Modulation:-

Modulation is defined as the process by which some characteristics of a signal called carrier signal is varied in accordance with the instantaneous values of the message signal.

Types of modulation:-

Modulation is basically contains of two types

(i) Continuous wave:-

The modulation is performed with continous wave (cw) of carrier signal called continuous wave modulation. Amplitude modulation and angle modulation techniques are examples of this method.

(ii) pulse modulation:-

The modulation is performed with pulse type waveform of carrier signal called pulse modulation. pulse Amplitude, pulse width and pulse position modulations are examples of this method.

Need for modulation:-

The message signal or baseband signal is used to modulate a high frequency carrier signal inside the transmitter. After modulation, the modulated signal is transmitted with the help of an antenna which is connected at the output side of the transmitter. This modulated signal then travels down the channel to reach at the input of the receiver.

(i) practicality of Antennas:-

when free space is used as a transmitting medium, information is transmitted and received with the helpof antennas. For efficient radiation and reception the transmitting and receiving antennas must have lengths comparable to a quarter wavelength of the frequency used. For an example, AM broadcasts systems, the maximum audio frequency transmitted from a radio station is of the order of 5KHz. theheight of the antenna required is

$$L = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3\times10^8}{4\times5\times10^3} = 5\text{ K.m.}$$

It is impossible to design 5 K.m. of antennaheight. this antenna height may be reduced by modulation technique and yet effective radiation and reception is achieved.

As an example, if an audio frequency is translated to a radio frequency carrier of frequency 3MHz, the antenna height required would be

$$l = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3\times10^8}{4\times3\times10^6} = 25 \text{ meters}$$

(ii) Interference minimization(or)reduce:-

In radio broadcasting, there are several radio stations. In this case, there is no modulation, all these stations transmit audio or sound signals in the range of 20Hz to 20KHz. Due to this transmission over same range, the programmes of different stations will get mixed or interfere.

In order to seperate various signals, it is necessary to translate them all to different portions of the electromagnetic spectrum (channel); each must be given its own bandwidth commonly known as channel bandwidth. this can be achieved by taking different carrier frequencies for different signal source.

Scanned with CamScanner

Increases the range of Communication:-

At low frequencies radiation is poor and signal gets highly attenuated. Therefore baseband signals cannot be transmitted directly over long distance. Modulation effectively increases the frequency of the signal to be radiated and thus increases the distance over which signals can be transmitted faithfully.

(iv) Reduction of Noise:-

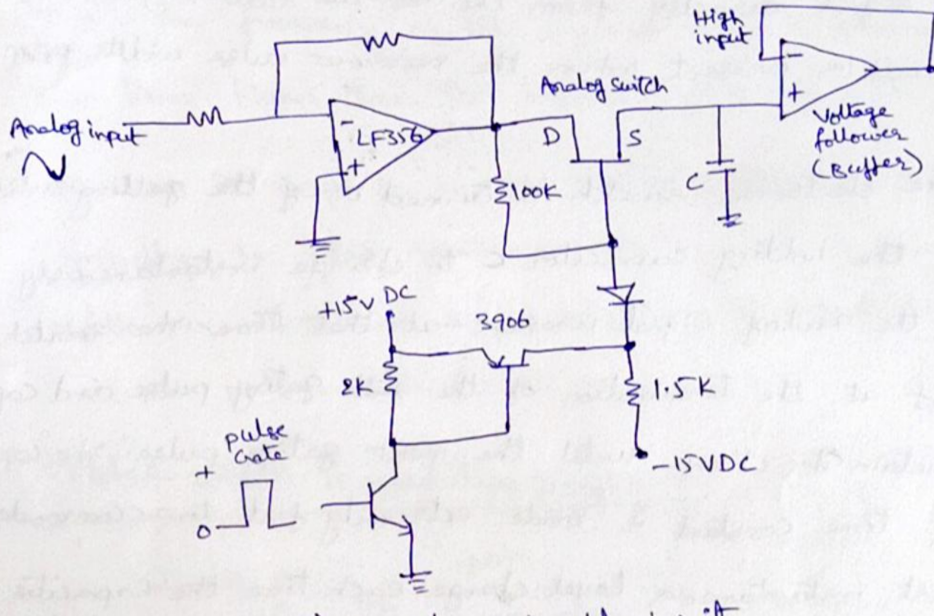
Noise is the major limitation of any Communication. Although noise cannot be eliminated completely, but with the help of several modulation schemes, the effect of noise can be minimized.

PULSE MODULATION

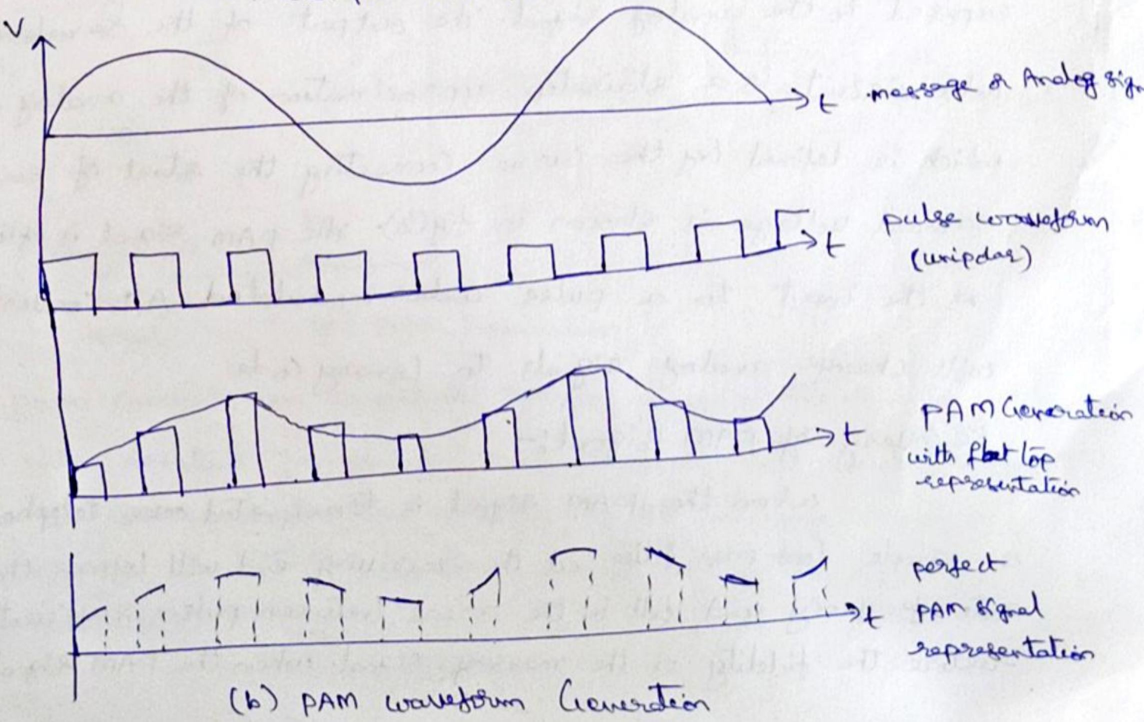
Reference Book:
1. Electronic Communications by Robert J. Schoenbeck
2. Communication Systems by Sanjay Sharma

Pulse Amplitude Modulation (PAM):-

One of the simplest ways to digitize an analog signal is seen when the sine wave message signal is mixed nonlinearly with a low duty cycle square wave. The circuits of class D amplifier and sample and hold circuits are typical PAM with a 20% duty cycle.



