular Mode:

If successive errors are of opposite polarity, then the delta modulator operates in the granular mode that is step size is reduced.

Slope overload Mode:

If successive errors are of same polarity then delta modulator is operates in slope overload mode. that is step size is incresed.

2.4.1 ADM Transmitter:-

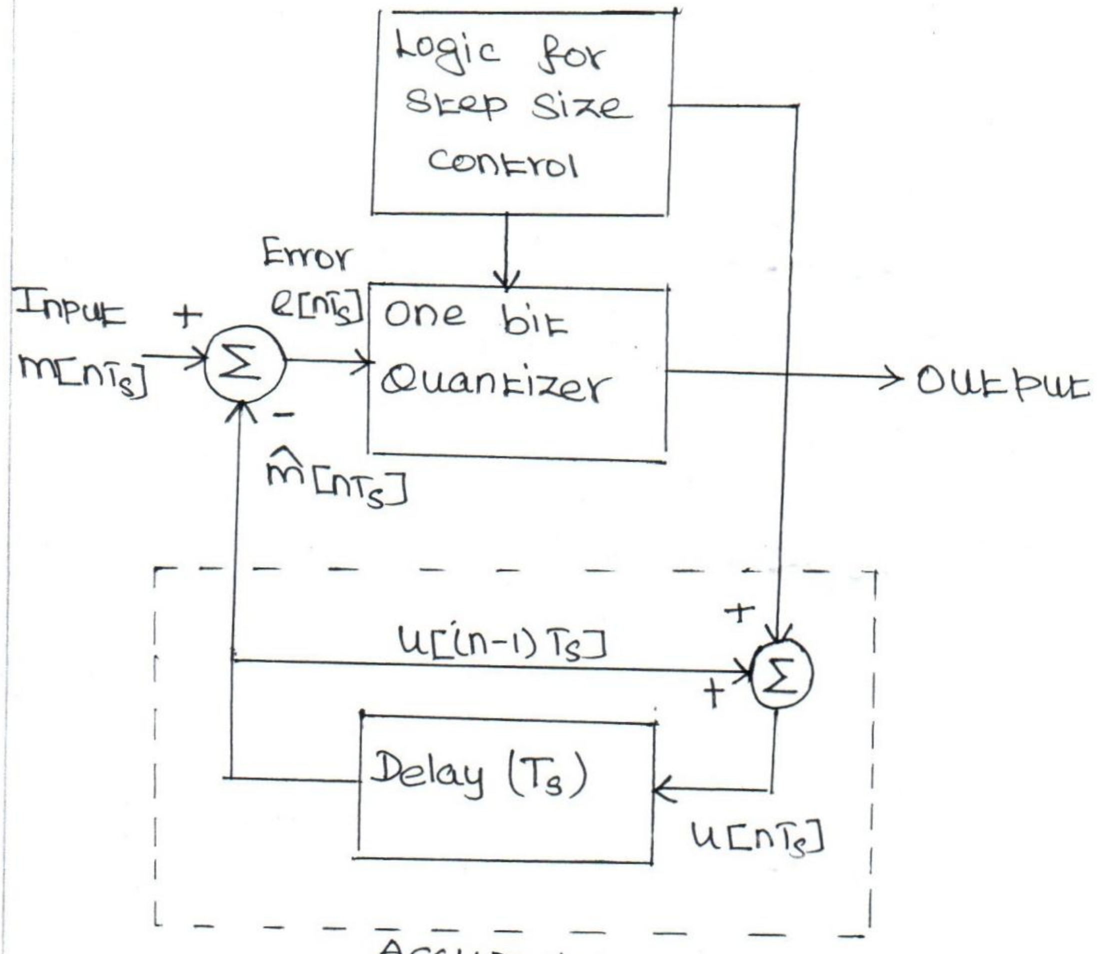


Fig: 6 ADM Transmitter.

* In ADM transmitter "logic for step size control" is the only additional block. the remaining blocks are same as that of DM transmitter.

* Depending on the output of one bit quantizer the step size is increased (or) decreased according to a specified logic.

2.4.2 ADM RECEIVER:

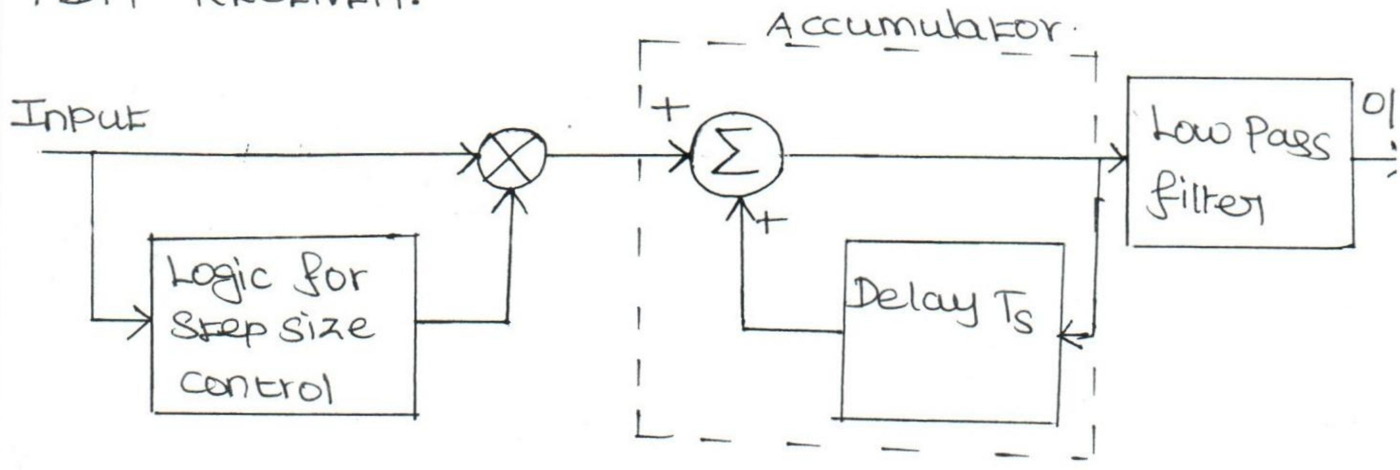


Fig: 7 ADM Receiver.

* The entire operation of receiver is divided into two parts. In the first stage, for each incoming bit step size is produced. the stepsize is decided by both previous and present inputs.

* Then it is applied to the accumulator whose function is to generate staircase waveform depending on the input applied to it.

* The original message signal is reconstructed from the staircase waveform using low pass filter, which allows only the maximum frequency of the message signal remaining high frequency will be attenuated.

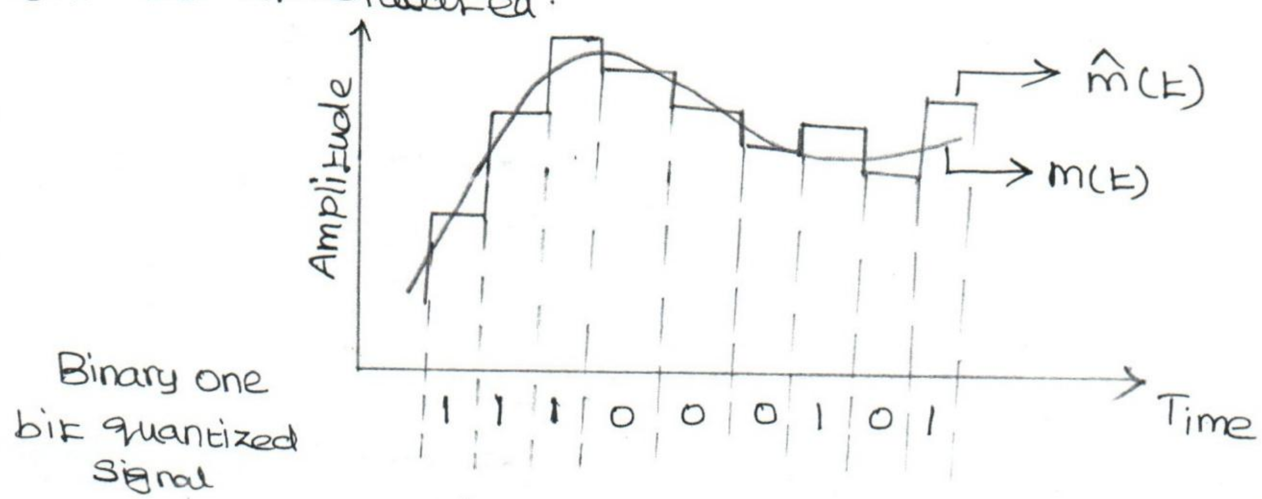


Fig: 8 Waveform of ADM

ADVANTAGES OF ADM:

1. The signal to noise ratio is better than ordinary DM because of reduction in slope overload distortion and granular noise.
2. Signaling rate is Low.
3. Bandwidth utilization is better than Delta modulation.
4. Wide dynamic range because of variable step size.

2.5 DIFFERENTIAL PULSE CODE MODULATION :- (DPCM)

In PCM, Successive samples are carrying the same information with little difference. when these samples are encoded by standard PCM system, resulting encoded signal contains redundant information. By removing the PCM information redundancy a more efficient coded signal may be obtained. This is done using DPCM.

Principle: DPCM works on the principle of Linear Prediction.

Definition: In DPCM, the difference in the amplitude of two successive samples is transmitted rather than actual value.

* In DPCM value of present sample is predicted by a linear combination of the past sample values.

* Error is computed between original and predicted samples. Error signal is small compared with original one. Hence less bits are required to encode them.

2.5.1 DPCM TRANSMITTER:

* The information signal $m(t)$ is sampled at a rate 'f_s' to produce the sampled signal $m[nT_s]$, where T_s is the interval between adjacent samples.

* The predictor filter produce a predicted version of the sampled input and the predictor output be denoted by $\hat{m}[nT_s]$.

