UNIT-II

Pulse modulation

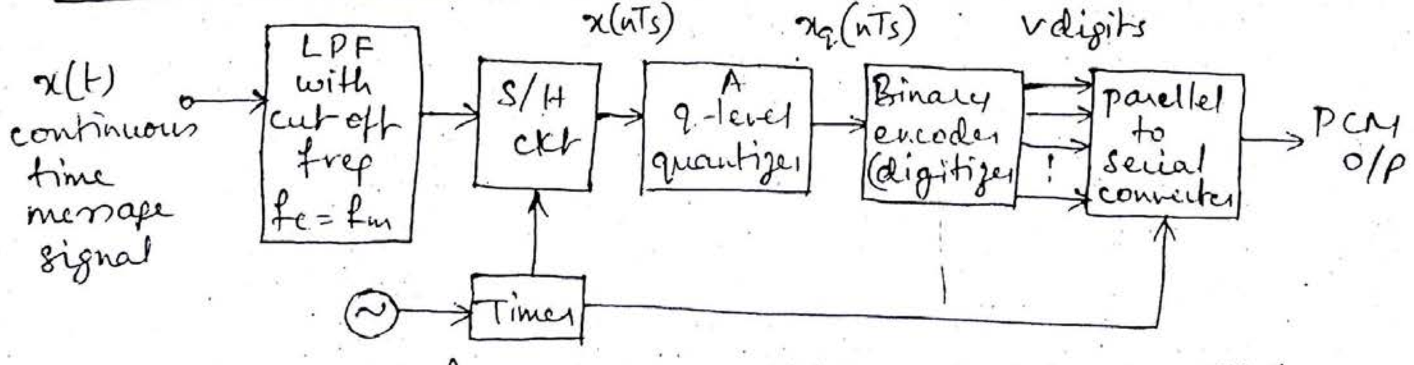
Classified as pulse analog modulation and pulse digital modulation. Pulse digital modulation is used for digital coding. In Pulse analog modulation only time is expressed in digital form. In Pulse digital modulation time & pulse parameter (amplitude) are expressed in digital form.

The simplest form of pulse digital modulation is PCM. In PCM the message signal is first sampled and then amplit of each sample is rounded off to the nearest quantization level (integer). Hence both time and amplitude are in discrete form.

Pulse Code modulation

PCM is known as digital pulse modulation technique. PCM o/p is in coded digital form of pulses of constant amplitude, width and position. The information is transmitt in the form of code words. PCM system consists of a PC encoder (transmitter) and PCM decoder (receiver).

PCM generator or transmitter



The signal is first passed through the low pass filter o cut off frequency f_m Hz. The LPF blocks all freq component which lie above f_m Hz.

This means that the signal $x(t)$ is bandlimited to f_m Hz. The sample and hold ckt then samples this signal at the rate f_s. The sampling frequency f_s is selected sufficiently above Nyquist rate to avoid aliasing. $f_s \geq 2f_m$.
The o/p of sample and hold ckt is denoted $x(nTs)$ which is discrete in time and continuous in amplitude.

A q-level quantizer compares input $x(nTs)$ with its fixed digital levels. It then assigns any one of the digital level to $x(nTs)$ which results in minimum distortion or error called quantization error. Thus o/p of quantizer is a digital level called $x_q(nTs)$.

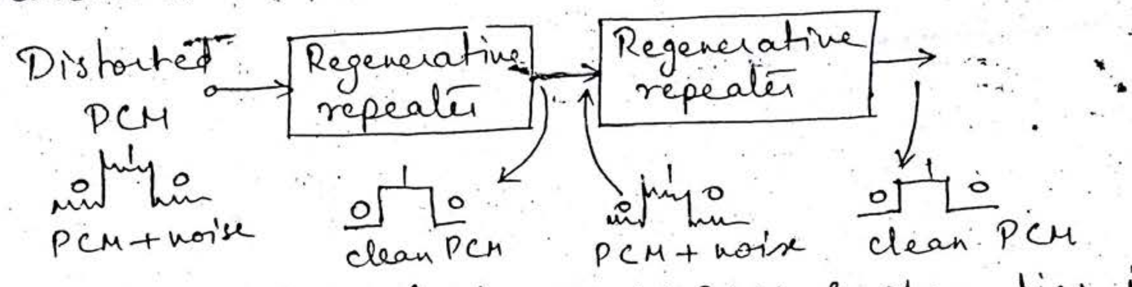
Now the quantized signal $x_q(nTs)$ is given to binary encoder which converts input signal to 'v' binary digits. Thus $x_q(nTs)$ is converted to 'v' binary bits. It is not possible to transmit each bit of binary word separately on a transmission line, therefore 'v' binary digits are converted to a serial bit stream to generate single base band signal.

The parallel to serial converter does this job. The o/p of a PCM generator is thus a single baseband signal of binary bits. The oscillator generates clock for S/H ckt and parallel to serial converter.

The S/H, quantizer and encoder combinely form an analog to digital converter.

PCM Transmission path

The path between the PCM transmitter and the PC receiver over which PCM signal travels is called PC transmission path.



The important feature of PCM system lies in its ability to control the effects of distortion and noise, when PCM wave travels on the channel, PCM accomplis this by using a chain of regenerative repeaters which are spaced close enough to each other on the path.

PCM receiver

Fig (a) shows the block diagram of a PCM receive and fig(b) shows the reconstructed signal. The regenerator at the start of PCM receiver reshapes the pulses and removes the noise. This signal is then conver to parallel digital words for each sample.

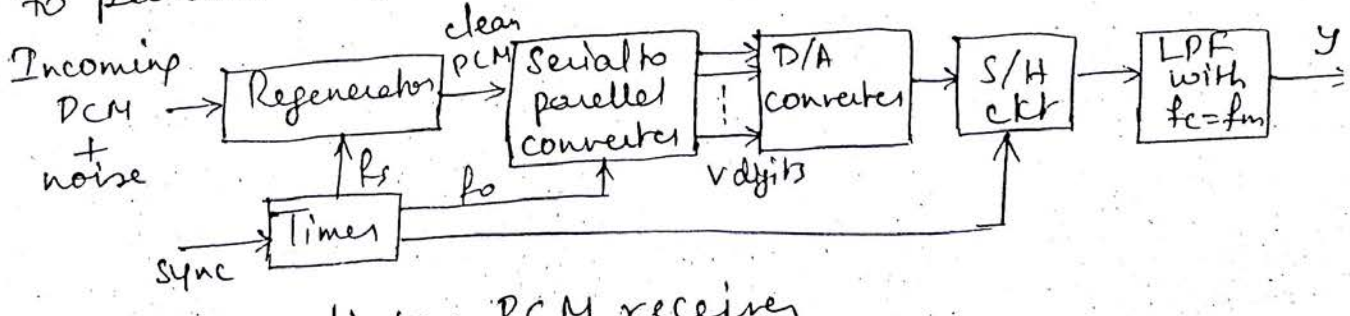


fig (a) PCM receiver

Now the digital word is converted to analog form denoted as $x_q(t)$ with the help of S/H circuit.

This signal at o/p of S/H ckt is allowed to pass through a low pass filter to get the appropiate signal.