(a) Sample and Hold circuit



(b) PAM waveform Generation

Digital circuits perform more efficiently with a ~~flat~~ flat-top square wave signal than with a rounded top. Fig (a) shows a sample and hold circuit that will modify PAM signal taken directly from an analog input and convert it to a flat top PAM signal. However, a square wave pulse will have a wider bandwidth than the original analog signal because of the high harmonic content of the square wave. The input to the driver amplifier is the analog signal directly from the source. However, less quantizing error will be present when the narrow pulse width PAM signal is used.

The electronic switch is turned on by the gating pulse, allowing the holding capacitor C to charge instantaneously to the level of the analog signal present at that time. The switch is turned off at the termination of the ~~gati~~ gating pulse and capacitor C will retain its charge until the next gating pulse. The capacitor's charging time constant is made extremely fast to accommodate an almost instantaneous level charge each time the capacitor is exposed to the analog signal. The output of the sample and hold circuit is a stairstep approximation of the analog signal, which is defined by the curve connecting the start of each sampled voltage is shown in fig(b). The PAM signal is often used as the input to a pulse code ~~modulated~~ modulated A/D converter that will change analog signals to binary code.

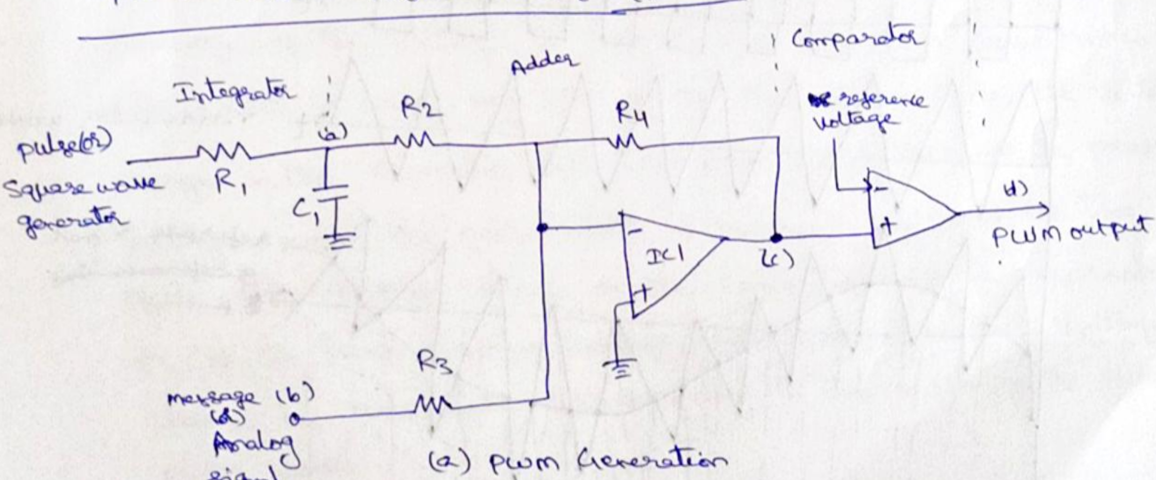
Recovery of PAM signal:-

when the PAM signal is transported over telephone lines, a simple low pass filter at the receiving end will bypass the pulse rate frequency and fill in the areas between pulses sufficiently to restore the fidelity of the message signal. When the PAM signal is used to directly modulate a higher carrier frequency for radio transmission, the AM detector at the receiver will act as the low pass filter to remove the pulse frequency. Again, no fidelity is lost. The only precaution to be observed in the recovery process is to ensure that the low pass filter has a flat frequency response over the entire baseband frequency range and provides sufficient attenuation at the pulse rate frequency.

Note:- The frequency of the square wave is generally selected to be five times higher than the highest frequency in the audio signal.

The PAM is used in audio amplifiers, the PAM signal is applied directly to the speakers with no loss of intelligibility and with no required demodulation of the signal. The speaker will filter out the pulsed frequencies.

Pulse width modulation:- (PWM)

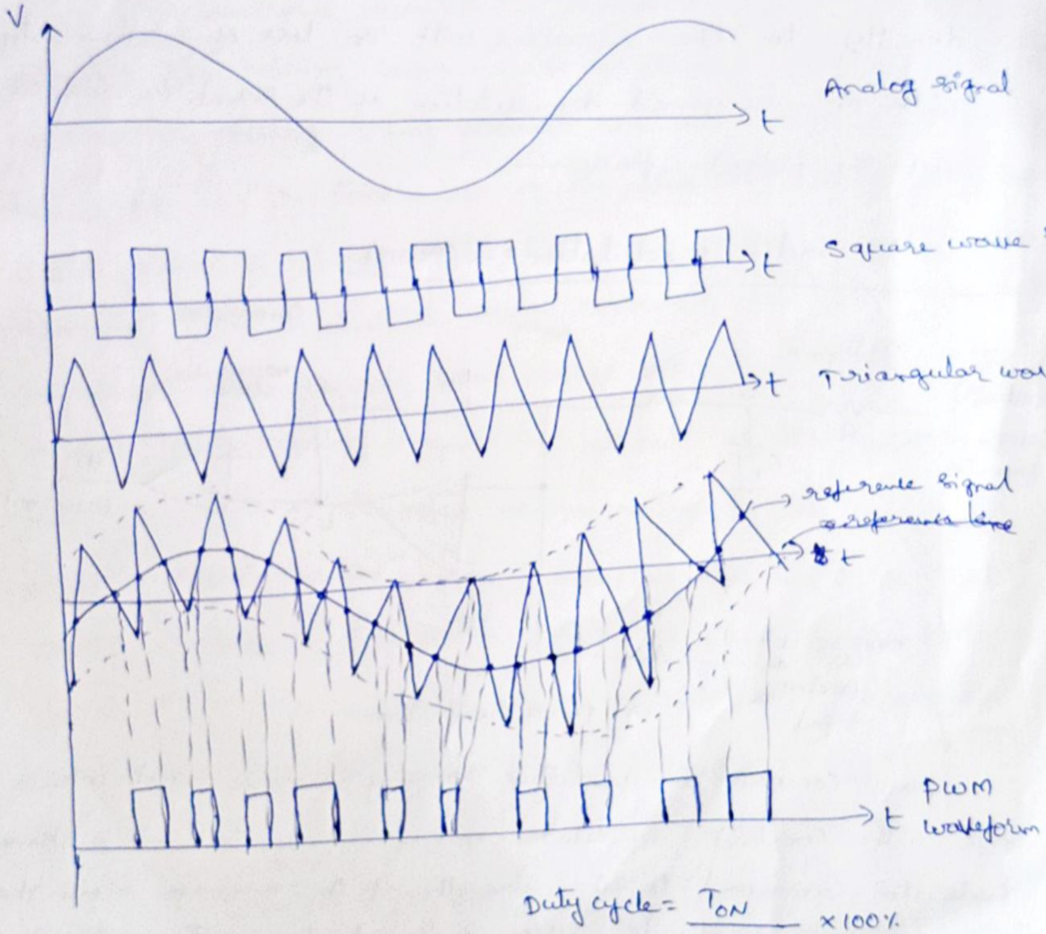


(a) PWM Generation

PWM converts an amplitude varying message signal into a square wave with constant amplitude and frequency, but which changes duty cycle to correspond to the strength of the message signal. The signal that changes amplitude will be referred to as the audio to simplify the description.

The audio signal is mixed with a triangular wave in a linear ad[der] as shown in fig (a). The triangular wave may originate from any function generator, such as 555 timers. The timer output wave is passed through an integrator (low pass filter), which, if properly selected, will change the square wave into a triangular wave.