* The input signal to quantizer is defined by

$$\boxed{e[nT_s] = m[nT_s] - \hat{m}[nT_s]} \rightarrow 1$$

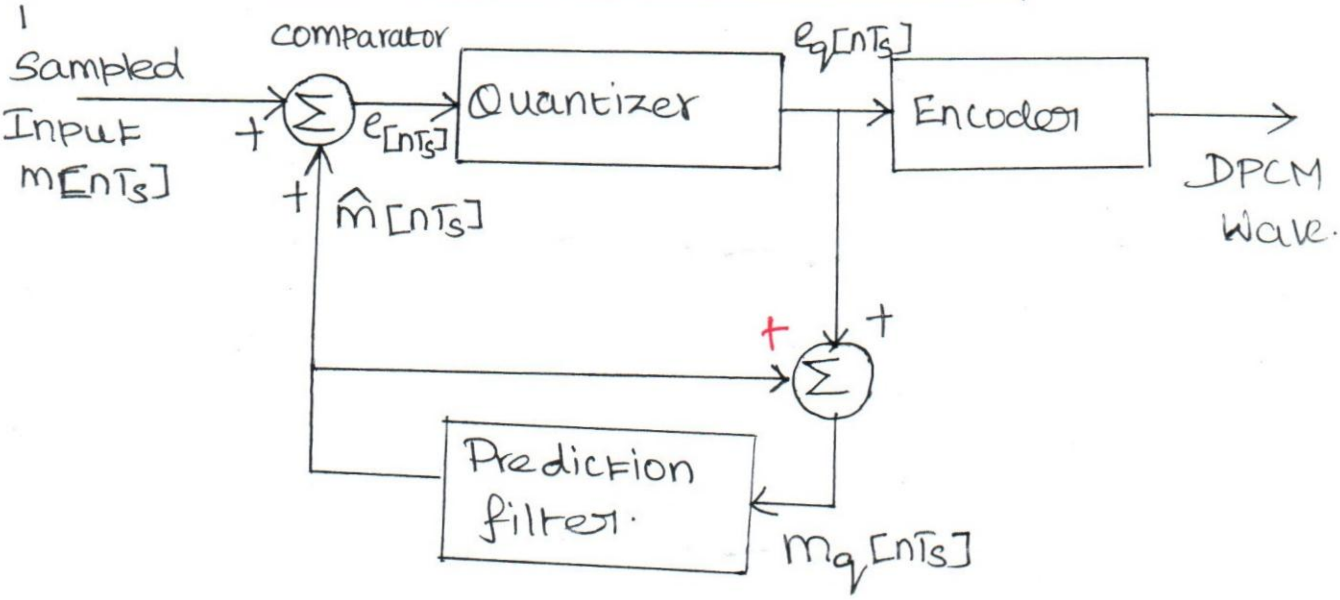


Fig: 9 DPCM Transmitter.

Prediction Error:-

* $e[nT_s]$ is called as prediction error signal because it represents the difference between the unquantized input sample $m[nT_s]$ and it's predicted value $\hat{m}[nT_s]$.

* Quantizer output may be expressed as,

$$e_q[nT_s] = e[nT_s] + q[nT_s] \rightarrow 2$$

$q[nT_s] \rightarrow$ Quantization Error

$e[nT_s] \rightarrow$ Prediction Error

* Input of Prediction filter is

$$m_q[nT_s] = \hat{m}[nT_s] + e_q[nT_s] \rightarrow \quad 3$$

Sustisuting Equation 2 in 3 then we get

$$m_q[nT_s] = \hat{m}[nT_s] + e[nT_s] + q[nT_s] \rightarrow \quad 4$$

By using equation 1 we can write equation 4 as,

$$m_q[nT_s] = m[nT_s] + q[nT_s] \rightarrow \quad 5$$

* Therefore, the quantized sample $m_q[nT_s]$ at the prediction filter input differs from the original input sample value $m[nT_s]$ by quantization error $q[nT_s]$. The quantization error can be positive or negative.

2.5.2 DPCM Receiver:

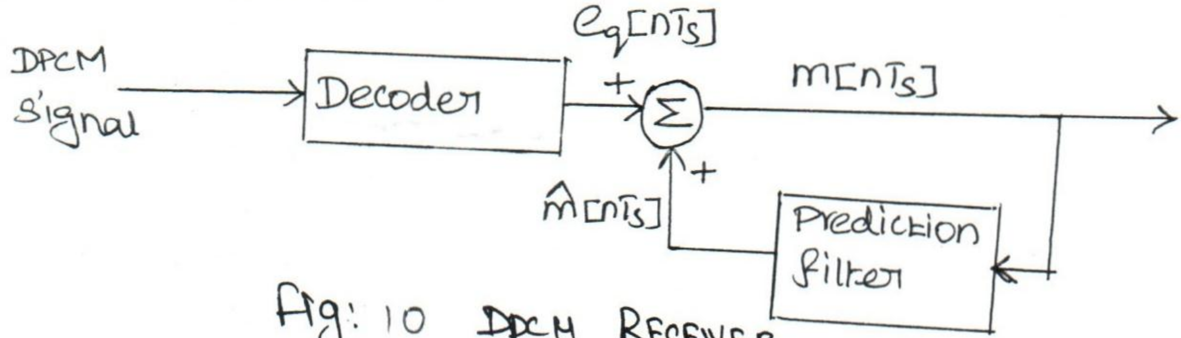


Fig: 10 DPCM RECEIVER

* The quantized error signal is $e_q[nT_s]$ is reconstructed from the incoming binary signal (DPCM) by the decoder block.

* The quantized version of the original signal is obtained by adding both predictor output and quantized error signal.

$$m[nT_s] = e_q[nT_s] + \hat{m}[nT_s] \rightarrow \quad 6$$

Where,

$m[nT_s]$ is receiver output.

2.5.3 SNR IN DPCM:-

Output signal to quantization noise,

$$(SNR)_0 = \frac{\sigma_M^2}{\sigma_Q^2} \rightarrow 7$$

Where, $\sigma_M^2 \rightarrow$ Variance of Input (or) Message Signal.

$\sigma_Q^2 \rightarrow$ Variance of Quantization Error

Multiply and Divide above Equation (7) by (σ_E^2) Variance of prediction Error.

$$(SNR)_0 = \frac{\sigma_M^2}{\sigma_E^2} \times \frac{\sigma_E^2}{\sigma_Q^2} \rightarrow 8$$

$$(SNR)_0 = G_P \; (SNR)_{PCM} \rightarrow 9$$

Where,

$G_P \rightarrow \frac{\sigma_M^2}{\sigma_E^2}$ is the Prediction Gain

$(SNR)_{PCM} \rightarrow \frac{\sigma_E^2}{\sigma_Q^2}$ is the signal to quantization noise ratio.

The factor G_p is the processing gain produced by the DPCM quantization scheme when $G_p > 1$, which is the case most of the time, it represents the gain in SNR obtained by using DPCM compared to PCM.

Advantages:-

1. Here only a small difference voltage is to be quantized and encoded. this will need less number of quantization levels and hence less number of bits to represent them.

2. The signaling rate and bandwidth of a DPCM system will be less than PCM.

Disadvantages:-

1. Needs the predictor circuit to be used which is very complex.

2. Practical usage is limited.

3 DIGITAL MODULATION SCHEME:

3.1 INTRODUCTION:

Transmission of digital signal classified into two types.

Digital Transmission

↓

- Base band Data Transmission
- Pass Band Data Transmission.

Base band Data Transmission:-

* Basic band of information alone transmitted to the receiver with out carrier.

* Base band transmission does not required carrier. Hence there is no need for Modulation also.

* This kind of transmission - suitable for short distance communication.

Pass Band Data Transmission

* Information and carrier signal are added together and transmit to the receiver.

* pass band transmission involves high frequency carrier. to modulate the information signal hence this method needs Modulation.

* Pass band data transmission is also referred as Band Pass transmission & this method is suitable for long distance communication.

3.1.1 Types of Pass Band Data Transmission.

- Linear (or) Memoryless Modulation
- Non- Linear (or) Modulation with Memory.

Memoryless Modulation:-

* Memoryles modulation is a scheme in which the waveform transmitted in any time interval does not depends on previous digital symbols.

* Memoryless modulation does not required memory device. Example: ASK, BPSK, BFSK QPSK, QASK, M-ary FSK, M-ary PSK.

Modulation with memory:-

Is a scheme in which the waveform transmitted in any time interval depends on one or more previous digital symbols.

* This type of modulation requires memory device to store previous symbols.

Example: continuous Phase FSK, MSK & DPSK.