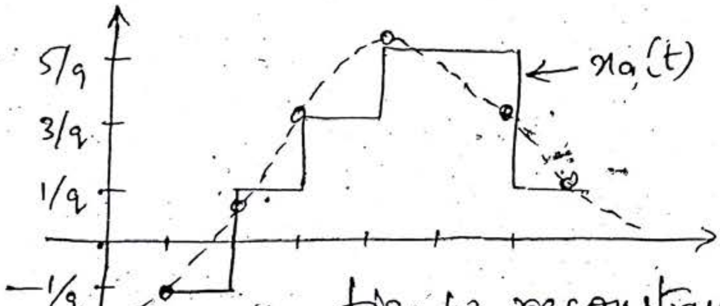


fig (b) reconstructed waveform

As shown in the reconstructed signal of fig (b) it is impossible to reconstruct exact original signal x(t) because of permanent quantization error introduced during quantization of the transmitter.

This quantization error can be reduced by increasing the binary levels.

<u>Quantizer</u>

As discussed previously, a q-level quantizer compares the discrete time input $x(nT_s)$ with its fixed digital levels. It assigns any one of the digital level to $x(nT_s)$ with its fixed digital levels, which results in minimum distortion or error.

<u>classification of quantization</u>

Quantization
- Uniform quantization
  - midtread type
  - mid riser type
  - biased type
- Non uniform quantization

Uniform Quantizer

A quantizer is called a uniform quantizer if the step size remains constant throughout the range.

Types of Uniform quantizers

(i) Mid tread quantizer

The transfer characteristics of a mid-tread quantizer is shown in fig (a). When I/p is between $-\delta/2$ to $+\delta/2$ the quantizer o/p is zero.

For $-\delta/2 \le x(nT_s) < \delta/2$, $x_q(nT_s) = 0$

$\delta/2 \le x(nT_s) < 3\delta/2$, $x_q(nT_s) = \delta$ (Similarly for other levels also)

where δ = step size of the quantizer.

Fig (b) shows the quantization error of mid-tread quantizer. Error is given as

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

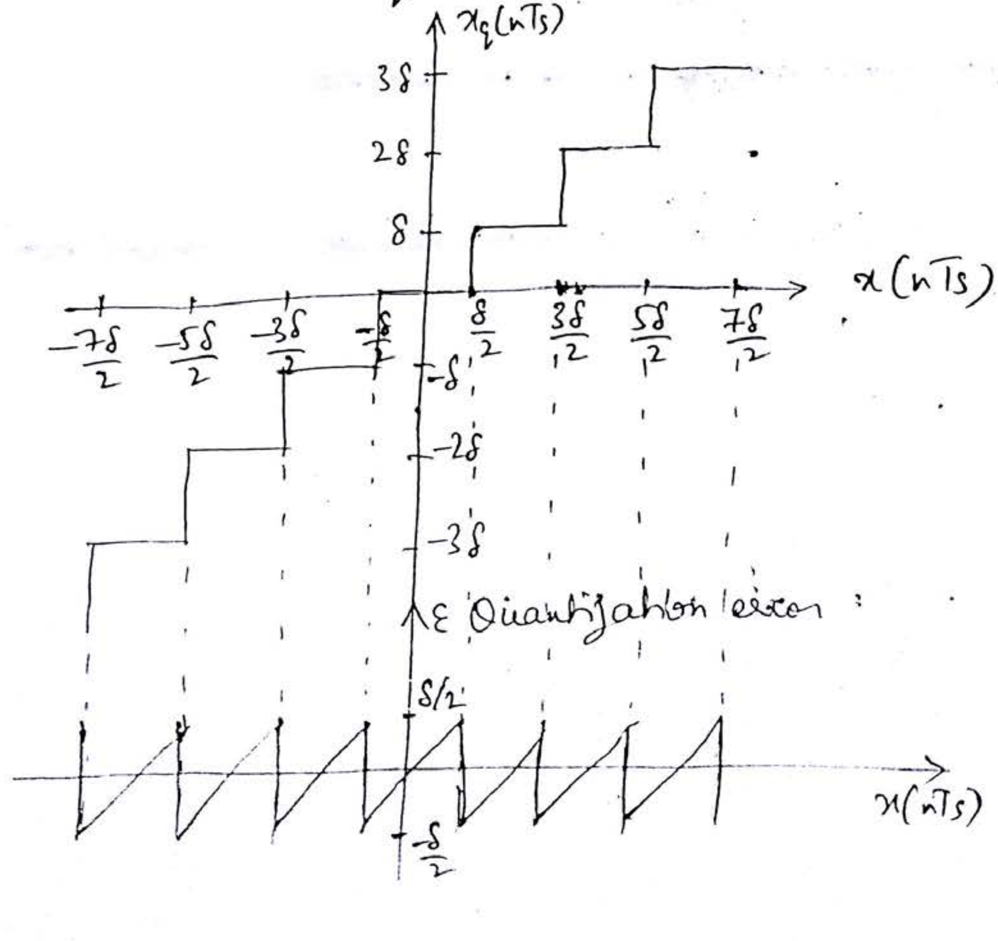


fig (a) Quantizer o/p

fig (b) quantization error

when $x(nT_s) = 0$, $x_q(nT_s) = 0$, hence error is zero at the origin.

when $x(nT_s)$ is between $\delta/2$ and $3\delta/2$, $x_q(nT_s) = \delta$

error $\varepsilon = \delta - \frac{\delta}{2} = \frac{\delta}{2}$

$\delta - \frac{3\delta}{2} = -\frac{\delta}{2}$

Thus the quantization error lies between $-\frac{\delta}{2}$ and $+\frac{\delta}{2}$

Maximum quantization error

$$\varepsilon_{max} = \left|\frac{\delta}{2}\right|$$

(ii) Midriser quantizer

The transfer characteristics is shown. When I/p is between 0 and δ, o/p is $\frac{\delta}{2}$. when I/p is between 0 and $-\delta$, o/p is $-\frac{\delta}{2}$

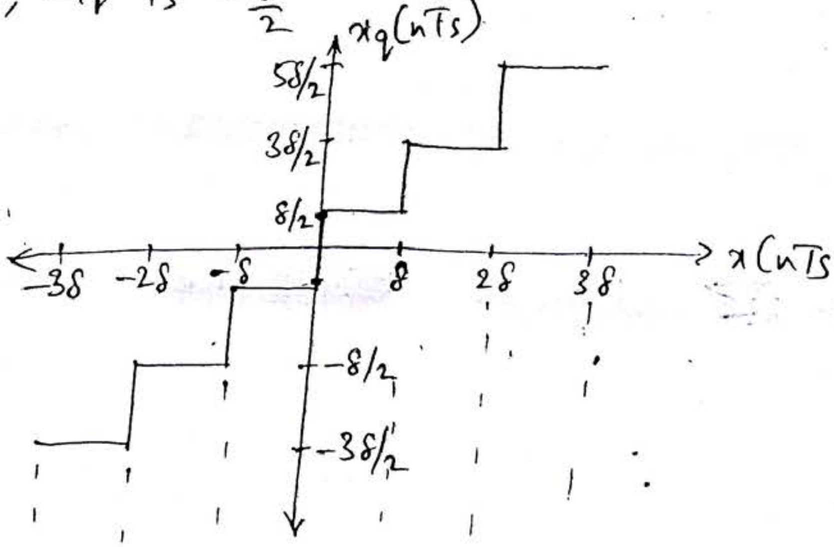


fig (a)