In the linear mixer (adder), neither of the two input signals will change the basic shape or frequency of the other. The result of two signals added algebraically is that the higher frequency signal will have a new reference line in the shape and size of the lower frequency signal. as shown in fig (b).



$$\text{Duty cycle} = \frac{T_{ON}}{T_{ON} + T_{OFF}} \times 100\%$$

(b) PWM waveform

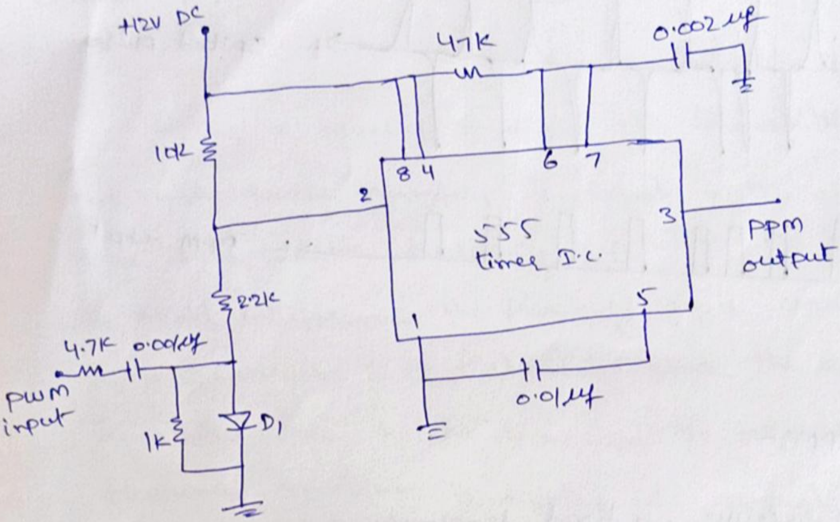
The combined triangular and audio waves are applied to a zero crossing comparator, which will have an output that varies between +Vcc and -Vee, depending on the exact instantaneous level of the input signal. The output of the comparator will have a constant amplitude and fixed frequency (75KHz), but one that has a changing duty cycle. The pwm signal may now be transported to its destination by telephone pairs or it can be used amplitude or frequency modulate another high frequency carrier signal. The idea of double modulation is not new, as you recall from the discussion of stereo transmission. Moreover, double modulation is one of the main principles behind color television.

PWM recovery:-

The pwm signal arrives at its destination by way of telephone lines, the recovery circuit used to decode the original signal is a simple integrator. The charge on the filter capacitor will be the average of the voltage in any cycle of the pwm wave. When the pulse width is wide, say 95% of the time of one cycle, the voltage charge on the capacitor will be approximately 95% of the peak carrier voltage. When the pulse width is narrow, say 5% of the time of one cycle, the voltage charge on the capacitor will be approximately 5% of the peak carrier voltage. The recovered output voltage will change in amplitude corresponding to the width of the pulses in the pwm wave.

Pulse position modulation (PPM):-

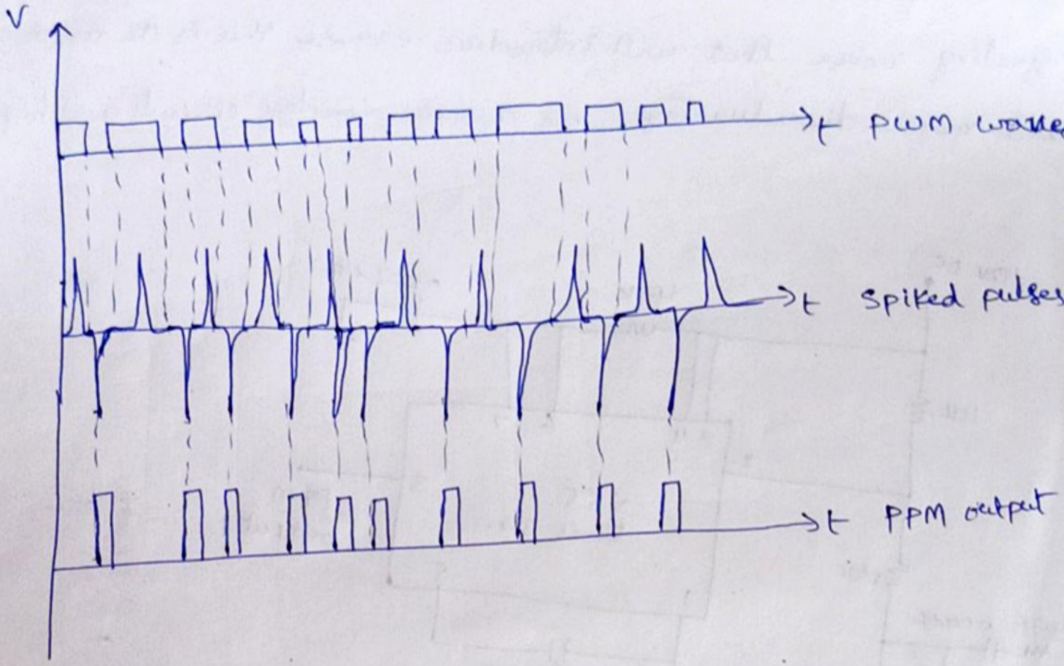
The analog signal is changed to a ~~ppm~~ PWM signal first, then the pwm signal is converted to a PPM signal. This double modulation at the transmitter may seem redundant, but the improvement in noise immunity is well worth the added effort. The reason the original PWM signal is not used becomes apparent when we compare the presence of errors at the receiver under high noise conditions for the three modulations (PAM, PWM and PPM). The PPM transmission is far superior over the other two systems in rejecting noise that will introduce errors. This is the major advantage and major disadvantages are a more complex circuits and higher costs.



(a) PPM Generation

The PWM signal is applied as input to PPM as shown fig(a). The goal here is to develop a square wave pulse of short but fixed duration and then change the pulse position relative to the timing of the cycle. How the signal is treated between the PPM input terminals and the input of the square wave generator will depend on which circuit use for the generator. Fig(a) represents the ~~positive~~ negative going trigger pulse. For positive trigger pulse shaping Schmitt trigger is used.

The PWM signal is first differentiated, changing the square w[ave]
pulses to spiked pulses shown in Fig(b). The negative spikes are
now time related to the trailing edge of the PWM input signal.
pulses will fire the 555 timer, which will generate a 10 µsec output
pulse.



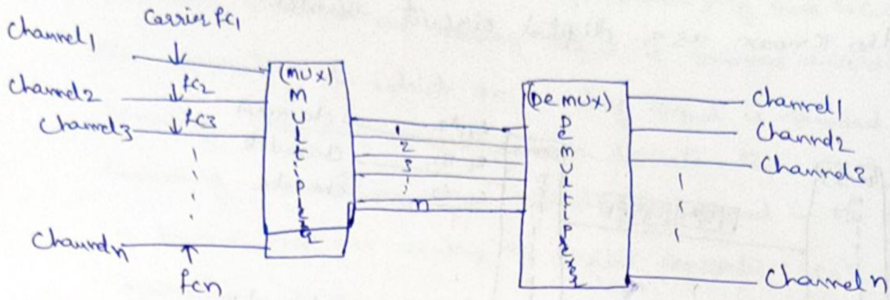
Recovery of PPM:-

At the receiver, a fixed frequency reference pulse is generated from the PPM input signal to activate a flip flop (bistable multivibrator). The PPM signal is also applied to the RESET terminal of the flipflop for shutdown. This recreates the PWM signal, which in turn may be demodulated by a simple low pass filter.