5.4.3 Detection of PSK:-

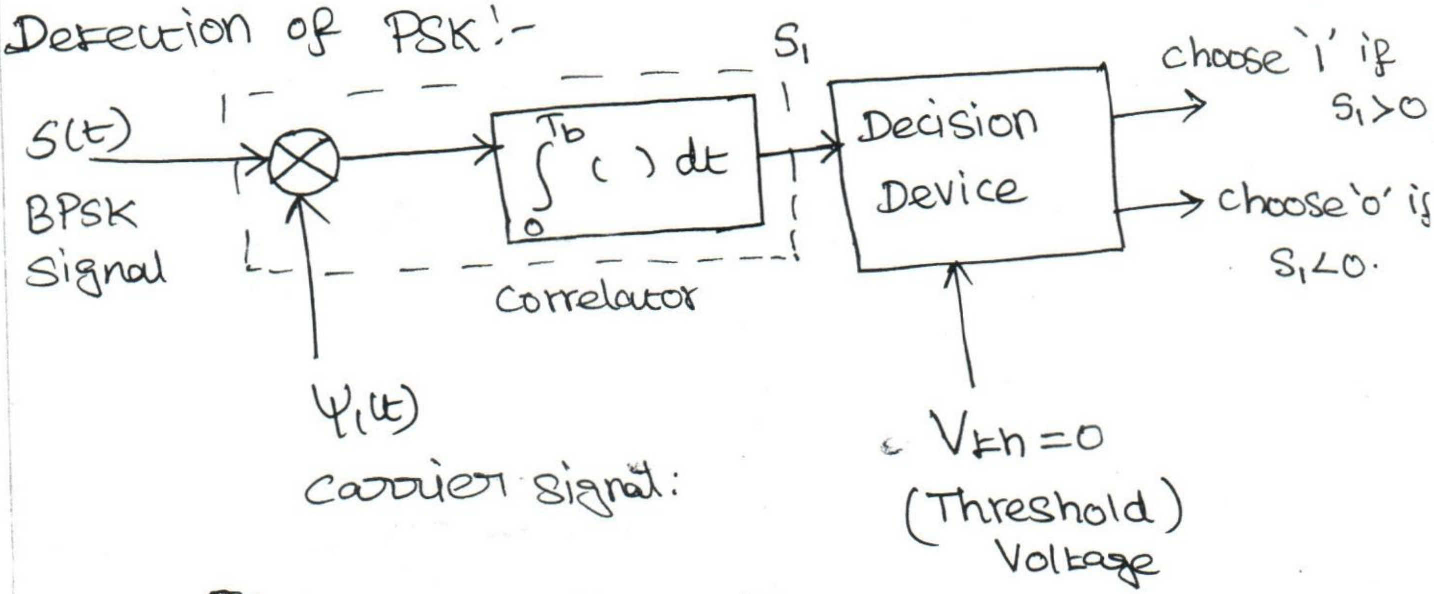


Fig: 11 Block diagram of PSK Detection

* At the input of correlator Noisy BPSK is present.
* The carrier signal $\varphi_1(t)$ is multiplied with $s(t)$ and integrate by integrator over the period of 0 to T_b.
* The output of integrator is compared with threshold at decision device.
* The correlator output S_1 is compared with threshold 0 Volts

If $S_1 > 0$, receiver decides in favour of symbol '1'

$S_1 < 0$, receiver decides in favour of symbol '0'.

3.4.4 Signal Space Diagram (or) constellation Diagram PSK:

Binary PSK has two different symbols '1' & '0'.

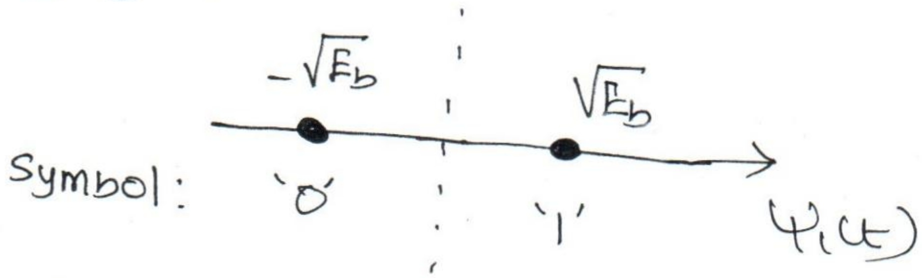


Fig: 11 SSD of BPSK

3.4.5 Waveforms of PSK:-

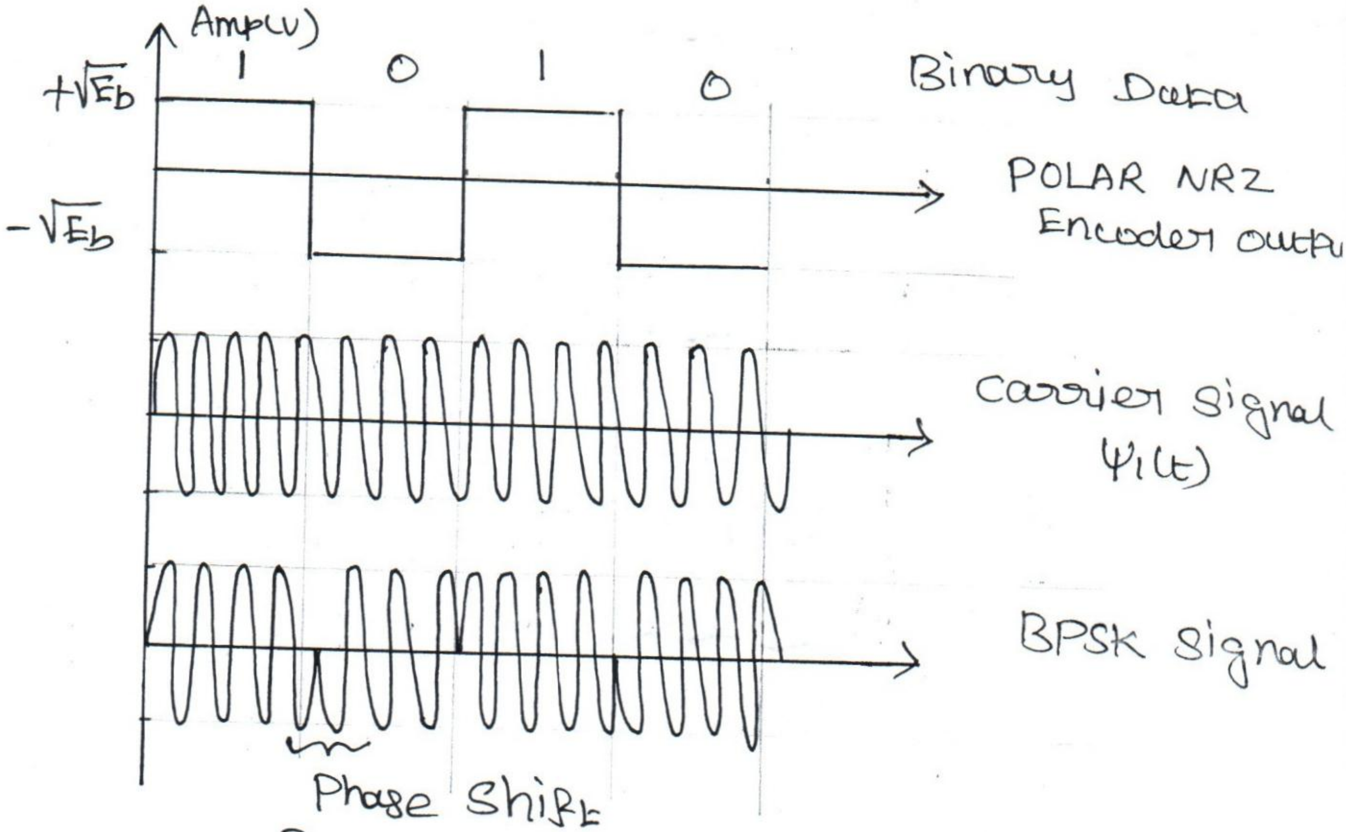


Fig: 12 Waveform of PSK

* Bandwidth of a BPSK signal is

$$\boxed{B.W = 2f_b}$$

* Probability of Error.

$$P_e = \frac{1}{2}\,\mathrm{erfc}\left(\sqrt{\frac{E_b}{N_0}}\right)$$

Advantages:-

* Good Noise immunity
* Simple Circuit
* Better performance in the presence of noise.

Disadvantages:

* It requires synchronization between transmitter and receiver
* It requires larger Bandwidth.

Applications:-

* Employed in communication system with higher Bit rates.
* Used in Satellite communication.

3.5 Quadrature Phase Shift Keying: QPSK:

3.5.1 Definition:-

"Phase of the carrier is shifted by 90° ~~an~~ in accordance with digital inputs."

* Phase shift occurs in four levels ($\pi/4$, $3\pi/4$, $5\pi/4$ & $7\pi/4$).

* Two successive bits are grouped together and represented by a different value of phase shift of the carrier. So signaling rate and bandwidth are reduced.

3.5.2 Representation of QPSK:-

$$S_{QPSK}(t) = \sqrt{P_s}\, b_o(t) \cos w_c t + \sqrt{P_s}\, b_e(t) \sin w_c t \quad \rightarrow (1)$$

Where $b_o(t) \rightarrow$ odd bit sequence

$b_e(t) \rightarrow$ Even bit sequence.

$P_s \rightarrow$ Normalized power

$\sqrt{P_s} \cos w_c t \rightarrow \varphi_1(t)$

$\sqrt{P_s} \sin w_c t \rightarrow \varphi_2(t)$

$\varphi_1(t)$ & $\varphi_2(t)$ are orthogonal carrier.

$\varphi_1(t)$ is used to carry odd bits of binary data, $\varphi_2(t)$ used to carry even bits of binary data stream.

Dibits:-

* Each QPSK symbol can be mapped to a unique pair of input bits called dibits.

* One of the bit in each dibit is transmitted by cosine carrier ($\psi_1(t)$) & other by sine carrier ($\psi_2(t)$).