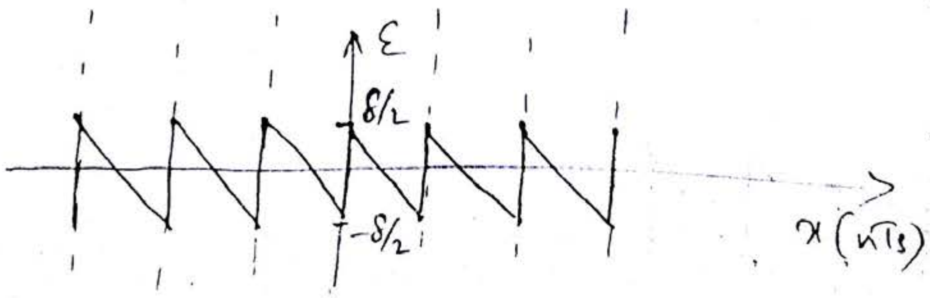


fig (b)

For $0 \le x(nT_s) < \delta$, $x_q(nT_s) = \delta/2$

$-\delta \le x(nT_s) < 0$, $x_q(nT_s) = -\delta/2$

Similarly when an input is between 3δ and 4δ o/p is $7\delta/2$, Fig (b) shows the quantization error.

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

$$= \frac{\delta}{2} - 0 = \frac{\delta}{2}$$

Thus the quantization error lies between $-\frac{\delta}{2}$ and $\frac{\delta}{2}$

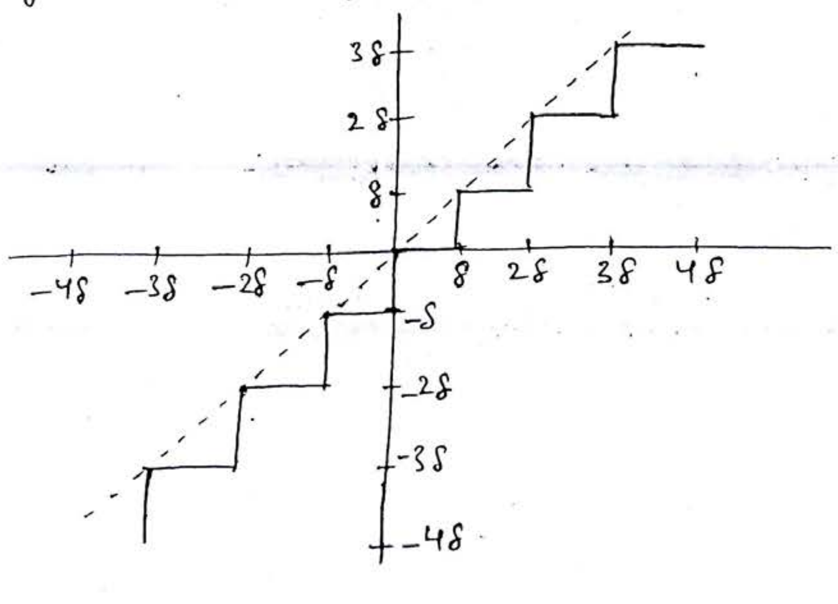
$$-\frac{\delta}{2} \le \varepsilon \le \frac{\delta}{2}$$

Maximum quantization error

$$\varepsilon_{max} = \left|\frac{\delta}{2}\right|$$

(iii) Biased Quantizer

the mid-tread and midriser quantizers are roundi[ng] quantizers. Biased quantizer is truncation quantizer



Fig(a)

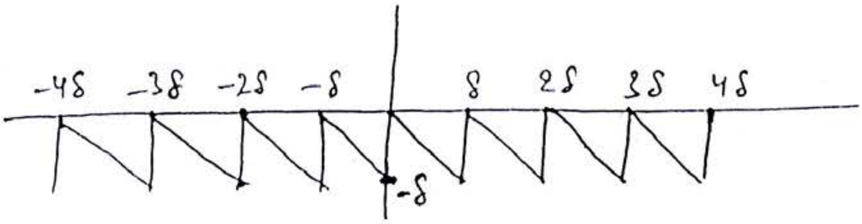


Fig (b)

when input is between '0' and 'δ', o/p is zero;

For $0 \leq x(nT_s) < \delta$, $x_q(nT_s) = 0$

For $-\delta \leq x(nT_s) < 0$, $x_q(nT_s) = -\delta$

Fig (b) shows the quantization error.

When I/p is δ, o/p is zero,

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

$= 0 - \delta = -\delta$

Thus the quantization error lies between 0 and −δ.

i.e. $-\delta \leq \varepsilon \leq 0$

Maximum quantization error

$\varepsilon_{max} = |\delta|$

The difference between staircase and dotted line gives the quantization error.

Signal to Noise ratio in PCM

step 1 :- Quantization error

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

step 2 :- Step size.

Let input $x(nT_s)$ be of continuous amplitude in the range $-x_{max}$ to $+x_{max}$.

For I/p of 4δ, o/p is $\frac{7}{2}\delta$

For I/p of −4δ, o/p is $-\frac{7}{2}\delta$

i.e $+x_{max} = \frac{7}{2}\delta$

$-x_{max} = -\frac{7}{2}\delta$

Total amplitude range $= x_{max} - (-x_{max}) = 2x_{max}$

Step size $= \frac{\text{Total amplitude range}}{\text{No. of levels}} = \frac{2x_{max}}{q}$

If $x(t)$ is normalized to minimum and maximum values equal to 1, $x_{max} = 1$, $-x_{max} = -1$

$\therefore$ step size $= \delta = \frac{2}{q}$ (for normalized signal)

<u>step 3</u> :- PDF of quantization error

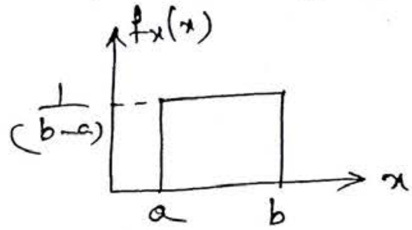
$$\varepsilon_{max} = \left|\frac{\delta}{2}\right| \quad \text{(for midriser or midtread)}$$

i.e $-\frac{\delta}{2} \leq \varepsilon_{max} \leq \frac{\delta}{2}$

Thus over the interval $\left(-\frac{\delta}{2}, \frac{\delta}{2}\right)$ quantization error is a uniformly distributed random variable.