Frequency Division multiplexing:-

multiplexing is used in the cases where the signals of lower bandwidth and the transmitting media is having higher bandwidth.

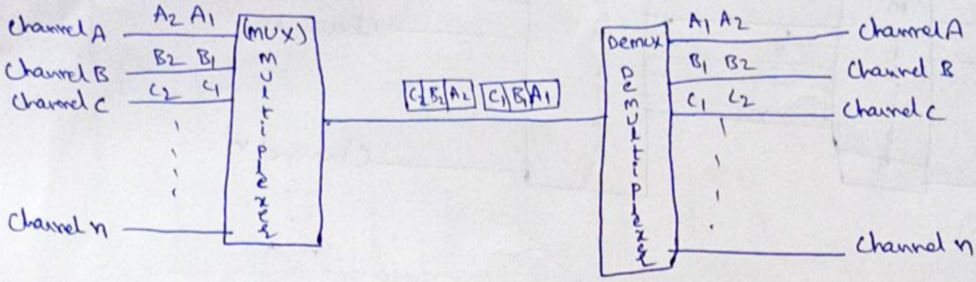
In this case, the possibility of sending a number of signals is more. In this the signals are combined into one and are sent over a link which has greater bandwidth of media than the communicating nodes.



In this a number of signals are transmitted at the same time, and each source transfers its signals in the alloted frequency range. There is a suitable frequency gap between the 2 adjacent signals to avoid interference. The frequency gap is created by applying the different carriers to the channels. Since the signals are transmitted in allotted time so this decreases the probability of collision. The frequency spectrum is divided into several logical channels, in which every user feels that they posses a particular bandwidth. A number of signals are sent simultaneously on the same time allocating seperate frequency band or channel to each signal. It is used in radio and TV transmission. To avoid interference between two successive channels Guardbands are used. This used for transmission of analog signals.

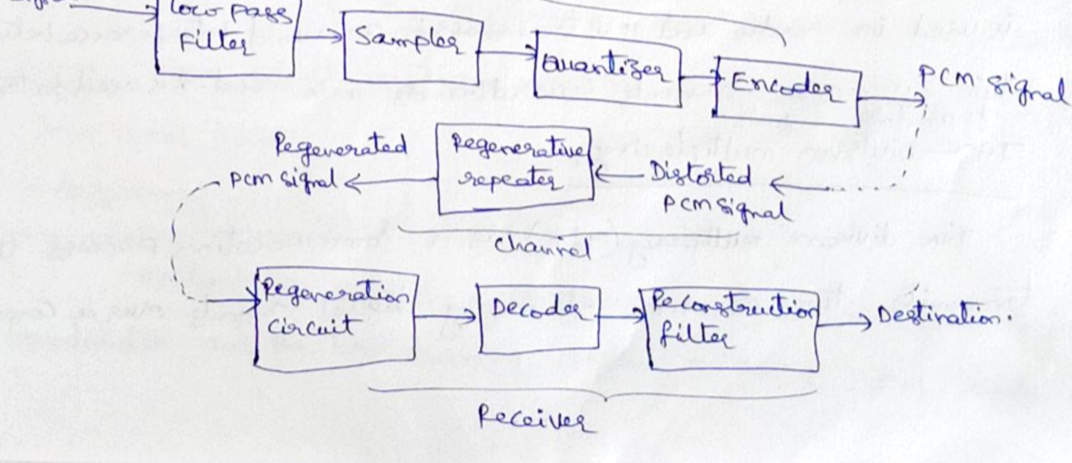
Time Division multiplexing:-

Time division multiplexing (TDM) is a communications process that transmits two or more streaming digital signals over a common channel. In TDM, incoming signals are transmitted over a shared medium and reassembled into their original format after de-multiplexing. Time slot selection is directly proportional to overall system efficiency. TDM is also known as a digital circuit switch.



From the above figure, TDM method is used for transmitting and receiving independent signals over a common signal path by means of synchronized switches (MUX & De-mux) at each end of the transmission line so that each signal appears on the line only a fraction of time in an alternating pattern. It is used when the bitrate of the transmission medium exceeds that of the signal to be transmitted. This type of signal multiplexing was developed in telecommunication systems.

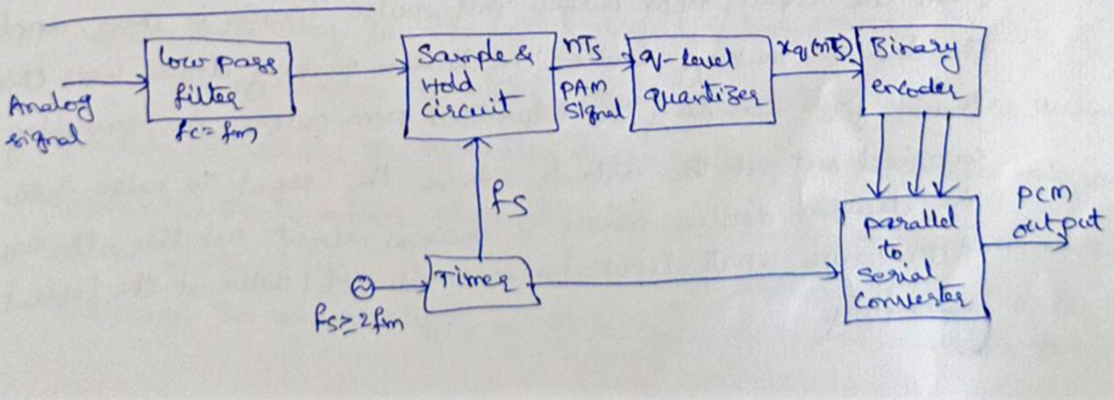
Pulse Code Modulation :- (PCM).



pulse code modulation is known as a digital pulse modulation technique. From the above figure, it consists of three main parts i.e., transmitter, transmission path and receiver. The essential operations in the transmitter of a PCM system are sampling, quantizing and encoding. The sampling is the operation in which an analog signal is sampled according to the sampling is the operation in which an analog signal is sampled according to the sampling theorem resulting in a discrete-time signal. The quantizing and encoding operations are usually performed in the same circuit which is known as an analog to digital converter (ADC), Also, the essential operations in the receiver are regeneration of impaired signals, decoding and demodulation of the train of quantized samples. These operations are usually performed in the same circuit which is known as a digital to analog converter (DAC).

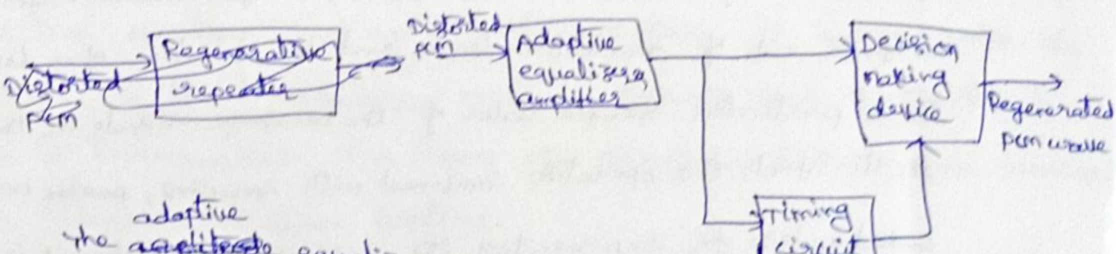
At receiver, regenerative repeaters are used to reconstruct the transmitted sequence of coded pulses in order to combat the accumulated effects of signal distortion and noise. The quantization refers to the use of a finite set of amplitude levels and the selection of a level nearest to a particular sample value of the message signals as the representation for it. Infact, this operation combined with sampling, ~~pealin~~ permits the use of coded pulses for representing the message signal. Thus, it is the combined use of quantizing and coding that distinguishes pulse code modulation from analog modulation techniques.

