DIGITAL COMMUNICATION

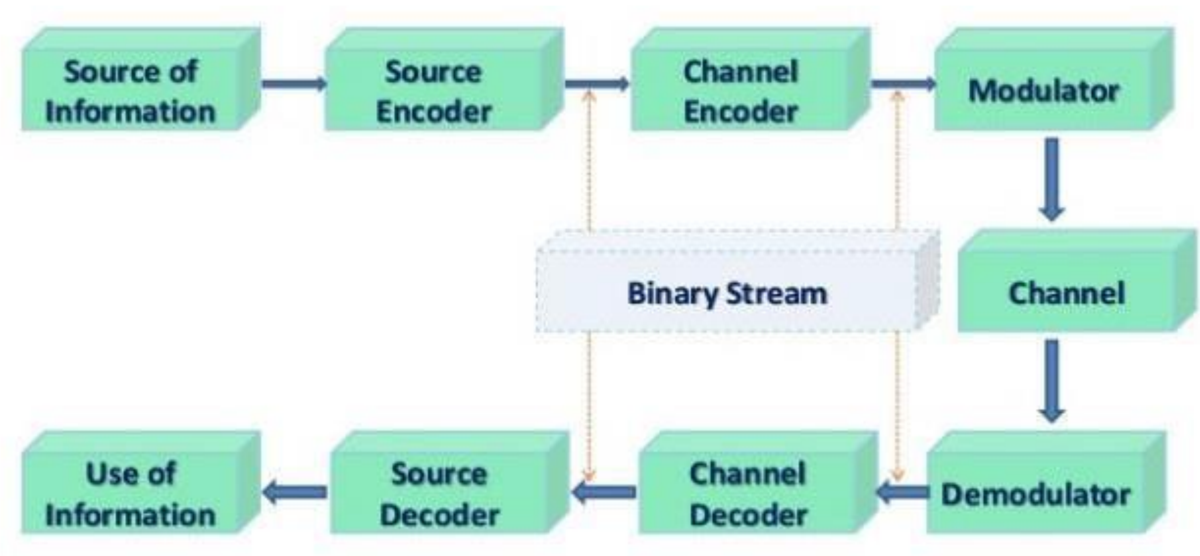
Module 1

Digital communication is the process of devices communicating information digitally.

Necessity of digital communication systems (Advantage of DC)

- Can withstand channel noise and distortion much better than analog communication
- Regenerative repeaters can be used in DC which will prevent accumulation of noise along the path.
- Digital hardware implementation is flexible.
- Digital signals can be coded to yield extremely low error rates, high fidelity and well as privacy.
- It is easier and more efficient to multiplex several digital signals.
- Digital signal storage is relatively easy and inexpensive.
- Reproduction with digital messages is extremely reliable without deterioration.
- The cost of digital hardware continues to halve every two or three years, while performance or capacity doubles over the same time period.

Block diagram digital communication systems



1. Information Source and Input Transducer:

The source of information can be analog or digital. In digital communication the signal produced by this source is converted into digital signal which consists of 1′s and 0′s. For this we need a source encoder.

2. Source Encoder:

Source Encoding or Data Compression: the process of efficiently converting the output of whether analog or digital source into a sequence of binary digits is known as source encoding.

3. Channel Encoder:

The purpose of the channel encoder is to introduce some redundancy in the binary information sequence that can be used at the receiver to overcome the effects of noise and interference encountered in the transmission on the signal through the channel.

4. Digital Modulator:

The binary sequence is passed to digital modulator which in turns convert the sequence into electric signals so that we can transmit them on channel . The digital modulator maps the binary sequences into signal wave forms

5. Channel:

The communication channel is the physical medium that is used for transmitting signals from transmitter to receiver. Eg: coaxial cable, twisted pair, optic fiber

6. Digital Demodulator:

Here digital demodulation is done which is the process of restoring the data bits back from a digitally modulated signal .