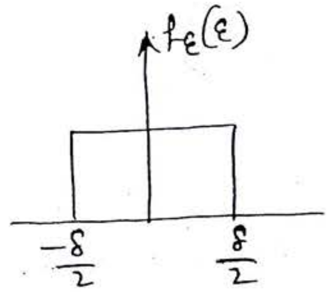
If x is a random variable, the PDF of x is

$$
\begin{aligned}
f_x(x) &= 0 \qquad x \leq a \\
&= \frac{1}{b-a} \qquad a < x \leq b \\
&= 0 \qquad x > b
\end{aligned}
$$



PDF of ε is

$$
\begin{aligned}
f_\varepsilon(\varepsilon) &= 0 \qquad \text{for } \varepsilon \leq \frac{\delta}{2} \\
&= \frac{1}{\delta} \qquad -\frac{\delta}{2} < \varepsilon \leq \frac{\delta}{2} \\
&= 0 \qquad \varepsilon > \frac{\delta}{2}
\end{aligned}
$$



<u>step 4</u> :- <u>Noise Power</u>

$$\text{Noise power} = \frac{V_{noise}^2}{R} = \frac{\text{mean square of noise}}{R}$$

Mean square value of a random variable x is

$$= \int_{-\infty}^{\infty} x^2 f_x(x)\, dx$$

$$\text{mean square value of } \varepsilon = \int_{-\frac{\delta}{2}}^{\delta/2} \varepsilon^2 f_\varepsilon(\varepsilon)\, d\varepsilon$$

$$= \int_{-\frac{\delta}{2}}^{\delta/2} \varepsilon^2 \cdot \frac{1}{\delta}\, d\varepsilon = \frac{1}{\delta}\left[\frac{\varepsilon^3}{3}\right]_{-\delta/2}^{\delta/2} = \frac{1}{3\delta}\left[\frac{\delta^3}{8} + \frac{\delta^3}{8}\right] = \frac{\delta^2}{12}$$

mean square of noise $= V^2_{noise} = \frac{\delta^2}{12}$

Noise power $= \frac{V^2_{noise}}{R} = \frac{\delta^2/12}{R}$

$= \frac{\delta^2/12}{1}$ (normalised when $R = 1\Omega$).

<u>step 5</u> :- S/N ratio

$$\frac{S}{N} = \frac{\text{Normalized signal power}}{\text{Normalized noise power}} = \frac{P}{\delta^2/12}$$

$$q = 2^v$$

$$\delta = \frac{2x_{max}}{q} = \frac{2x_{max}}{2^v}$$

$$\therefore \frac{S}{N} = \frac{P}{\left(\frac{2x_{max}}{2^v}\right)^2/12} = \frac{P}{\frac{4x^2_{max}}{2^{2v}} \cdot \frac{1}{12}} = \frac{3P \cdot 2^{2v}}{x^2_{max}}$$

If $x(t)$ is normalized, $x_{max} = 1$

$$\therefore \frac{S}{N} = 3 \times P \times 2^{2v}$$

If P is normalized, $P = 1$

$$\frac{S}{N} = 3 \times 2^{2v}$$

$$\frac{S}{N} \text{ in dB} = 10 \log_{10}\left(\frac{S}{N}\right) \text{dB}$$

$$= 10\log_{10}(3 \times 2^{2v}) = (4.8 + 6v) \text{ dB}.$$

<u>Problem</u> :- Determine the S/N ratio for 512 quantization levels.

Sol:- $\frac{S}{N}$ in dB $= 4.8 + 6V$

$= 4.8 + 6 \times 9$

$= 58.8$ dB

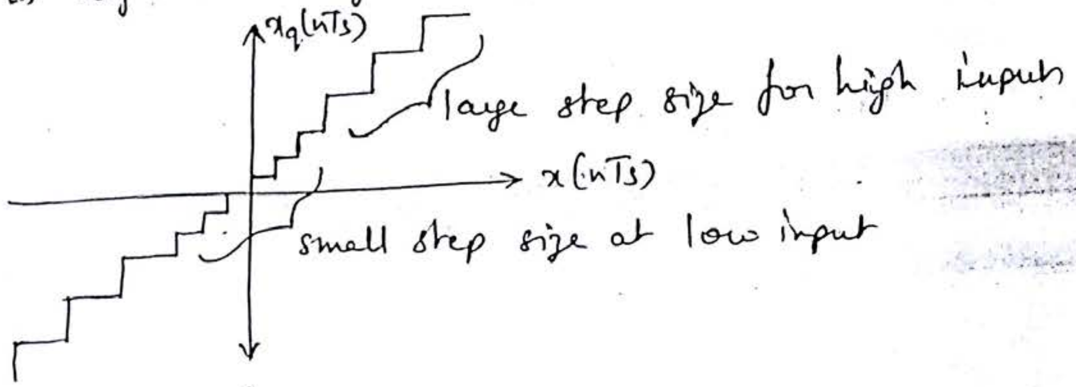
$q = 2^V = 512$

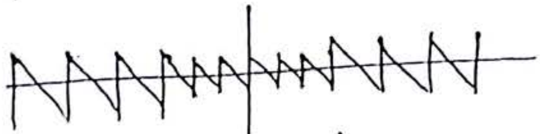
$V = \log_2 512$

$= 9$

Non Uniform Quantization

The step size is not fixed, it varies as per the input signal. The figure shows that the step size is small at low input signal levels, hence the quantization error is also small, therefore S/N is improved at low signal levels. Step size is high at higher input levels.





Necessity of non uniform quantization

The maximum quantization error $\varepsilon_{max} = \left|\frac{\delta}{2}\right|$

step size $\delta = \frac{2x_{max}}{q} = \frac{2}{q}$ (if $x(t)$ is normalized)

$q = 2^v$ (No. of quantization levels)

For example if $v = 4$ bits

$$q = 2^4 = 16$$

$$\delta = \frac{2}{16} = \frac{1}{8}$$

$$\varepsilon_{max} = \left|\frac{\delta}{2}\right| = \frac{\left(\frac{1}{8}\right)}{2} = \frac{1}{16}$$

i.e the quantization error is $\frac{1}{16}$th of the full scale voltage. For example let us assume that the full scale voltage is 16V.

$$\text{max. quantization error} = \frac{16}{16} = 1V$$

For large signal amplitudes of 15v, 16v etc, 1v is a small value which is negligible. But for small signal amplitudes of 2v, 3v, the error of 1v is quite high 50%, 30%.

This problem arises because of uniform quantization. Hence non uniform quantization is to be used in such cases.

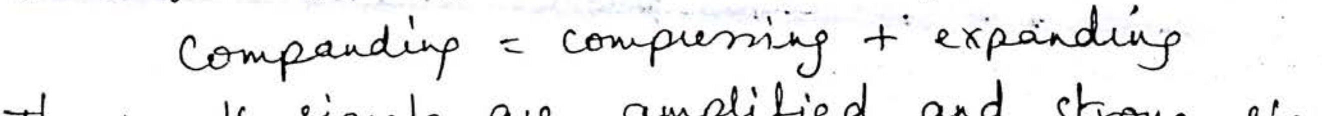
<u>Companding</u>

Non uniform quantization is achieved using companding. This is required to improve the S/N ratio of weak signals. In uniform quantization, the step size is fixed, quantization noise power is constant. But signal power is not constant, it is proportional to the square of the signal amplitude.