PCM Transmitter:-



From the above figure, the analog signal ~~x(t)~~ is first passed through low pass filter of cutoff frequency f_m Hz. this low pass filter blocks all the frequency components which are lying above f_m Hz. this means that now the signal $x(t)$ is bandlimited to f_m Hz. the sample and hold circuit then samples this signal at the rate of f_s. Sampling frequency f_s is selected sufficiently above nyquist rate to avoid aliasing effect. the output of sample and hold circuit is denoted by $x(nT_s)$. this signal $x(nT_s)$ is discrete in time and continuous in amplitude. A q- level quantizer compares input $x(nT_s)$ with its fixed digital levels. It assigns any one of the digital level to $x(nT_s)$ which results in minimum distortion or error. this error is called quantization error. thus, output of quantizer is a digital level called $x_q(nT_s)$. Now, the quantized signal level $x_q(nT_s)$ is given binary encoder. this encoder converts input signal to 'n' digits binary word. thus $x_q(nT_s)$ is converted to 'n' bits. this encoder is also known as digitizer.

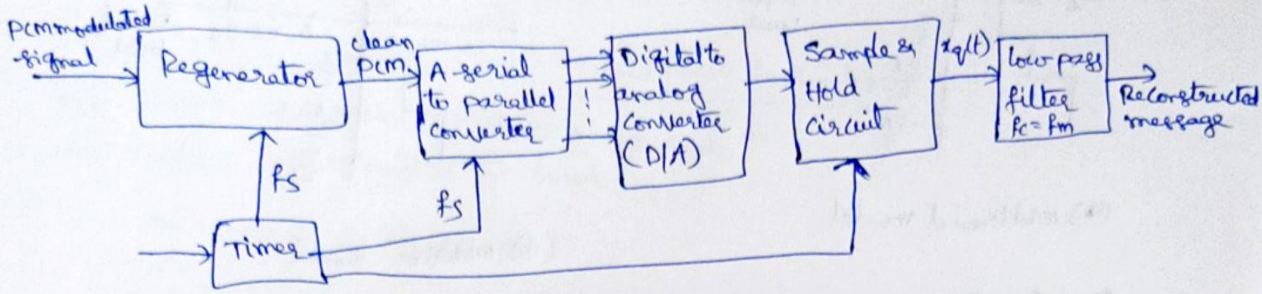
PCM Transmission path:-



The ~~amplitude~~ adaptive equalizer shapes the distorted PCM wave so as to compensate for the effects of amplitude and phase distortions. The timing circuit produces a periodic pulse train which is derived from the input PCM pulses. this pulse train is then applied to the decision making device. The decision making device uses this pulse train for sampling the equalized PCM pulses. The sampling is carried out at the instants where the signal to noise ratio is maximum. The decision device makes a decision about whether the equalized PCM wave at its input has a 0 value or 1 value at the instant of sampling.

such a decision is made by comparing equalized PCM with a reference level called decision threshold. At the output of the decision device, we get a clean PCM signal without any trace of noise.

PCM Receiver:-



The regenerator at the start of PCM receiver reshapes the pulse and removes the noise. The signal is then converted to parallel digital words for each sample. Now, the digital word is converted to its analog value denoted as $x_q(t)$ with the help of a sample and hold circuit. This signal, at the output of sample and hold circuit, is allowed to pass through a lowpass reconstruction filter to get the appropriate original message signal. ~~dep~~

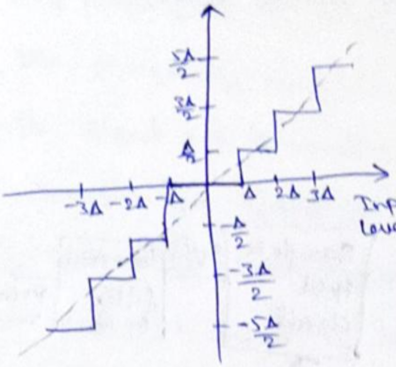
Quantizer:-

The quantization process can be classified into two types as
(i) uniform quantization
(ii) Non- uniform quantization

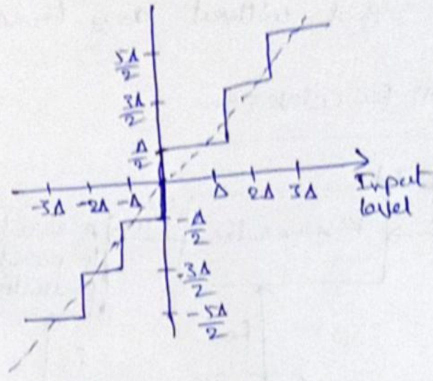
A uniform quantizer is that type of quantizer in which the step size remains same throughout the input range.

A non uniform quantizer is that type of quantizer in which the step-size varies according to the input signal values.

In a uniform quantizer, the representation levels are uniformly spaced, otherwise, the quantizer is non uniform. Let us consider only uniform quantizers, the quantizer characteristics can also be midtread or midrise. In the midtread type, the origin lies in the middle of a tread of the staircase. In midrise type, in which the origin lies in the middle of a rising part of the stair case. the both models are symmetric about the



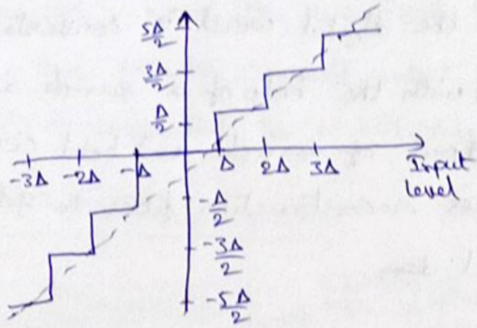
(a) midtread model



(b) midrise model

Quantization error may be expressed as

$$\varepsilon = x_q(nT_s) - x(nT_s)$$



(c) Non uniform quantization

the probability density function (PDF) for quantization error defined as

$$f_\varepsilon(\varepsilon) = \begin{cases} 0 & \varepsilon \leq \frac{\Delta}{2} \\ \frac{1}{\Delta} & -\frac{\Delta}{2} < \varepsilon \leq \frac{\Delta}{2} \\ 0 & \varepsilon > \frac{\Delta}{2} \end{cases}$$

the mean square value of ε is

$$E[\varepsilon^2] = \int_{-\Delta/2}^{\Delta/2} \varepsilon^2 \left(\frac{1}{\Delta}\right) d\varepsilon \Rightarrow \frac{1}{\Delta}\left[\frac{\varepsilon^3}{3}\right]_{-\Delta/2}^{\Delta/2}$$

$$E[\varepsilon^2] = \frac{\Delta^2}{12}$$

signal to quantization noise ratio of the quantizer is

$$\frac{S}{N} = \frac{\text{Signal power (normalised)}}{\text{Noise power (normalised)}}$$

Noise power = $\frac{V^2_{noise}}{R}$

V^2_{noise} = mean square value = $\frac{\Delta^2}{12}$

Noise power (normalized) = $\frac{\Delta^2}{12}/R$

$= \frac{\Delta^2}{12}$ [$\because$ let $R = 1\Omega$]

$\therefore$ signal to quantization noise = $\frac{\text{Signal power (normalized)}}{\Delta^2/12}$

But signal to Noise ratio (SNR) of PCM is $(1.8+6n)$ db.