3.5.2 Generation of QPSK:-

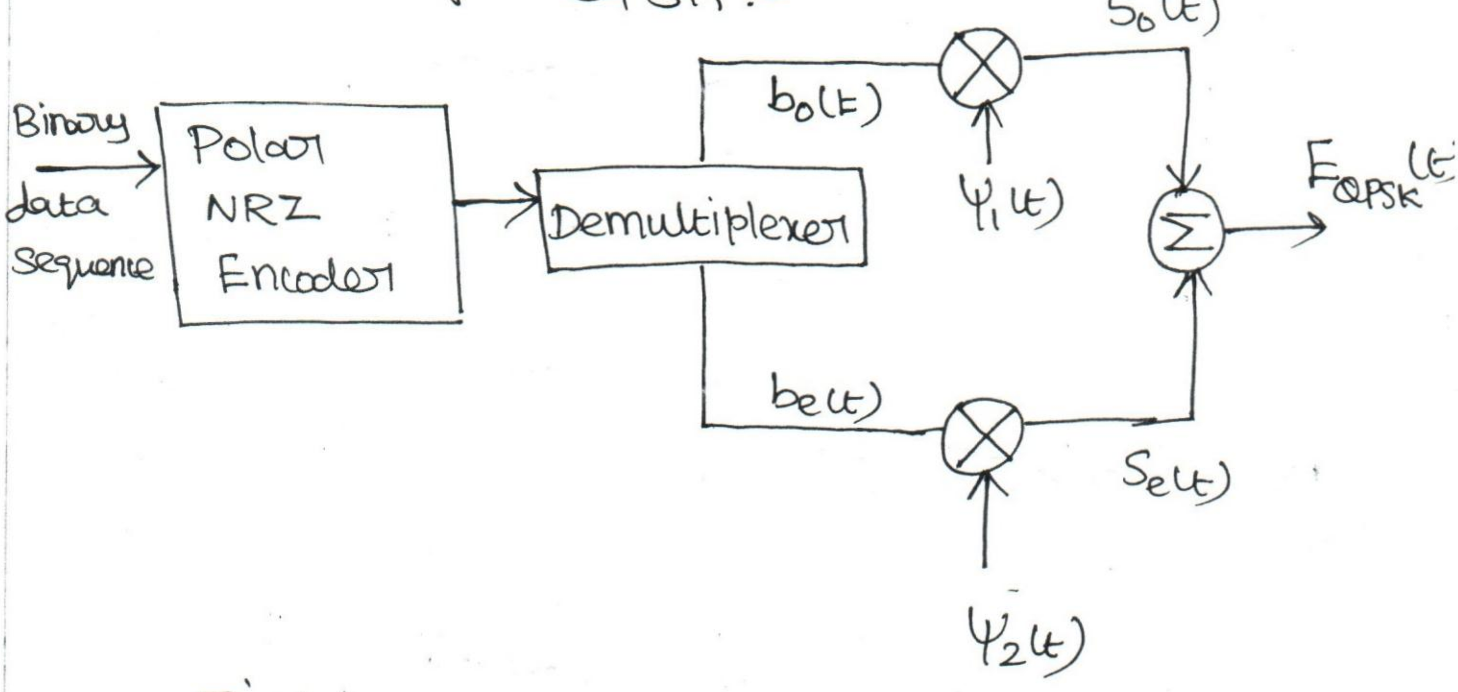


Fig: 13 Block diagram of QPSK Transmitter.

* The Polar NRZ Level encoder converts binary Input '1' to $+\sqrt{E_b}$ V

'0' to $-\sqrt{E_b}$ V

* Each symbol in QPSK system have two bits, symbol energy is twice the bit energy

$$\boxed{E = 2E_b}$$

E → symbol energy & E_b → bit energy

* Incoming binary signal from Encoder is divided by means of demultiplexer in to two seperate sequence

    (i) odd bit sequence $S_o(t)$

    (ii) Even bit sequence $S_e(t)$

* $S_o(t)$ is multiplied by $\varphi_1(t)$, the same way $S_e(t)$ is multiplied by $\varphi_2(t)$ at the product modulator i-e multiplier.

* The result is a pair of BPSK signals, These signals are added to produce desired QPSK signal.

* From Eqn (17).

Symbol	$b_o(t)$	$b_e(t)$	$F_{QPSK}(t)$
1 1	1	1	$\sqrt{P_s}\cos\omega_c t + \sqrt{P_s}\sin\omega_c t$
0 1	-1	1	$-\sqrt{P_s}\cos\omega_c t + \sqrt{P_s}\sin\omega_c t$
0 0	-1	-1	$-\sqrt{P_s}\cos\omega_c t - \sqrt{P_s}\sin\omega_c t$
1 0	1	-1	$\sqrt{P_s}\cos\omega_c t - \sqrt{P_s}\sin\omega_c t$

3.5.4 Detection of QPSK:-

The QPSK receiver consists of a pair of correlators called in phase channel and Quadrature channel.

* Received QPSK signal is applied to correlators as common input and which is multiplied with respective carrier $\varphi_1(t)$ & $\varphi_2(t)$.

* correlator outputs i-e inphase channel output S_1 & Quadrature channel output S_2 are compared with threshold voltage at Decision device.

* In Phase Channel output is 'S_1' is

If $S_1 > 0$ (i-e V_{th}) Receiver detects '1'

$S_1 < 0$ (i-e V_{th}) Receiver detects '0'.

* Quadrature Channel output is 'S_2'

If $S_2 > 0$ (i-e) V_{th} Receiver detects '1'

$S_2 < 0$ (i-e) V_{th} Receiver detects '0'

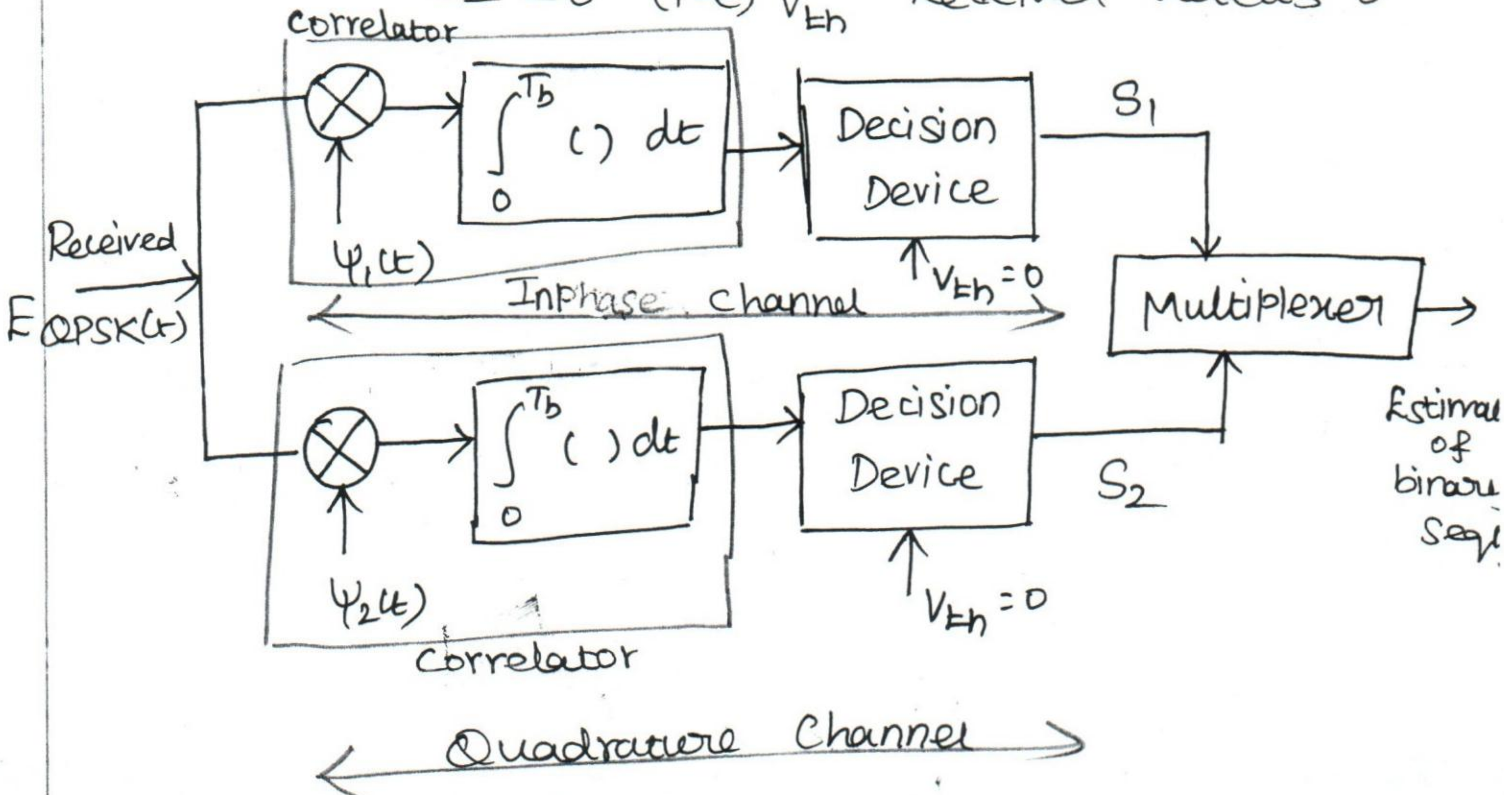


Fig: 14 Detection of QPSK

3.5.b Constellation Diagram or Signal Space Diagram

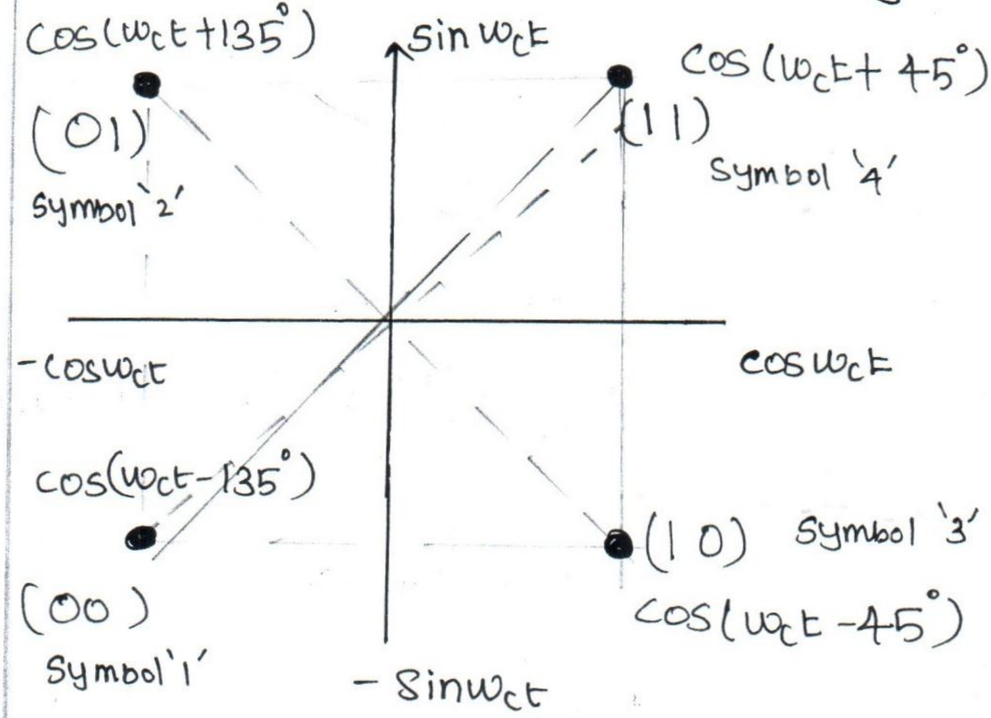


Fig. 15

SSD of QPSK

	Binary Input		QPSK Phase
	odd bit	even bit	
QPSK symbol 1,	0	0	-135°
symbol 2,	0	1	135°
symbol 3,	1	0	-45°
symbol 4,	1	1	45°

Truth Table

* Bandwidth of QPSK; $BW = f_b$
* Bit error Probability of QPSK is :

$$P_e = \frac{1}{2} \operatorname{erfc} \sqrt{\frac{E_b}{N_0}}$$

$E_b \rightarrow$ Bit energy

$N_0 \rightarrow$ PSD of Noise.