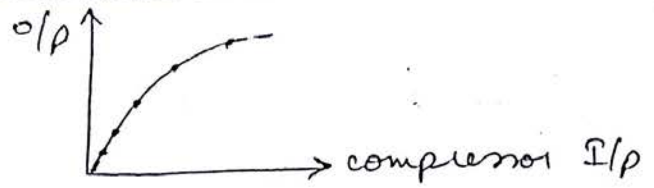
Hence signal power is small for weak signals but quantization noise power is constant. Therefore the S/N ratio for weak signals is very poor which affects the signal quality. Hence companding is used.

Companding = compressing + expanding

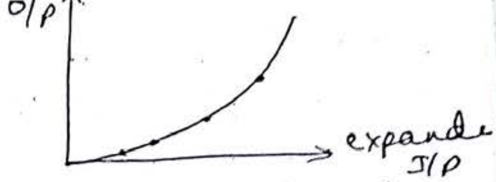
The weak signals are amplified and strong signals are attenuated before applying them to a uniform quantizer. At the receiver exactly opposite is followed called expansion done by an expander.



Compressor characteristics



Expander characteristics



Compressor provides a high gain to weak signals and small gain to strong signals. Thus weak signals are boosted to improve the S/N ratio.

Expander characteristics

All the artificially boosted signals by the compressor are brought back to their original amplitudes.

Compander characteristics

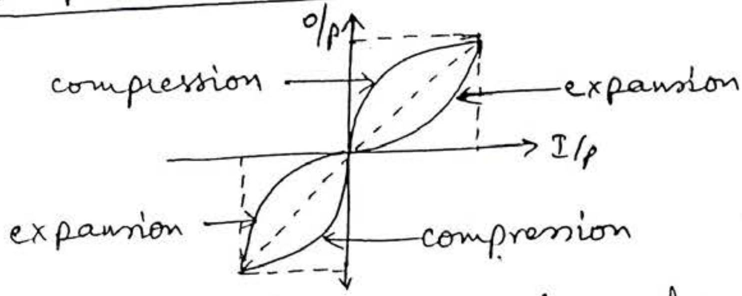


Fig. shows the compander characteristics which is the combination of compressor and expander characteristics. Due to the inverse nature of compressor and expander, the overall characteristics of the compander is a straight line.

This indicates that the boosted signals are brought back to their original amplitudes.

Types of Companders

i) μ-law companding

ii) A-law companding

μ-law Companding

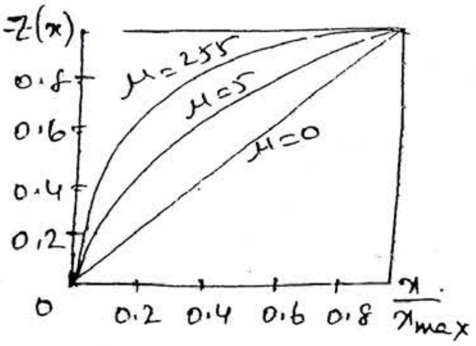
Here the compressor characteristics is continuous. It is approximately linear for smaller values of input levels and logarithmic for high input levels. The μ-law compressor characteristics is expressed as

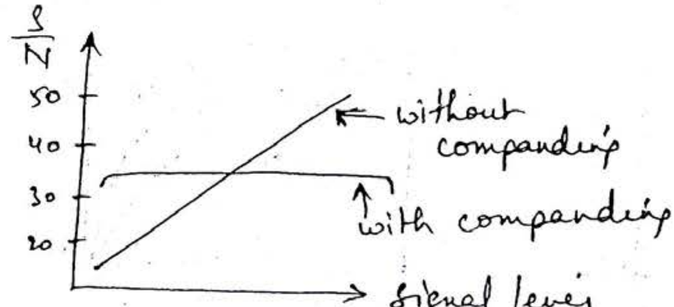
$$Z(x) = \operatorname{sgn}(x) \frac{\ln\left(1 + \mu |x|/x_{max}\right)}{\ln(1+\mu)}$$

where $0 \le \frac{|x|}{x_{max}} \le 1$

Here $Z(x)$ represents the o/p and x is the input to the compressor. Also $|x|$ represents the normalized value of input w.r.t maximum value x_{max}. $\operatorname{Sgn}(x)$ represents ± 1, positive and negative values of input and output.

The μ-law compressor characteristics for different values of μ is shown.



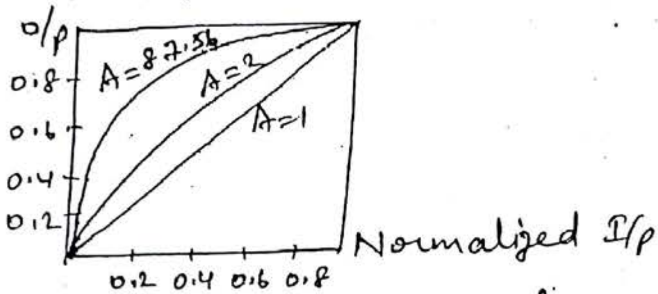


Practical value of μ is 255. When μ=0, it corresponds to uniform quantization. μ-law is used for speech and music signals. SNR is constant at all signal levels when companding is used.

A-law Companding

The compressor characteristic is piecewise, made up of a linear segment for low level inputs and logarithmic segment for high level inputs.

- Fig. shows the A-law compressor characteristics for different values of A. For A=1, the characteristics is linear which corresponds to uniform quantization. Practical value of A is 87.56.



A-law companding is used for PCM telephone systems in Europe. The linear segment of the characteristics is for low level inputs, logarithmic segment is for high level inputs.

$$\frac{Z(x)}{x_{max}} = \begin{cases} \dfrac{A|x|/x_{max}}{1+\log_e A} & \text{for } 0 \le \dfrac{|x|}{x_{max}} \le 1 \\[2ex] \dfrac{1+\log_e\left[A|x|/x_{max}\right]}{1+\log_e A} & \text{for } \dfrac{1}{A} \le \dfrac{|x|}{x_{max}} \le 1 \end{cases}$$

Applications of PCM

1) In Telephony
2) In space communication, space craft transmits signals to earth. Here the transmitted power is very low (10 to 15W) and the distances are huge (few million Km) hence due to noise immunity, only PCM systems are used.

Advantages of PCM

1) High noise immunity
2) Due to digital nature of the signal, repeaters can be placed between the transmitter and receiver which reduce the effect of noise.
3) PCM can be stored due to its digital nature
4) various coding techniques can be used so that only the desired person can decode the received signal