The number of quantization levels $q = 2^n$

$n \rightarrow$ number of bits | sample

Signaling rate (or) the number of bits per second is $r = n f_s$

$f_s \rightarrow$ number of samples | sec.

as of the nyquist rate $f_s \geq 2 f_m$

Bandwidth of PCM = $\frac{1}{2} r$

$= \frac{1}{2} n f_s$ [$\because r = n f_s$]

$= \frac{1}{2} n \cdot 2 \cdot f_m$ [$\because f_s = 2 f_m$]

Bandwidth of PCM $= n f_m$

signal to quantization Noise ratio (SQNR) for normalized values of power P and amplitude of input x(t) will be $(S/N) dB \leq (4.8 + 6n) dB$

Companding in PCM:-

Companding is a non-uniform quantization. It is required to be implemented to improve the signal to quantization noise ratio of weak signals. Companding is a term derived from two words compression and expansion.

Ideally, we need a linear compressor characteristics for small amplitude of the input signal and a logarithmic characteristic elsewhere there are two types of companders are available

(i) μ-law companding

(ii) A-law companding

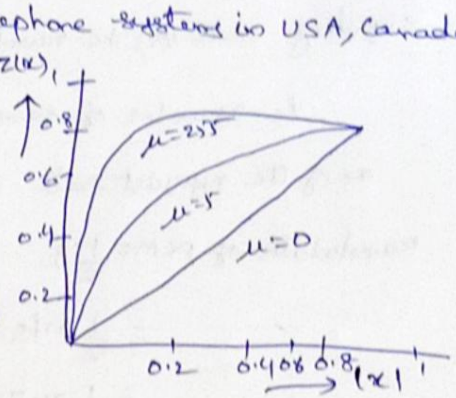
(i) μ-law Companding:-

In this the compressor characteristics is continuous. It is approximately

linear for smaller values of input levels and logarithmic for high in[put]
levels. The μ-law compressor characteristic is expressed as

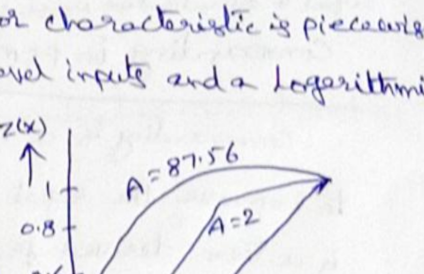
$$z(x) = (\operatorname{sgn} x) \frac{\ln(1+\mu|x|)}{\ln(1+\mu)} \qquad 0 \le |x| \le 1.$$

$|x|$ represents the normalised value of input with respect to the maximum value. sgnx term represents ±1 i.e. positive and negative values of input and output. The practically used value of μ is 255. It may be noted that the characteristic corresponding to $\mu=0$ corresponds to the uniform quantisation. The μ-law companding is used for speech and music signals. It is also used for PCM telephone systems in USA, Canada & Japan.
~~the~~ From the figure, the variations of signal to quantization noise ratio with respect to signal level, with and without companding. It is obvious that SNR is almost constant at all the signal levels when companding is used.



(ii) A-law Companding:-

In the A-law companding, the compressor characteristic is piecewise, madeup of a linear segment for low level inputs and a logarithmic segment for high level inputs.
From the figure, A=1 we observe that the characteristic is linear which corresponds to a uniform quantization. The practically used value of A is 87.56. The A-law companding is used for PCM telephone systems in Europe.



The A-law compressor characteristic is expressed as

$$z(x) = \begin{cases} \operatorname{sgn}(x) \dfrac{A|x|}{1+\ln(A)} & \text{for } 0 \le |x| \le 1 \\ \operatorname{sgn}(x) \dfrac{1+\ln[A|x|]}{1+\ln(A)} & \text{for } \dfrac{1}{A} \le |x| \le 1 \end{cases}$$

Advantages of PCM:-

1. It provides high noise immunity
2. It is possible to store the PCM signal due to its digital nature

Applications:-

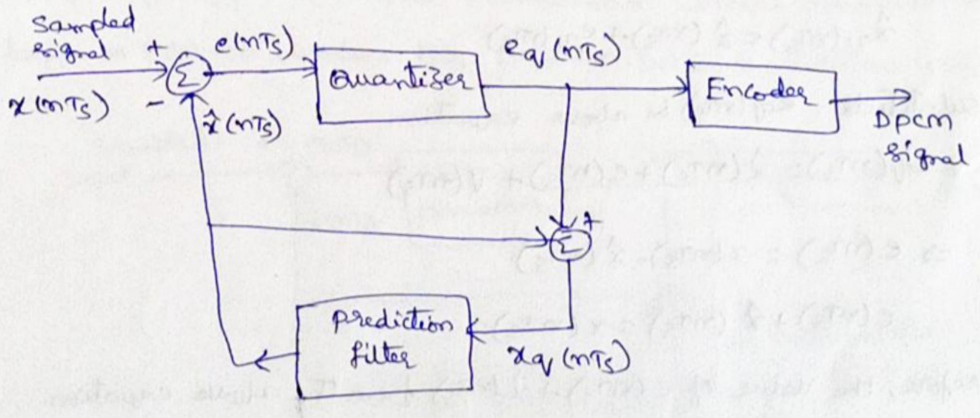
1. with the advent of fiber optic cables, PCM is used in telephony
2. In space communication, space craft transmits signals to earth because of high noise immunity, only PCM systems can be used.

Drawbacks of PCM:-

1. PCM requires a large bandwidth as compared to other systems
2. Encoding, decoding and quantizing circuitry of PCM is complex.

Differential pulse code modulation (DPCM):-

In PCM, some of the samples of a signal are highly correlated with each other. This is due to the fact that any signal doesnot change fast. The adjacent samples of the signal carry the same information with a little differerce. when these samples are encoded by a standard PCM system, the resulting encoded signal contains some redundant information. This redundant information is overcome in DPCM.



(a) Transmitter

Infact the DPCM works on the principle of prediction. The value of the present sample is predicted from the past samples. The prediction may not be exact but it is, very close to the actual sample value. From the above figure, the transmitter of DPCM system, the sampled signal is denoted by $x(nT_s)$ and the predicted signal is denoted by the comparator finds out the difference between the actual sample value $x(nT_s)$ and predicated sample value $\hat{x}(nT_s)$. This is known as prediction error and it is denoted by $e(nT_s)$. It is defined as

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

Thus, error is the difference between unquantized input sample $x(nT_s)$ and prediction of it $\hat{x}(nT_s)$. The predicted value is produced by using a prediction filter. The quantizer output signal gap $e_q(nT_s)$ and previous prediction is added and given as input to the prediction filter. This signal is called $x_q(nT_s)$. This makes the prediction more and more close to the actual sampled signal. We can observe that the quantised error signal $e_q(nT_s)$ is very small and can be encoded by using small number of bits thus number of bits per sample are reduced in DPCM.