3.5.7 Waveforms of QPSK:

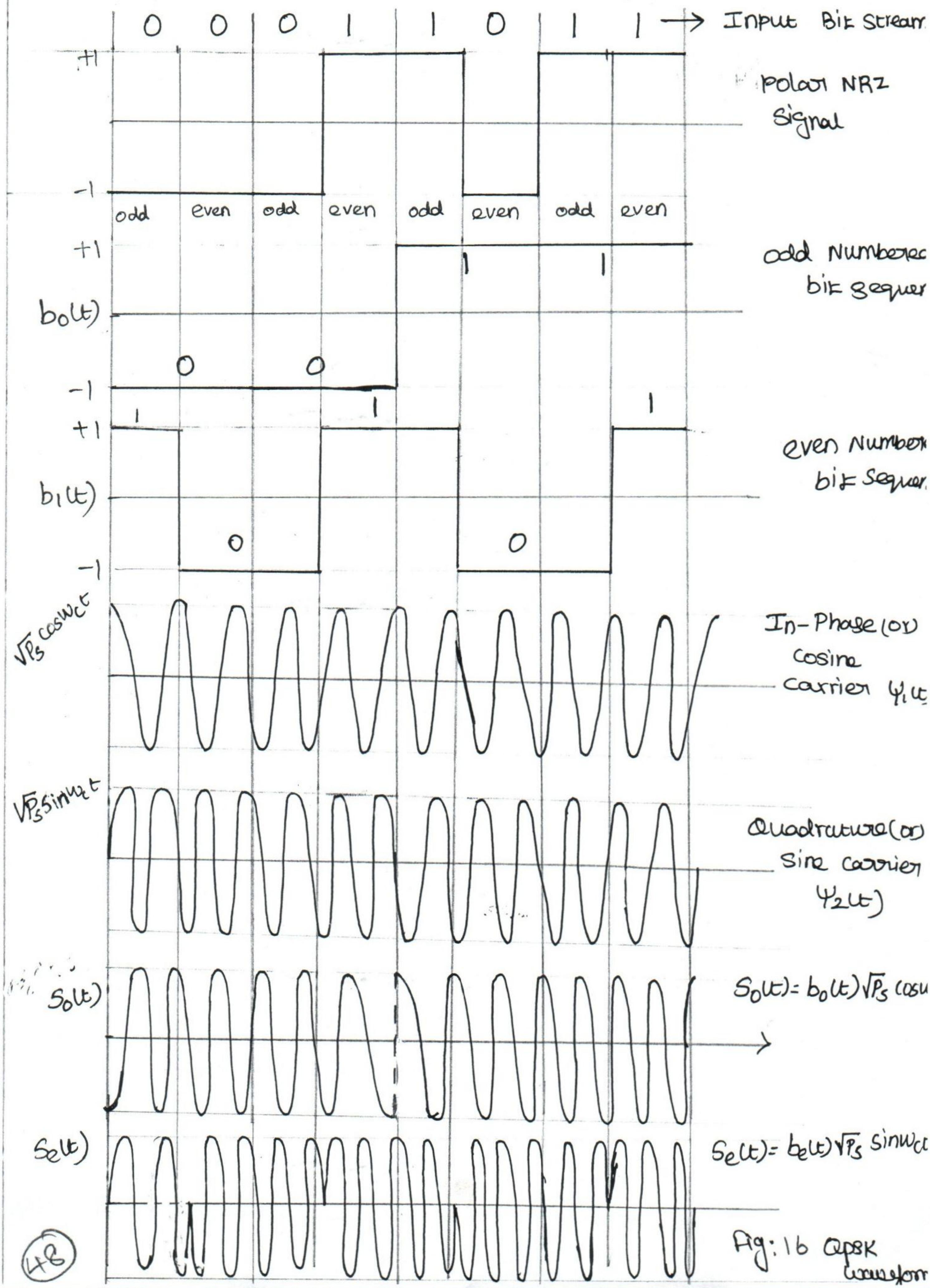


Fig: 16 QPSK waveform

Advantages of QPSK:-

* Efficient Bandwidth Utilization Method.
* Bandwidth required by QPSK is reduced to half as compared to BPSK
* Good Noise immunity.
* Less Error Probability

Disadvantages of QPSK:-

* Complex generation and detection.

3.6 QUADRATURE AMPLITUDE MODULATION (QAM)

(or)

QUADRATURE AMPLITUDE SHIFT KEYING

3.6.1 Definition:-

QAM is a combination of amplitude and phase modulation scheme.

If the amplitude and phase of carrier is varied noise immunity is increased. Such a system called QAM (i-e)

"In QAM both amplitude and phase of the carrier signal ~~is~~ are varied in accordance with digital input signal".

3.6.2 Representation:-

$$S_i(t) = \sqrt{\frac{2E_s}{T_s}}\, K_i \cos \omega_c t - \sqrt{\frac{2E_s}{T_s}}\, L_i \sin \omega_c t \quad \rightarrow (1)$$

where, $E_s \rightarrow$ Symbol Energy

$T_s \rightarrow$ Symbol duration

K_i & $L_i \rightarrow$ A pair of constant choosen according to the location of particular signal point.

* Two orthogonal carriers are used (i-e)

$$\varphi_1(t) = \sqrt{\frac{2}{T_s}} \cos \omega_c t \quad \rightarrow (2)$$

$$\varphi_2(t) = \sqrt{\frac{2}{T_s}} \sin \omega_c t \quad \rightarrow (3)$$

by substituting (2) & (3) in (1)

(1) can be written as.

$$S_i(t) = \sqrt{E_s}\, K_i\, \psi_1(t) - \sqrt{E_s}\, L_i\, \psi_2(t) \quad -(4)$$

Types of QAM:-

1) 4 QAM 2) 8 QAM 3) 16 QAM 4) 32 QAM

5) 64 QAM.

* In 4 QAM, 4 different symbols available. Each symbol has 2 bits.

* In 8 QAM, 8 different symbols available. Each Symbol has 3 bits.

* Simillarly, 16 QAM,- 16 symbols - 4 bits per symbol

32 QAM - 32 symbols - 5 bits/symbol

64 QAM - 64 " - 6 bits/symbol

3.6.3 Generation of 16 QAM:-

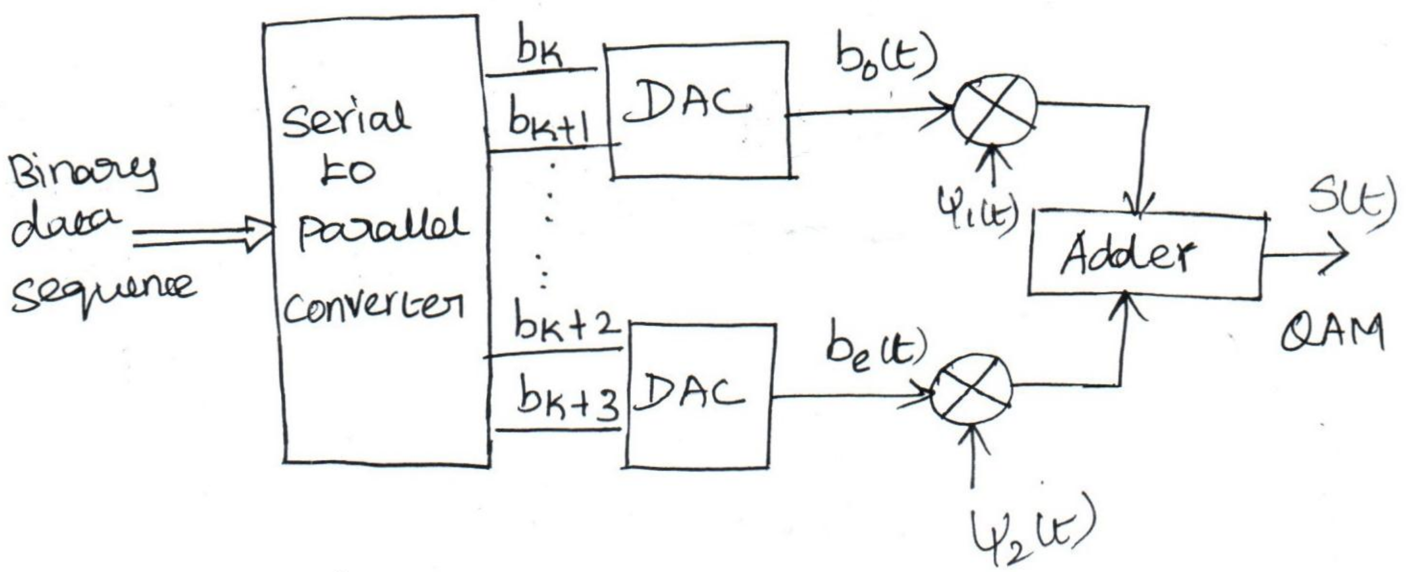


Fig: 17 16 QAM Transmitter.