Disadvantages

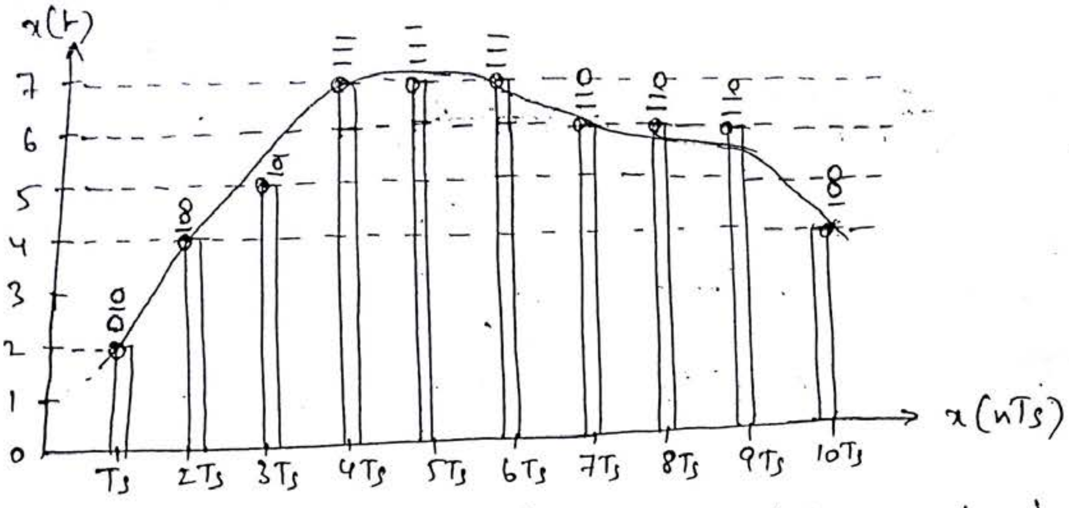
1) the encoding, decoding and quantizing circuitry is complex.
2) PCM requires large bandwidth as compared to other systems.

Differential Pulse Code modulation (DPCM)

Redundant information in PCM

The samples of a signal are highly connected with each other, because the signal does not change fast i.e its present sample value and the next sample does not differ by a large amount. Adjacent samples of the signal carry the same information with a little difference. When these samples are encoded by a standard PCM system the resulting encoded signal contains some redundant information.

Fig. shows a continuous time signal $x(t)$ by a dotted line. This is sampled at intervals T_s, $2T_s$, $3T_s$ ... nT_s. The samples are encoded using 3-bit PCM (7-levels, $2^3 = 8$ = 0 to 7 levels). The sample is quantized to the nearest digital level shown by small ~~levels~~ circles



The encoded binary values of each sample is written on the top of the samples. The samples at 4Ts, 5Ts, 6Ts are encoded to the same value 111. This information can be carried by only one sample, hence it is redundant. The samples at 2Ts and 3Ts differ only by one bit (100, 101), the first two bits are redundant.

Principle of DPCM

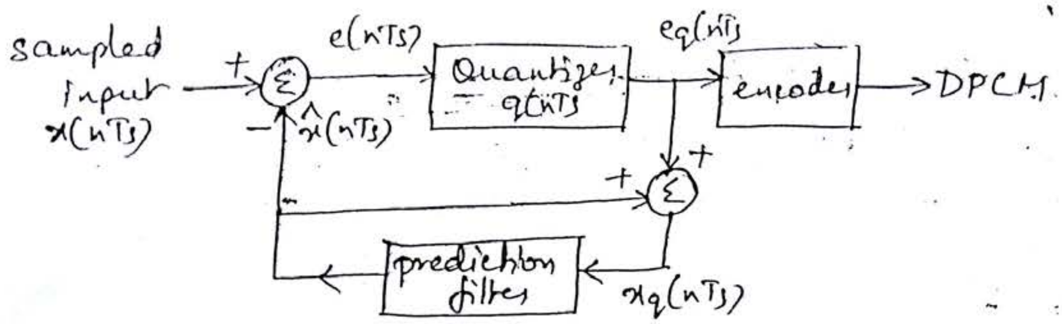
If this redundancy is reduced, then overall bit rate will decrease and the no. of bits required to transmit one sample will also be reduced

DPCM Transmitter

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.

Fig. shows the transmitter of DPCM system. The sampled signal is denoted by $x(nT_s)$ and the predicted sample signal is denoted $\hat{x}(nT_s)$. The comparator finds the difference between the actual sample $x(nT_s)$ and the predicted signal $\hat{x}(nT_s)$ called error denoted by $e(nT_s)$ where

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



Thus error is the difference between unquantized input sample $x(nTs)$ and prediction of it ~~is~~ $\hat{x}(nTs)$. The predicted value is produced by using a prediction filter. The quantizer o/p signal $e_q(nTs)$ and previous prediction is added and given as input to the prediction filter which is $x_q(nTs)$.

This makes the prediction more and more close to the actual sampled signal. The quantized signal $e_q(nTs)$ is very small and it can be encoded using small no. of bits. Thus no. of bits per sample are reduced in DPCM.

The quantizer o/p can be written as

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

$$q(nTs) = \text{quantization error}$$

The prediction filter input $x_q(nTs)$ is obtained by summing $\hat{x}(nTs)$ and quantizer o/p.

$$x_q(nTs) = \hat{x}(nTs) + e_q(nTs)$$

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

Substitute $e(nTs) = x(nTs) - \hat{x}(nTs)$

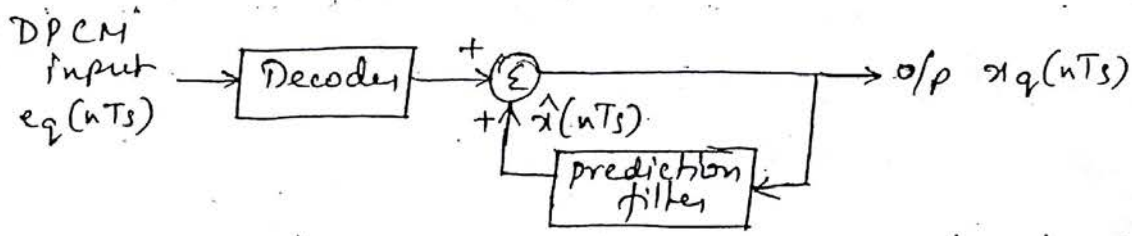
$$\therefore x_q(nTs) = \cancel{\hat{x}(nTs)} + x(nTs) - \cancel{\hat{x}(nTs)} + q(nTs)$$

$$= x(nTs) + q(nTs)$$

Thus the quantized version of the signal $x_q(nTs)$ is the sum of original sample value and quantization error $q(nTs)$. The quantization error can be positive or negative.

DPCM Receiver

Fig shows the block diagram of DPCM receiver.



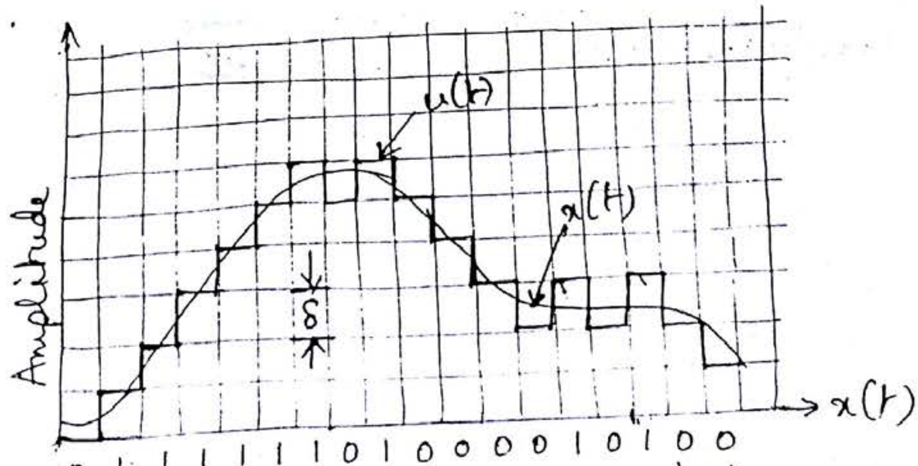
The decoder first reconstructs the quantized error signal from the incoming binary signal. The prediction filter o/p and quantized error signals are summed up to give the quantized version of the original signal.