The quantizer output can be

$$e_q(nT_s) = e(nT_s) + q(nT_s)$$

$q(nT_s) \rightarrow$ the prediction filter input $x_q(nT_s)$ is obtained by sum $\hat{x}(nT_s)$ and quantiser output.

$$\hat{x}_q(nT_s) = \hat{x}(nT_s) + e_q(nT_s)$$

Substitute $e_q(nT_s)$ in above equation

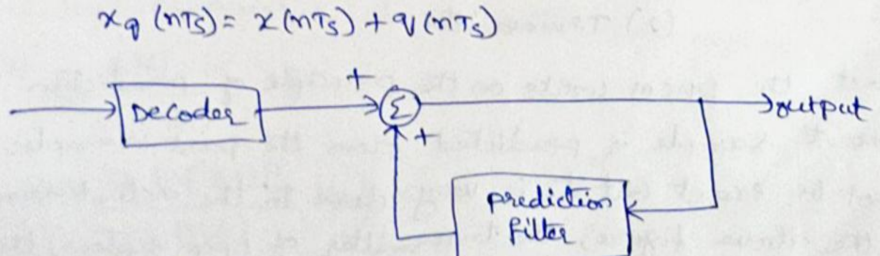
$$x_q(nT_s) = \hat{x}(nT_s) + e(nT_s) + q(nT_s)$$

$$\Rightarrow e(nT_s) = x(nT_s) - \hat{x}(nT_s)$$

$$e(nT_s) + \hat{x}(nT_s) = x(nT_s)$$

Therefore, the value of $e(nT_s) + \hat{x}(nT_s)$ from the above equation

$$x_q(nT_s) = x(nT_s) + q(nT_s)$$



(b) Receiver

From the above figure, the decoder first reconstructs the quantized error signal from incoming binary signal. The prediction filter output and quantised error signals are summed up to give the quantized version of the original signal. Thus the signal at the receiver differs from actual signal by quantisation error $q(nT_s)$, which is introduced permanently in the ~~reconstructed~~ reconstructed signal.

The prediction gain G_p is defined as

$$G_p = \frac{\sigma_x^2}{\sigma_E^2}$$

$\sigma_x^2 \rightarrow$ Variance of original input signal

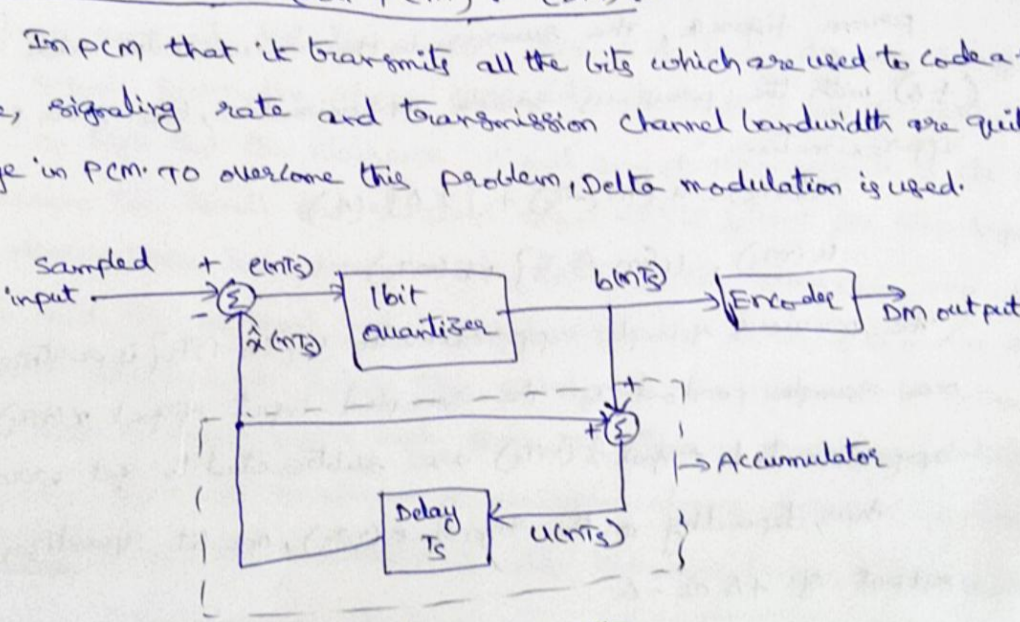
$\sigma_E^2 \rightarrow$ Variance of prediction error.

Advantages:-

1. This will require less number of quantisation levels and hence less number of bits to represent them.
2. Signaling rate and bandwidth of DPCM system will be less than that of PCM.

Delta modulation:- (1bit PCM) :- (DM) :-

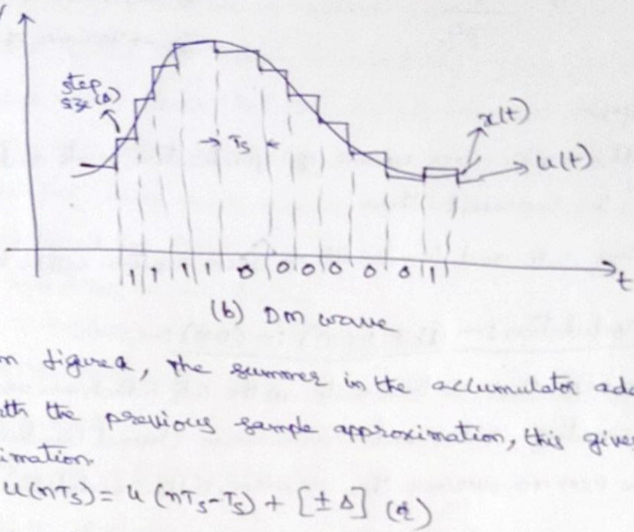
In PCM that it transmits all the bits which are used to code a sample. Hence, signaling rate and transmission channel bandwidth are quite large in PCM. To overcome this problem, Delta modulation is used.



(a) Transmitter

Delta modulation transmits only one bit per sample. Here, the present sample value is compared with the previous sample value and this

result whether the amplitude is increased or decreased is transm[itted].
Input signal $x(t)$ is approximated to step signal by the delta modu[lator]
this stepsize is kept fixed. The difference between the input signal $x(t)$
and staircase approximated signal is confined to two levels $+\Delta$ and $-\Delta$.
Now, if the difference is positive, then approximated signal is
increased by one step i.e. '$+\Delta$'. If the difference is negative, then approximate[d]
signal is reduced by '$-\Delta$'. When the stepsize is reduced, '0' is transmitte[d]
and if the step is increased 1 is transmitted shown in fig(b)



(b) DM wave

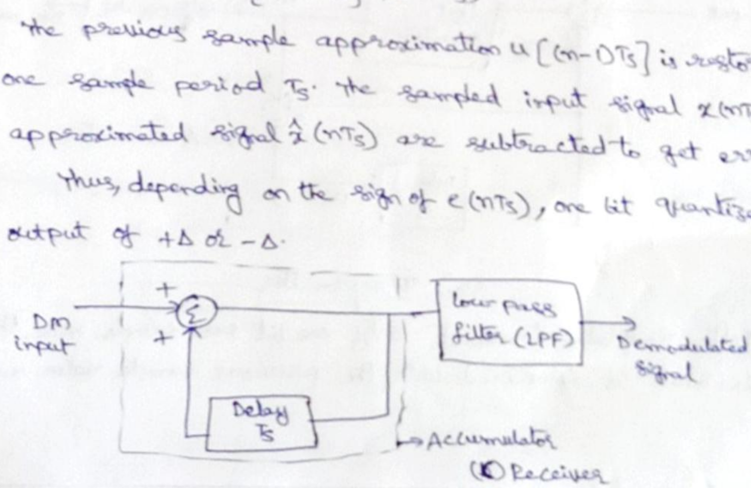
From figure a, the summer in the accumulator adds quantizer output ($\pm\Delta$) with the previous sample approximation, this gives present sample approximation

$$u(nT_s) = u(nT_s - T_s) + [\pm\Delta] \quad (a)$$

$$u(nT_s) = u[(n-1)T_s] + b(nT_s)$$

The previous sample approximation $u[(n-1)T_s]$ is restored by delaying one sample period T_s. The sampled input signal $x(nT_s)$ and staircase approximated signal $\hat{x}(nT_s)$ are subtracted to get error signal $e(nT_s)$.