7. Channel Decoder:

which attempts to reconstruct the original information sequence from the knowledge of the code used by the channel encoder and the redundancy contained in the received data

8. Source Decoder:

At the end, if an analog signal is desired then source decoder tries to decode the sequence from the knowledge of the encoding algorithm. And which results in the approximate replica of the input at the transmitter end.

9. Output Transducer:

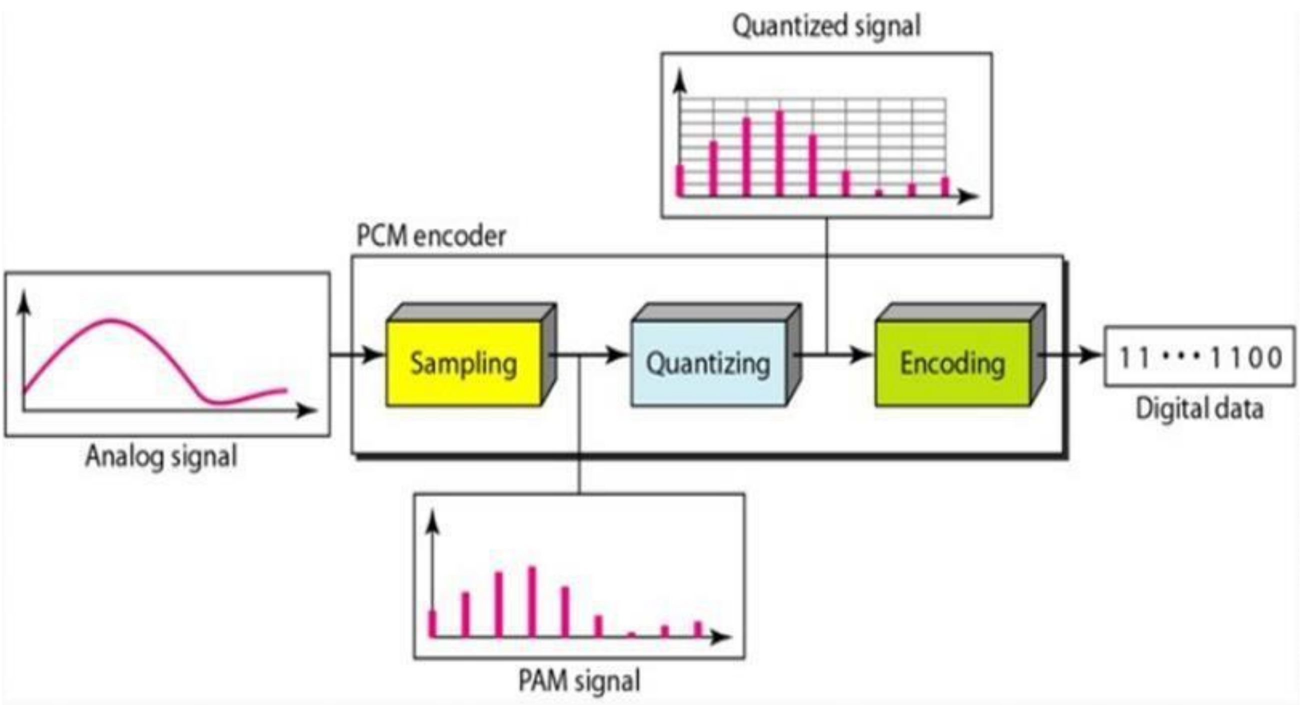
Finally we get the desired signal in desired format analog or digital

Three steps involved in conversion of analog signal to digital signal

1) Sampling

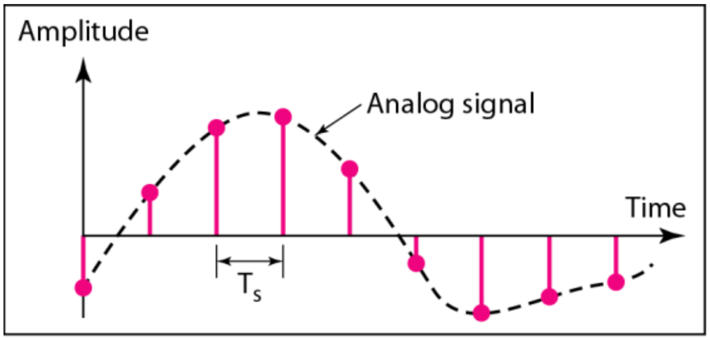
2) Quantization

3) Binary encoding



Sampling

Sampling is the process of measuring the instantaneous value of an analog signal in the discrete form.ie, it is a Process of converting analog signal into discrete signal.