* 16 QAM cosists 16 symbols, each symbol has 4 bits.

* Serial to parallel converter is used to convert serial data in to 4 successive ~~parr~~allel bits

* The bits b_K & b_{K+1} are applied to the upper DAC, b_{K+2} & b_{K+3} are applied to lower DAC.

* The output of DAC are $A_o(t)$ & $A_e(t)$, these are multiplied with respective carriers $\varphi_1(t)$ & $\varphi_2(t)$.

* The output of the adder ~~is~~ s(t) is QAM signal

$$S(t) = A_o(t)\sqrt{P_s}\cos\omega_c t + A_e(t)\sqrt{P_s}\sin\omega_c t.$$

3.6.14 Detection of QAM:-

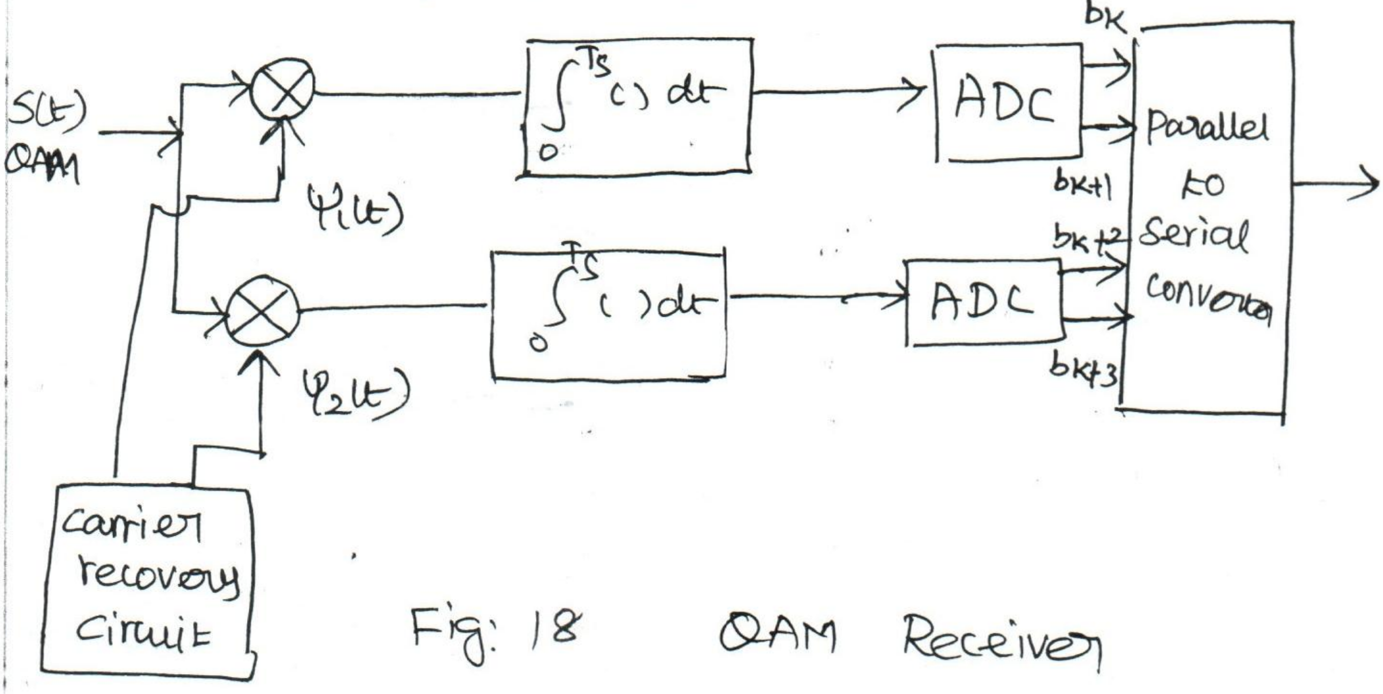


Fig: 18 QAM Receiver

* The carrier recovery circuit used to recover the carrier from received signal.
* The received signal is multiplied with this carrier then applied to the correlator
* Parallel to serial converter used to convert parallel bits into serial bits.

3.6.5 Constellation Diagram :-

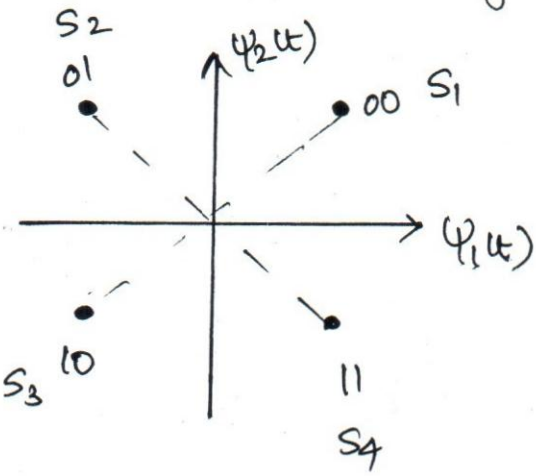


Fig. 19. a) 4 QAM

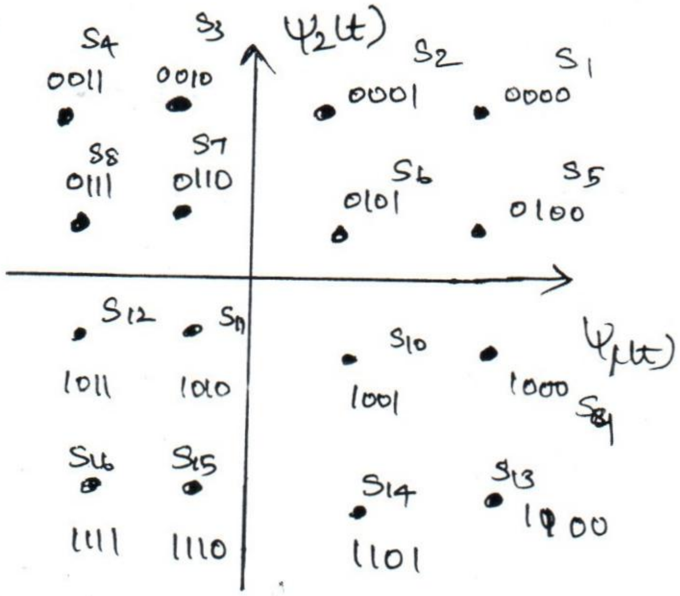


Fig 19. b) 16 QAM

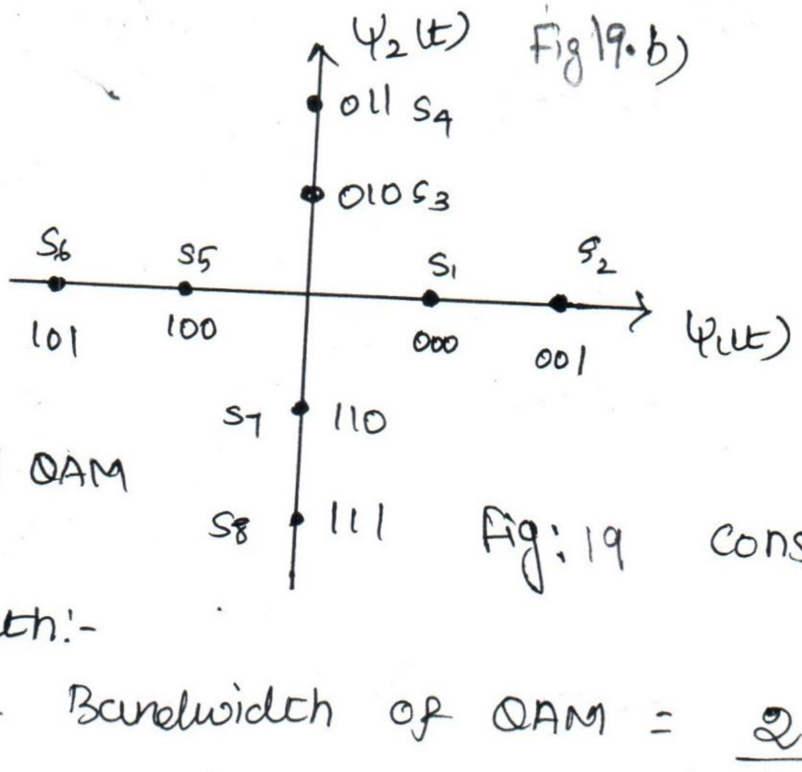


Fig 19. c) 8 QAM

Fig: 19 Constellation Diagram

Bandwidth :-

* Bandwidth of QAM $= \dfrac{2}{NT_b} = \dfrac{2f_b}{N}$
* Probability of error $P_e \simeq 2\left(1 - \frac{1}{\sqrt{M}}\right) \operatorname{erfc}\left(\sqrt{E_b/N_0}\right)$

3.7 MINIMUM SHIFT KEYING (MSK):

3.7.1 Introduction:-

* MSK is a spectrally efficient modulatio scheme.

* In MSK the output waveform is continuous in phase hence there are no abrupt changes in amplitude.

Definition:-

Minimum shift keying is a type of binary continuous phase frequency shift keying (CPFSK), where in the peak frequency deviation is equal to $1/2$ the bit rate & modulation index $h = 1/2$.

(i-e) Frequency difference between symbol '1' and symbol '0' is equal to half the data rate, and Modulation Index is 0.5.

Frequency Deviation:-

$$\Delta_f = \frac{1}{2T_b} \quad \text{—} (1)$$

$T_b \rightarrow$ Bit duration.