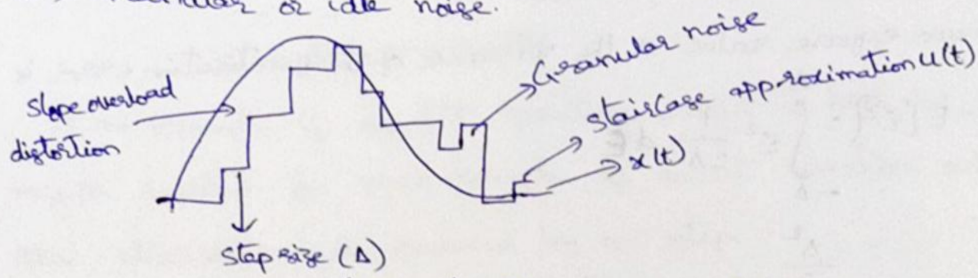
Thus, depending on the sign of $e(nT_s)$, one bit quantizer generates an output of $+\Delta$ or $-\Delta$.



(c) Receiver

From the figure (c), the accumulator and low pass filter (LPF) are used. The accumulator generates the staircase approximated signal output and is delayed by one sampling period T_s. It is then added to the input signal. If input is binary 1 then it adds $+\Delta$ to the previous output (which is delayed). If input is binary '0' then one step Δ is subtracted from the delayed signal. Also, the LPF has the cutoff frequency equal to highest frequency in $x(t)$. This LPF smoothens the staircase to reconstruct original message signal $x(t)$.

The DM has two major drawbacks as
(i) Slope overload distortion
(ii) Granular or idle noise.



(d) Quantization errors in DM

Slope overload distortion arises because of large dynamic range of the input signal. From the above figure (d), the rate of rise of input signal $x(t)$ is so high that the staircase signal cannot approximate it, the stepsize 'Δ' becomes too small for staircase signal $u(t)$ to follow the step segment of $x(t)$. Hence, there is a large error between the staircase approximated signal and the original input signal $x(t)$. This error is known as slope overload distortion. To reduce this error, the stepsize must be increased when slope of signal $x(t)$ is high. Since the stepsize of delta modulator remains fixed, its maximum or minimum slopes occur along straight lines. Therefore, this modulator is also known as linear Delta modulator.

Granular or ideal noise occurs when the stepsize is too large compared to small variations in the input signal. This means that for very small variations in the input signal, the staircase signal is changed by large amount (Δ) because of large stepsize. From the above figure (d), when the input signal is almost flat, the staircase signal $u(t)$

keeps on oscillating by $\pm\Delta$ around the signal. The error between the and approximated signal is called granular noise. The solution to the problem is to make step size small.

Signaling rate = Number of samples/sec × Number of bits/sec

$$= f_s \times 1 = f_s$$

The channel bandwidth for the delta modulation system is reduced to a great extent as compared to that for the PCM system.

Quantization error for DM is

$$f_\varepsilon(\varepsilon) = \begin{cases} \frac{1}{2\Delta} & -\Delta \le f_\varepsilon(\varepsilon) \le +\Delta \\ 0 & \text{otherwise} \end{cases}$$

The mean square value or the variance of the quantization error is

$$E[\varepsilon^2] = \int_{-\Delta}^{\Delta} \varepsilon^2 \frac{1}{2\Delta} d\varepsilon$$

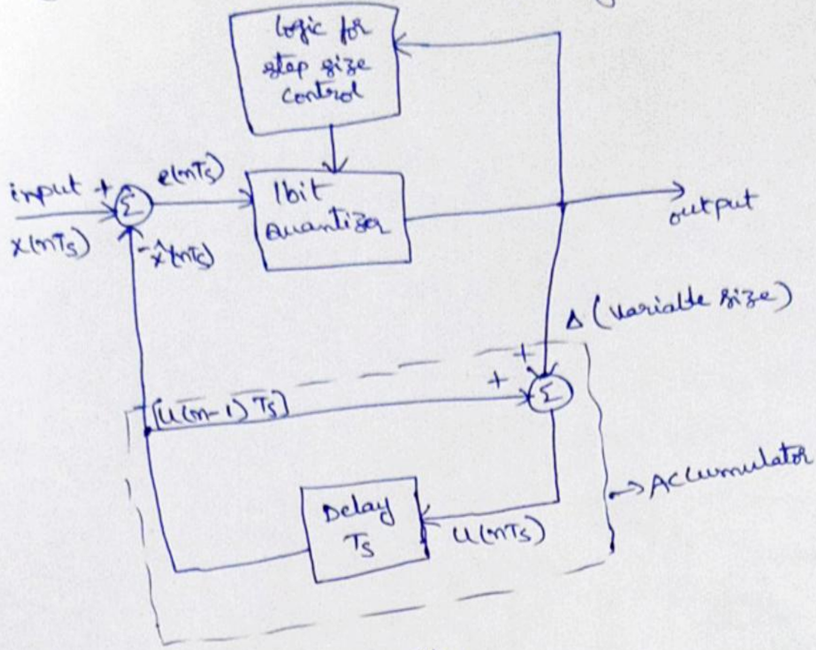
$$= \frac{\Delta^2}{3}$$

Normalised quantization noise power $N_q = \frac{\varepsilon^2}{1} = \frac{\Delta^2}{3}$

Adaptive Delta Modulation:-

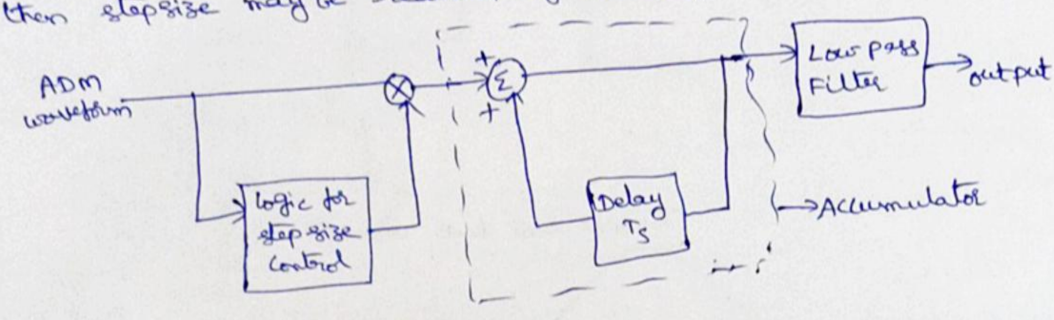
To overcome the quantization errors due to slope overload and granular noise, the stepsize (Δ) is made adaptive to variations in the input signal $x(t)$. Particularly in the steep segment of the signal $x(t)$, the stepsize is increased. Also, if the input is varying slowly, the stepsize is reduced. Then, this method is known as Adaptive Delta Modulation (ADM). The adaptive delta modulators can take continuous changes in stepsize or discrete changes in stepsize.

The logic for step size control is added to the Delta modulator and is shown in figure (a) and this figure is the transmitter of the Adaptive delta modulation. The stepsize is increases or decreases according to a specified rule depending on one bit quantizer output.



(a) Transmitter

As an Example, if one bit quantizer output is high or 1, then step size may be doubled for next sample. If one bit quantizer output is low, then stepsize may be reduced by one step.



(b) Receiver

In the receiver of adaptive delta modulation there are two portions as shown in Figure(b). The first portion produces the step size from each incoming bit. Exactly the same process is followed as that in transmitter. The previous input and present input decides the step size. It is then applied to an accumulator which builds up staircase waveform. The low pass filter then smoothens out the staircase waveform to reconstruct the original signal.

The bandwidth of ADM is better than delta modulation.