a. Ideal sampling

- The signal is sampled at regular intervals such that each sample is proportional to amplitude of signal at that instant
- Analog signal is sampled every T_s $Secs$, called sampling interval. $f_s=1/T_s$ is called sampling rate or sampling frequency.

Types of sampling techniques

a. Ideal/Instantaneous Sampling or Impulse Sampling:

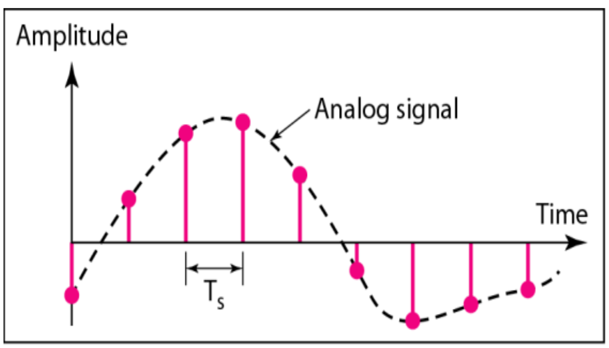
- An impulse at each sampling instant.
- 0Sampling function is train of spectrum remains constant impulses throughout frequency range. It is not practical.

b. Natural sampling

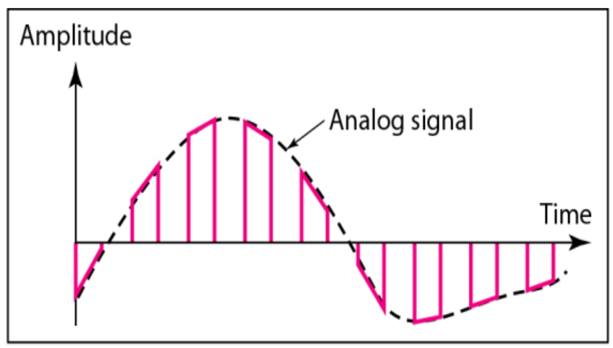
- A pulse of Short width with varying amplitude.
- The spectrum is weighted by a **sinc** function.
- Amplitude of high frequency components reduces.

c. Flat Top sampling

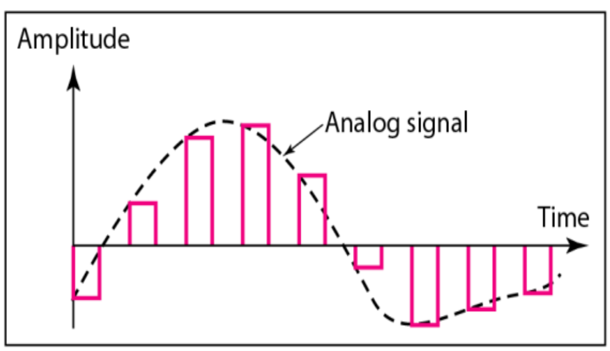
- Uses sample and hold, like natural but with single amplitude value.
- In the spectrum high frequency components are attenuated due sinc pulse roll off. This is known as **Aperture effect.**
- If pulse width increases aperture effect is more i.e. more attenuation of high frequency components.



a. Ideal sampling



b. Natural sampling



c. Flat-top sampling

Sampling theorem

The sampling theorem, also known as the Nyquist-Shannon sampling theorem, states that in order to accurately reconstruct a continuous-time signal from its samples, the sampling frequency must be greater than twice the highest frequency component present in the signal.