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

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

Delta Modulation

Delta modulation transmits only one bit per sample, i.e the present sample value is compared with the previous sample. The indication whether the amplitude is increased or decreased is also sent. The input signal $x(t)$ is approximated to step signal by delta modulator, the step size is fixed.



The difference between input $x(t)$ and staircase approximated signal is confined to two levels $+\delta$ and $-\delta$

If the difference is positive, then the approximated signal is increased by one step i.e 'δ'. If the difference is negative, the approximated signal is reduced by 'δ'.

when the step is reduced, '0' is transmitted and if step size is increased, '1' is transmitted. Thus for each sample, only one binary bit is transmitted.

The principle of delta modulation can be explained by the following set of equations.

The error $e(nT_s)$ is

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

where $e(nT_s)$ = error at present sample

$x(nT_s)$ = sampled signal of $x(t)$

$\hat{x}(nT_s)$ = last sample approximation

Let the quantity $b(nT_s)$ be defined as

$$b(nT_s) = \delta \operatorname{sgn}[e(nT_s)] \quad (\text{sign of } e(nT_s))$$

Depending on the sign of error $e(nT_s)$ the sign of step size δ will be decided.

$$b(nT_s) = +\delta \quad \text{if} \quad x(nT_s) > \hat{x}(nT_s)$$
$$= -\delta \quad \text{if} \quad x(nT_s) < \hat{x}(nT_s)$$

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

$= -\delta$, binary '0' is transmitted

T_s = sampling interval,

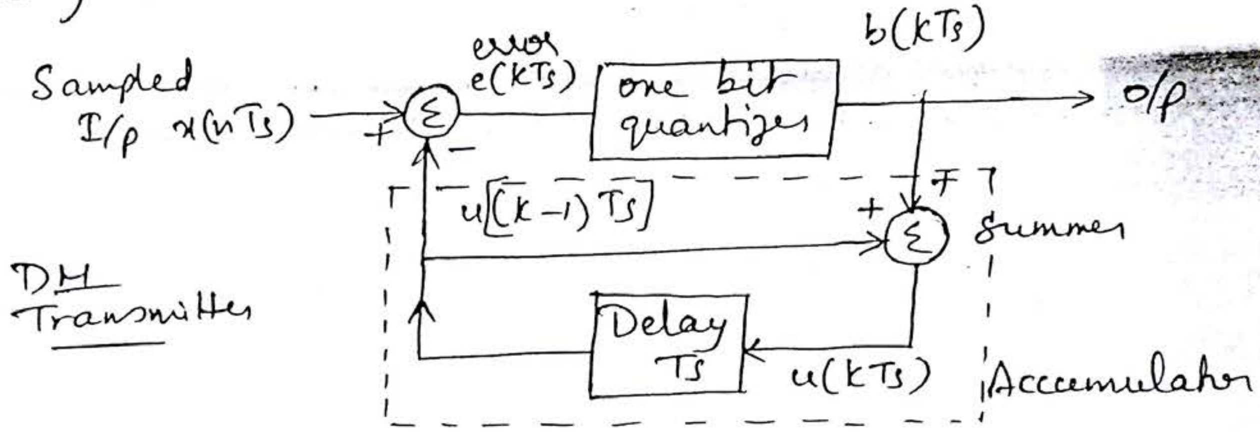
Delta Modulation Transmitter

Fig shows the transmitter. The summer in the accumulator adds quantizer o/p ($\pm\delta$) with the previous sample approximation. This gives the present sample approximation.

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

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 approximate signal $\hat{x}(nT_s)$ are subtracted to get error signal $e(nT_s)$.

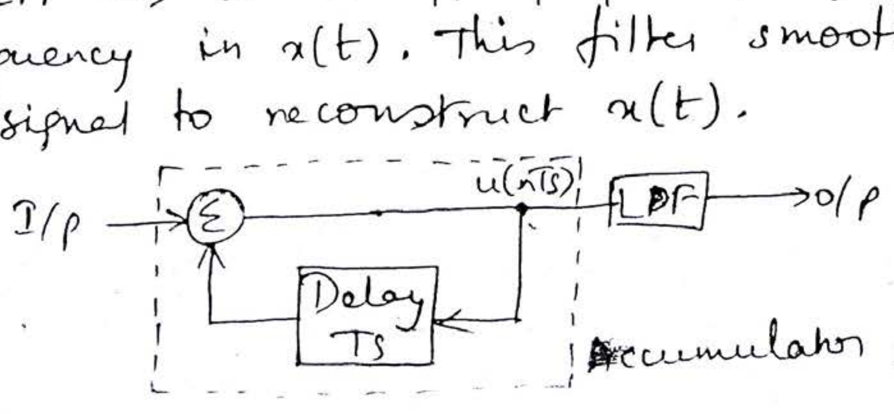
Depending upon the sign of $e(nT_s)$, the one-bit quantizer produces an o/p step of $+\delta$ or $-\delta$. If the step size is $+\delta$, binary '1' is transmitted, else binary '0' is transmitted.



Delta Modulation Receiver

At the receiver shown, the accumulator and low pass filter are used. The accumulator generates the stair case approximated signal o/p and is delayed by one sampling period T_s. It is then added to the input signal. If the input is binary '1' then it adds $+\delta$ step to the previous o/p (which is delayed). If the input is binary '0', then it adds $-\delta$ to the delayed signal.

The LPF has a cut off frequency equal to the highest frequency in $x(t)$. This filter smoothens the staircase signal to reconstruct $x(t)$.



Advantages of Delta modulator

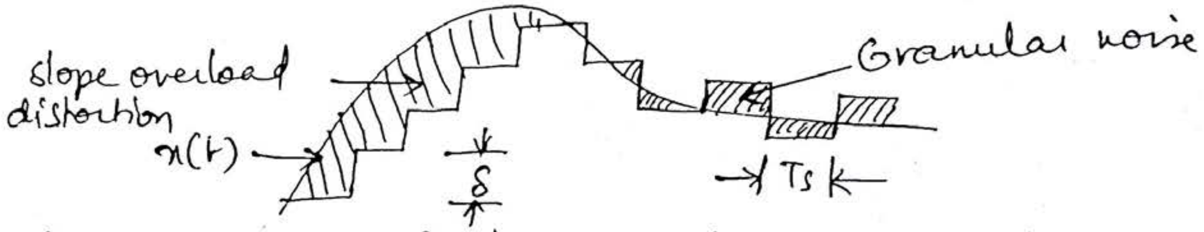
1) DM Transmits only one bit per sample, thus the signalling rate and the transmission channel bandwidth is quite small.

2) The transmitter and receiver implementation is very much simple for delta modulation as there is no A/D converter.

Disadvantages

1) Slope overload distortion.