* Let the frequency f_1 & f_2 represent the symbol '1' & symbol '0' transmission respectively.

* Carrier frequency $f_c = \frac{f_1 + f_2}{2}$ — ②

* Frequency Deviation also defined in terms of f_1 & f_2.

$$\Delta_f = f_1 - f_2 \quad — ③$$

* Frequency for symbol '1' (i-e) f_1 can be defined as

$$f_1 = f_c + \frac{\Delta_f}{2} \quad — ④$$

Substitute ② & ③ in ④

$$\therefore f_1 = \frac{f_1 + f_2}{2} + \frac{f_1 - f_2}{2} \quad — ⑤$$

* Similarly Frequency for symbol 'o'

$$f_2 = f_c - \Delta_f/2 \quad — ⑥$$

Substitute ② & ③ in ⑥

$$f_2 = \frac{f_1 + f_2}{2} - \frac{f_1 - f_2}{2} \quad — ⑦$$

3.1.2
Representation:-

$$S(t) = E_c \cos[2\pi f_c t + \phi(t)] \quad — ⑧$$

where, $\phi(t) = \pm \pi \Delta_f t$ — ⑨

$\phi(t) \rightarrow$ Phase of the carrier.

If $\Delta f = \frac{1}{2T_b}$ (9) Eqn can be written as

$$\phi(t) = \frac{\pi t}{2T_b} \quad \text{--- (10)}$$

For binary symbol '1' $(8)^{th}$ Eqn can be written as, sub (10) in (8)

$$S(t) = E_c \cos\left[2\pi f_c t + \frac{\pi t}{2T_b}\right] \rightarrow (11)$$

For binary symbol '0', sub $\phi(t) = -\frac{\pi t}{2T_b}$ in Eqn (8),

$$\therefore S(t) = E_c \cos\left[2\pi f_c t - \frac{\pi t}{2T_b}\right] \rightarrow (12)$$

3.7.3 MSK Transmitter:-

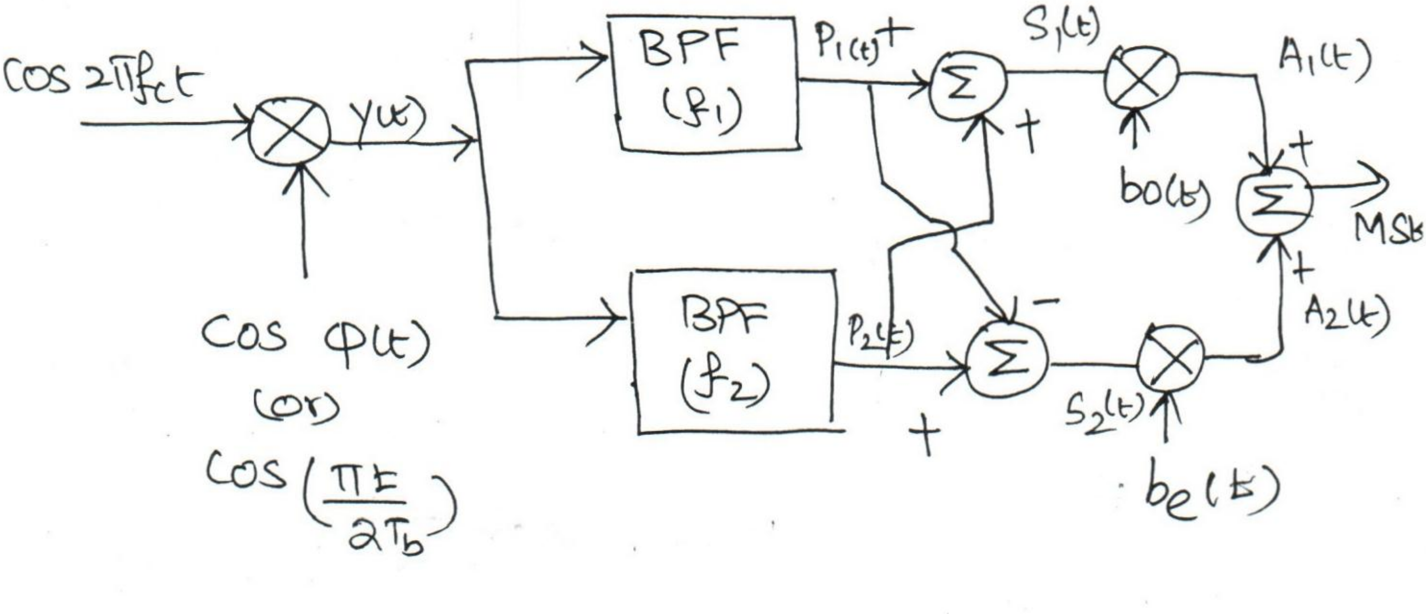


Fig: 20 Block diagram of MSK Transmitter.

* In MSK Transmitter, Two sinusoidal signals are applied to the product modulator.

* The output the modulator is two sinusoidal signals with frequencies 'f_1 & f_2'

The output of Modulator = $\cos 2\pi f_c t \cos \phi(t)$

(i-e) $= \cos 2\pi f_c t \cos\left(\frac{\pi t}{2T_b}\right)$

$$y(t) = \frac{1}{2}\left[\cos 2\pi\left(f_c + \frac{1}{4T_b}\right)t + \cos 2\pi\left(f_c - \frac{1}{4T_b}\right)t\right]$$

$$y(t) = \frac{1}{2}\left[\cos 2\pi\left(f_c + \frac{f_b}{4}\right)t + \cos 2\pi\left(f_c - \frac{f_b}{4}\right)t\right]$$

where

$$f_1 = f_c + \frac{f_b}{4}, \quad f_2 = f_c - \frac{f_b}{4}.$$

* These two frequencies are seperated from each other by two band pass filters centered at f_1 and f_2 respectively.

$$P_1(t) = \frac{1}{2}\cos 2\pi\left(f_c + \frac{f_b}{4}\right)t$$

$$P_2(t) = \frac{1}{2}\cos 2\pi\left(f_c - \frac{f_b}{4}\right)t.$$

* The out puts of band pass filters are then added and subtracted to produce $S_1(t)$ & $S_2(t)$

$$S_1(t) = \frac{1}{2}\cos 2\pi\left(f_c + \frac{f_b}{4}\right)t + \frac{1}{2}\cos 2\pi\left(f_c - f_b/4\right)$$

$$S_2(t) = \frac{1}{2}\cos 2\pi\left(f_c + f_b/4\right) - \frac{1}{2}\cos 2\pi\left(f_c - f_b/4\right).$$

* Finally $S_1(t)$ & $S_2(t)$ are multiplied with odd and even bit of the binary input $b_o(t)$ & $b_e(t)$ with bit rate equal to $1/2\, T_b$.

* The output of the multipliers $A_1(t)$ & $A_2(t)$ are added to produce MSK signal.

3.7.4 MSK Receiver:-

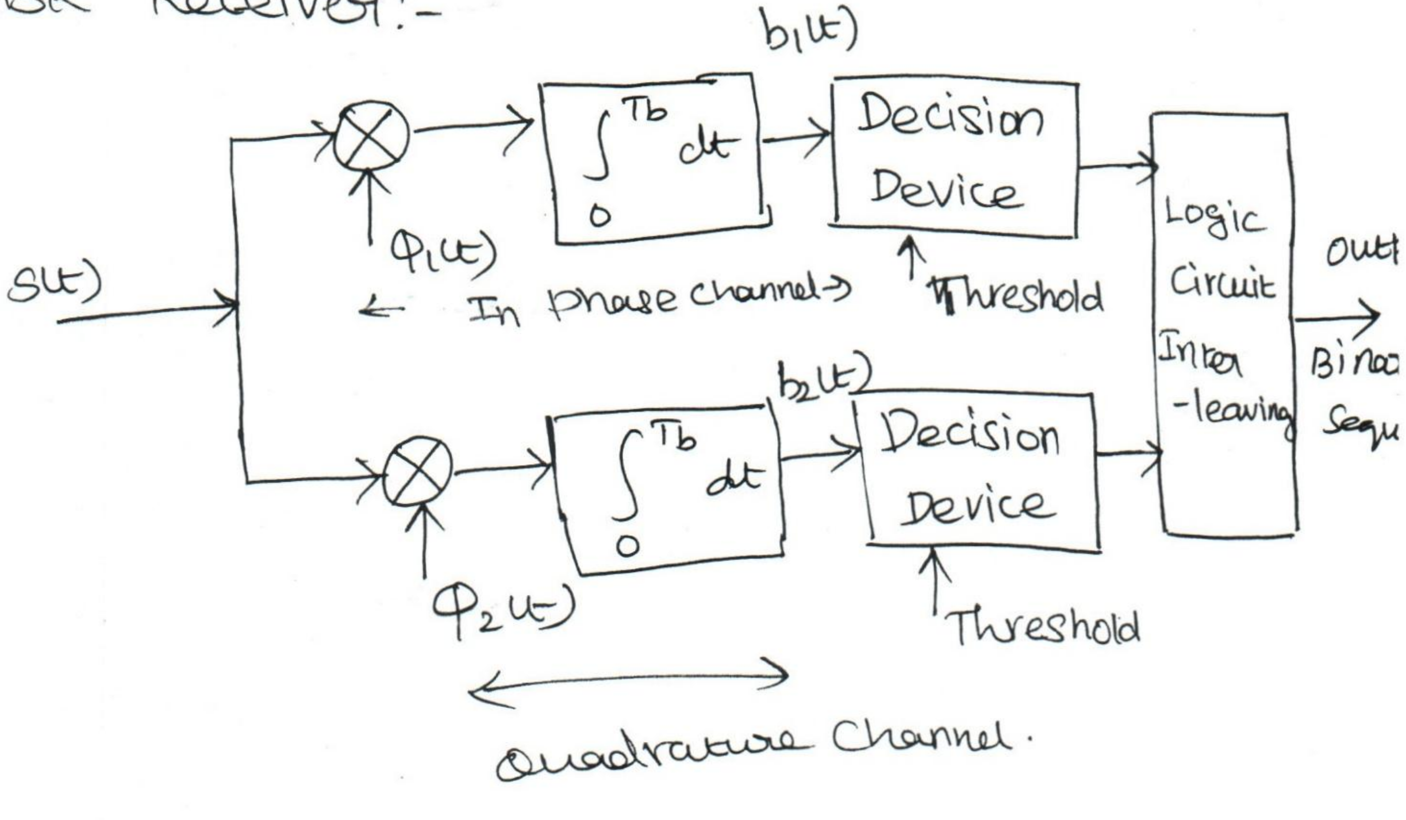


Fig: 21 Block diagram of MSK Receiver

* The received signal is correlated with $\phi_1(t)$ & $\phi_2(t)$. in multiplier.