In mathematical terms, the sampling theorem can be stated as follows:

If a continuous-time signal x(t) contains no frequencies higher than fmax, then it can be reconstructed perfectly from its samples taken at a rate **fs >= 2fmax**, where fs is the sampling frequency.

This means that in order to avoid aliasing and loss of information, the sampling rate must be at least double the highest frequency component in the signal. By doing so, the original signal can be reconstructed without distortion or loss of information.

Nyquist rate

The Nyquist rate, also known as the Nyquist frequency or Nyquist limit, refers to the minimum sampling rate required to accurately represent or reconstruct a continuous-time signal.

Mathematically, the Nyquist rate (fs) can be calculated as:

fs = 2 * fmax

Where fs is the sampling rate and fmax is the highest frequency component in the signal.

Quantization

Quantization is representing the sampled values of the amplitude by a finite set of levels, which means converting a continuous-amplitude sample into a discrete-time signal.

- Both sampling and quantization result in the loss of information.
- The quality of a Quantizer output depends upon the number of quantization levels used.
- The discrete amplitudes of the quantized output are called as representation levels
- The spacing between the two adjacent representation levels is called a step-size.

There are two types of Quantization

Uniform Quantization

The type of quantization in which the quantization levels are uniformly spaced

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Non-uniform Quantization.

The type of quantization in which the quantization levels are unequal and mostly the relation between them is logarithmic

In non uniform quantization the step size should be decreased for weak signal and it will increase for strong signal for getting a good SNR

There are two types of uniform quantization. They are Mid-Rise type and Mid-Tread type. The following
figures represent the two types of uniform quantization

The **Mid-Rise** type is so called because the origin lies in the middle of a raising part of the staircase like graph. The quantization levels in this type are even in number.

The **Mid-tread** type is so called because the origin lies in the middle of a tread of the stair-case like graph. The quantization levels in this type are odd in number.

Both the mid-rise and mid-tread type of uniform quantizer are symmetric about the origin