This arises because of the large dynamic range of the input signal. As seen from the fig. the rate of rise of input signal x(t) is so high that the stair case signal cannot approximate it.



The step size δ becomes too small for staircase signal u(t) to follow the steep segment of x(t). Thus there is a large error between the staircase approximated signal and original input signal x(t). This error is called slope overload distortion.

To reduce this error, the step size should be increased when slope of the signal is high.

2) Granular noise

This occurs when the step size is too large compared to small variations in input signal. For very small variations in the input, the staircase signal is changed by a large amount δ because of large step size. Fig shows that when input signal is almost flat, the staircase signal u(t) keeps on oscillating by ±δ around the signal. The error between the input and approximated signal is called granular noise.

The solution to this problem is to make the step size small. Thus large step size is required to accommodate wide range of the input signal to reduce slope overload distortion and small step size is required to reduce the granular noise. Adaptive delta modulation is used to overcome these errors.

Adaptive Delta modulation

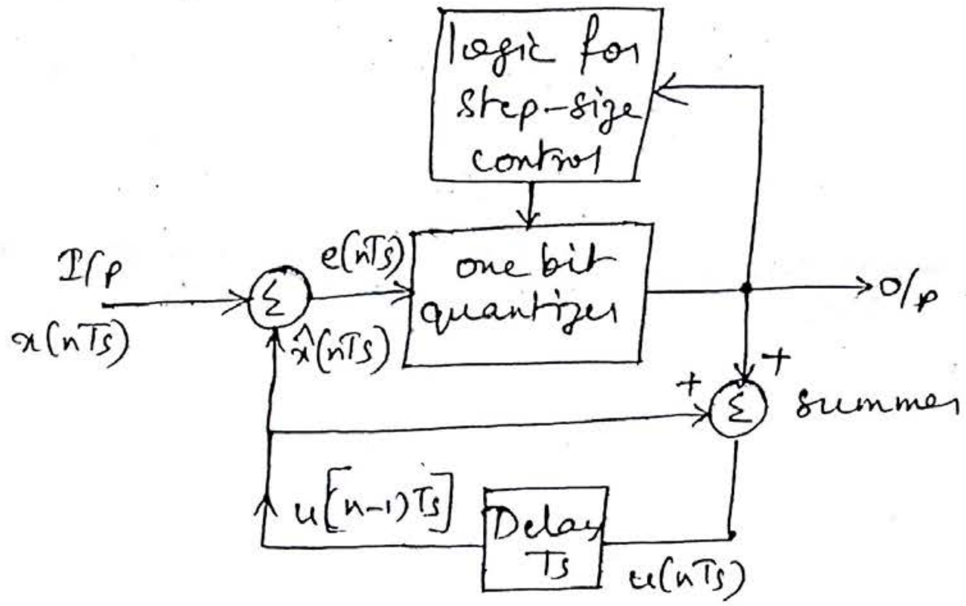
operating principle

To overcome the quantization errors due to slope over load distortion and granular noise, the step size δ is made adaptive to the variations in the input signal x(t). In the steep segment of the signal x(t), the step size is increased, and when the input is varying slowly, the step size is reduced, this is known as adaptive delta modulation.

The Adaptive delta modulators can take continuous changes in step size or discrete changes in step size.

ADM Transmitter

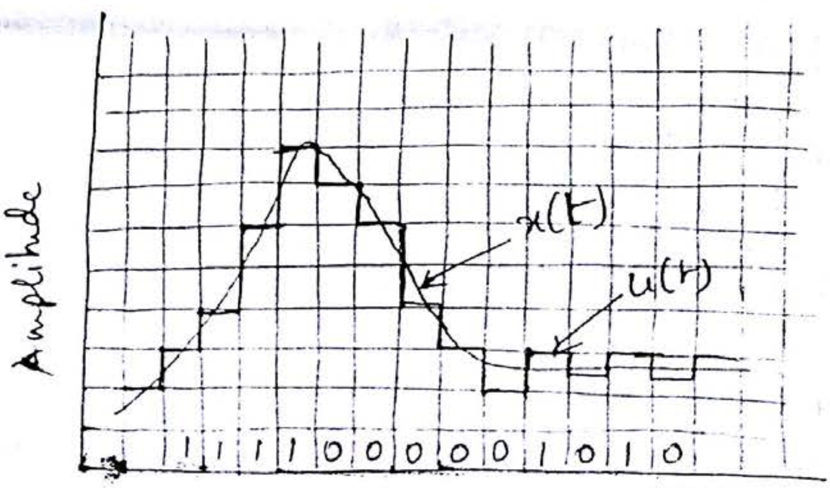
Fig shows the transmitter of ADM. The logic for step size control is added. The step size increases or decreases according to certain rule depending on the output of one-bit quantizer.



ADM Transmitter

For example, if one bit quantizer o/p is high (1) then the step size may be doubled for next sample. If the one bit quantizer o/p is low, then step size may be reduced by one step.

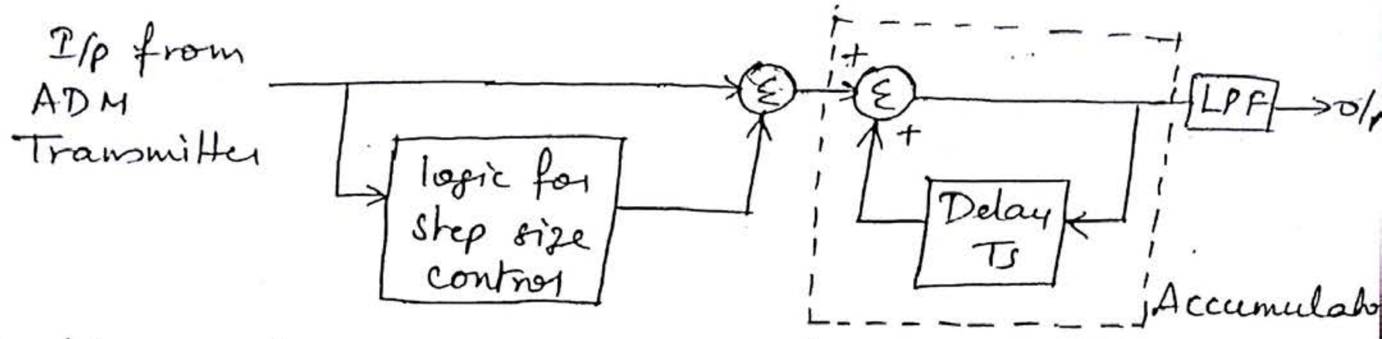
Fig below shows the waveforms of adaptive delta modulator and the sequence of bits transmitted



$x(t)$ = I/p

$u(t)$ – staircase o/p

ADM Receiver



In the receiver of ADM, the first part generates the step size from each incoming bit. Exactly the same process is followed as that in transmitter. The previous input and present input decide the step size.

It is then given to an accumulator which builds up staircase waveform. The LPF then smoothens out the staircase waveform to reconstruct the smooth signal.

Advantages over Delta modulation

1) the S/N ratio is better than ordinary delta modulation because of the reduction in slope overload distortion and granular noise.
2) Because of the variable step size, the dynamic range of ADM is wide
3) Utilization of bandwidth is better than delta modulation.