* The resulting signal is applied to the integrator where integration interval is $2T_b$ sec.

* Integration in quadrature channel is delayed by T_b seconds with respect to in phase channel.

* Decision device used to estimate phase of the signal & send to the interleaver.

* These phase decisions are interleaved to reconstruct original signal.

<u>Waveforms:-</u>

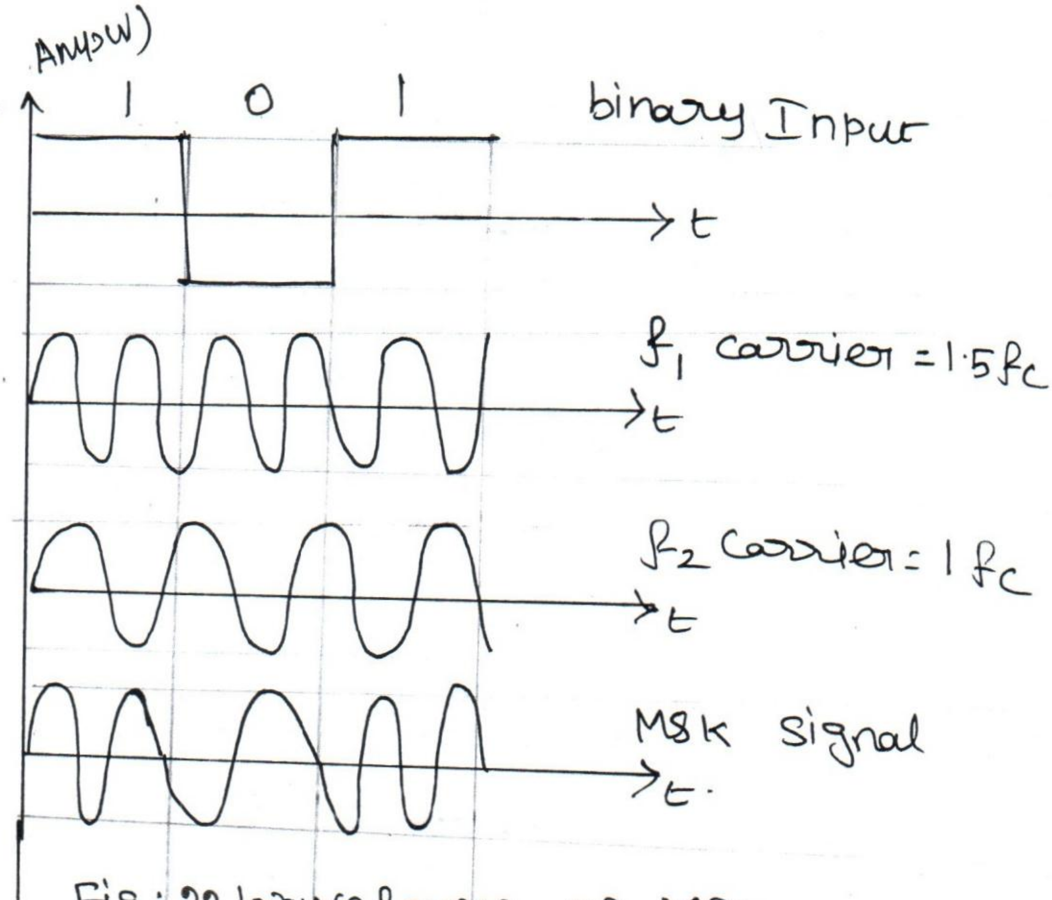


Fig: 22 waveforms of MSK

3.8 GAUSSIAN MINIMUM SHIFT KEYING (GMSK)

Definition:-

GMSK is a modification of MSK with modulation index 0.5.

" Simple binary modulation scheme, derivative of MSK".

Gaussian Filter:-

The word Gaussian refers to the filter called Gaussian filter.

This filter should have a sharp cut off, narrow bandwidth and its impulse response should show no over shoot.

* To reduce ISI & Sideband extension the baseband signal is passed through a Gaussian filter.

* This filter smooths the phase distortion of MSK signal. The result is FM signal like this basic MSK is converted in to GMSK.

3.8.1 GMSK Transmitter

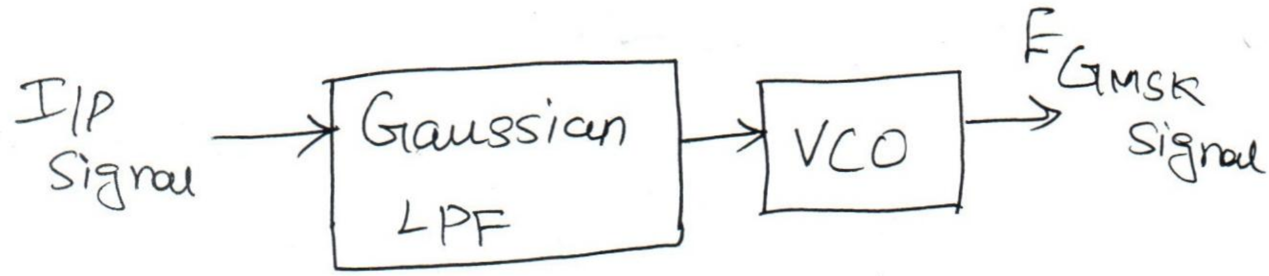


Fig: GMSK Transmitter

* Modulating signal is filtered using gaussian filter. Then it's applied to a frequency modulator where modulation index is 0

Error Probability:-

$$P_e = Q\left\{ \sqrt{\frac{2\gamma E_b}{N_0}} \right\}$$

$\gamma \rightarrow$ constant (0.68)

$E_b \rightarrow$ Transmitted bit energy.

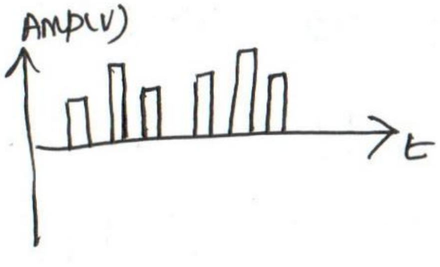
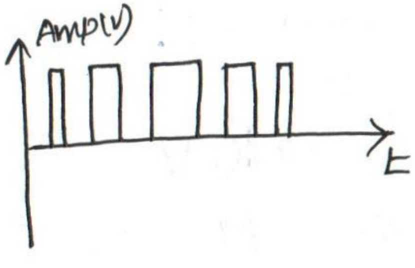
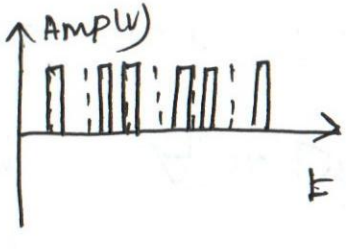
$N_0 \rightarrow$ Noise spectral density.

5 COMPARISION OF DIGITAL PULSE MODULATION SYSTEMS

S.NO	Parameter	PCM	DM	DPCM	ADM
1.	Definition	In PCM, PAM signal is quantized and converted to a digital code. this signal is called as Pulse code Modulated signal. this process called as Pulse code modulation	one bit or Two level version of DPCM. Delta modulation provides a staircase approximation to the oversampled version of message Signal	Difference between two successive Samples are transmitted rather than actual value	step size is varying with respect to changes present in the input Signal.
2.	Number of bits used	4.8 or 16 bits per sample	only one bit per sample	one bit used to encode the Sample	More than one bit.
3.	Noise / Errors	Quantization error	slope overload distortion & granular Noise	Granular Noise & slope overload distortion	Less Granular Noise

(4) Comparison of PAM, PWM and PPM systems:-

COMPARISION OF PAM, PWM AND PPM SYSTEMS

S.NO	PARAMETER	PAM	PWM	PPM
1.	Variable parameter of carrier	Amplitude	width	Position.
2	Power of Transmitter	Varies with amplitude changes	varies with variation in width	Constant.
3.	Transition Bandwidth	$B_T \geq \frac{1}{2\tau}$ $\tau \rightarrow$ width of the pulse	$B_T \geq \frac{1}{2t_r}$ $t_r \rightarrow$ Rise time of the pulse	$B_T \geq \frac{1}{2t_r}$ $t_r \rightarrow$ rise time of the pulse.
4.	Noise interference	Maximum	Minimum	Minimum.
5.	Synchronization	Not necessary	Not necessary	Necessary
6.	Similarity with other Modulation Method	AM	FM	PM
7.	Output waveforms			

S.NO	Parameter	ASK	FSK	2 PSK	QPSK	QAM
4.	Number of bits per symbol (N)	1	1	1	2	1
5.	Possible symbols $M = 2^N$	2	2	2	4	2^N
6.	Detection method	Coherent	Non coherent	Coherent	Coherent	Coherent
7.	Symbol duration T_s	T_b	T_b	T_b	$2T_b$	NT_b
8.	Minimum Bandwidth	f_b	$2(f_b + \Delta f)$	$2f_b$	f_b	$\frac{2f_b}{N}$