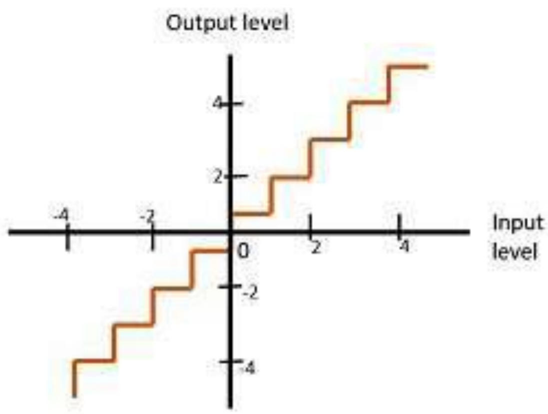


Fig 1 : Mid-Rise type Uniform Quantization

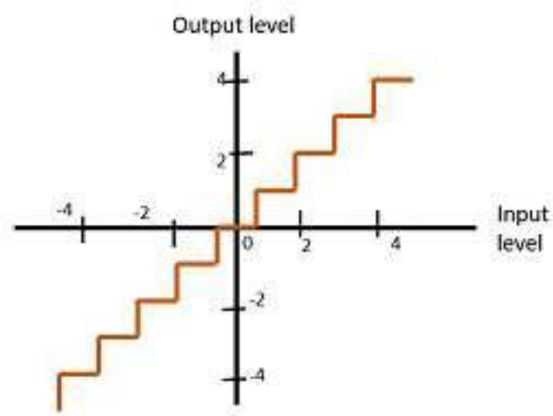


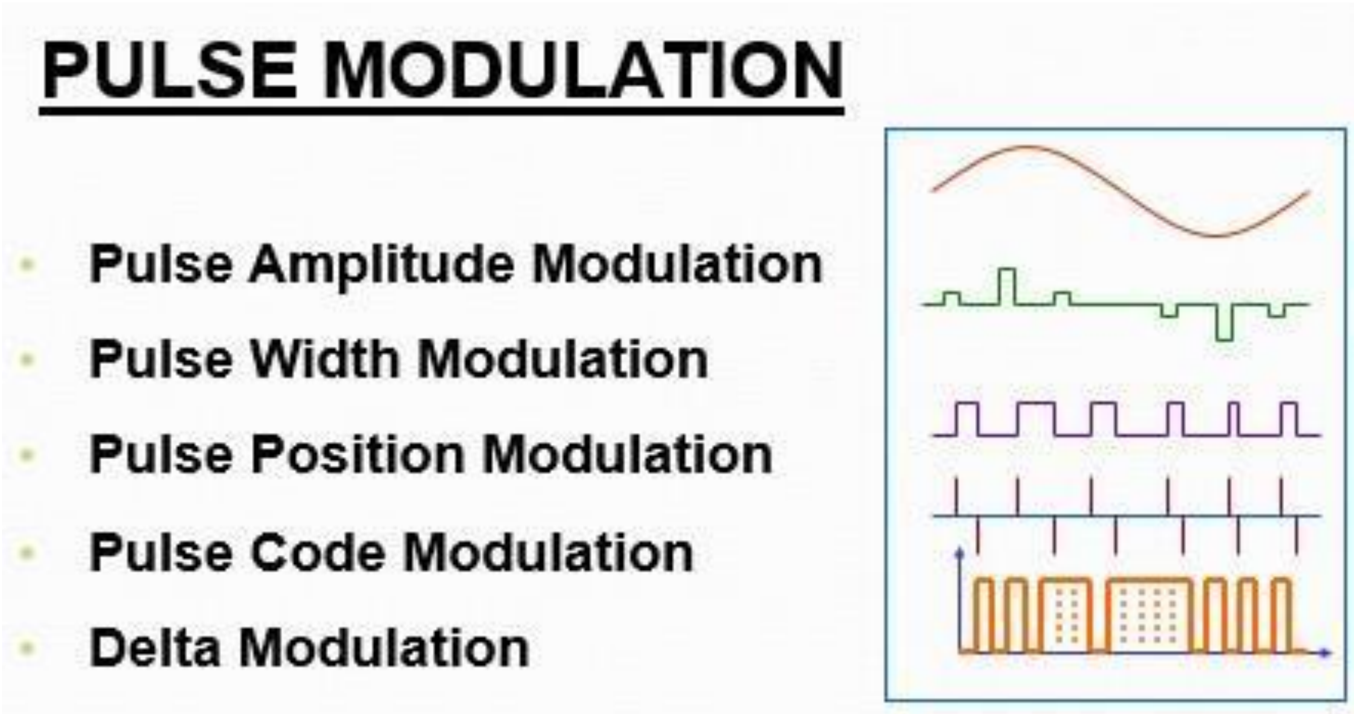
Fig 2 : Mid-Tread type Uniform Quantization

Quantization error

Quantization error refers to the difference between the original analog signal and its quantized representation in a communication system. It arises due to the discrete nature of quantization.

When an analog signal is quantized, each sample is mapped to the nearest quantization level. The quantization error occurs because the original analog signal may not exactly match any of the available levels. As a result, there is a difference between the original signal and its quantized representation

ie, **Quantization error = sampled value – quantized value**



Pulse Code Modulation(PCM)

Instead of a pulse train, PCM produces a series of numbers or digits, and hence this process is called as digital. Each one of these digits, though in binary code, represent the approximate amplitude of the signal sample at that instant.

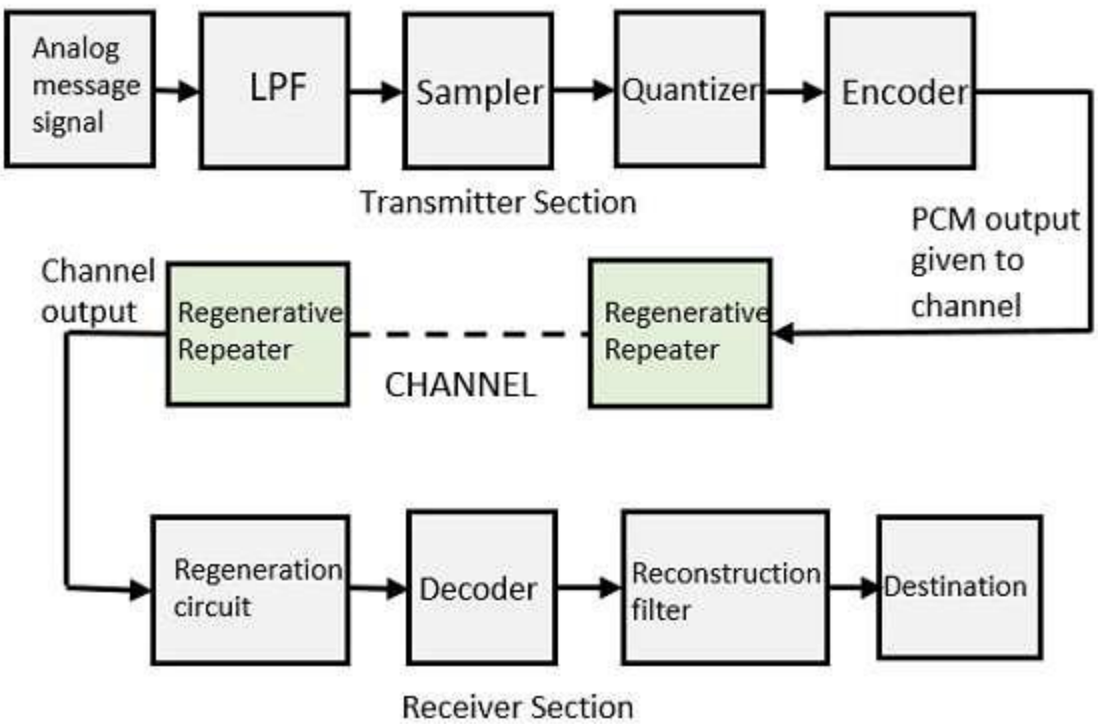
In PCM, the message signal is represented by a sequence of coded pulses. This message signal is achieved by representing the signal in discrete form in both time and amplitude.

Basic Elements of PCM

The transmitter section of a Pulse Code Modulator circuit consists of **Sampling, Quantizing** and **Encoding**, which are performed in the analog-to-digital converter section. The low pass filter prior to sampling prevents aliasing of the message signal.

The basic operations in the receiver section are **regeneration of impaired signals, decoding,** and **reconstruction** of the quantized pulse train.

Block diagram of PCM



Low Pass Filter

This filter eliminates the high frequency components present in the input analog signal which is greater
than the highest frequency of the message signal, to avoid aliasing of the message signal.

Sampler

This is the technique which helps to collect the sample data at instantaneous values of message signal, so
as to reconstruct the original signal. The sampling rate must be greater than twice the highest frequency
component of the message

Quantizer

Quantization is representing the sampled values of the amplitude by a finite set of levels, which means converting a continuous-amplitude sample into a discrete-time signal.

Encoder

The digitization of analog signal is done by the encoder. It convert each quantized level to a binary code. If we are using 'n' bit encoder then each quantized level is converter to n bit code

Regenerative Repeater

This section increases the signal strength. The output of the channel also has one regenerative repeater circuit, to compensate the signal loss during the transmission of signal through the channel and reconstruct the signal, and also to increase its strength

Decoder

The decoder circuit decodes the pulse coded waveform to reproduce the original signal. This circuit acts as the demodulator.

Reconstruction Filter

After the digital-to-analog conversion is done by the regenerative circuit and the decoder, a low-pass filter is employed, called as the reconstruction filter to get back the original signal.

Hence, the Pulse Code Modulator circuit digitizes the given analog signal, codes it and samples it, and then transmits it in an analog form. This whole process is repeated in a reverse pattern to obtain the original signal.

Bandwidth requirements of PCM

A digitized signal will always need more bandwidth than the original analog signal.The bandwidth required to transmit this signal depends on the type of line encoding used.

• Bandwidth for PCM signal = n.fm

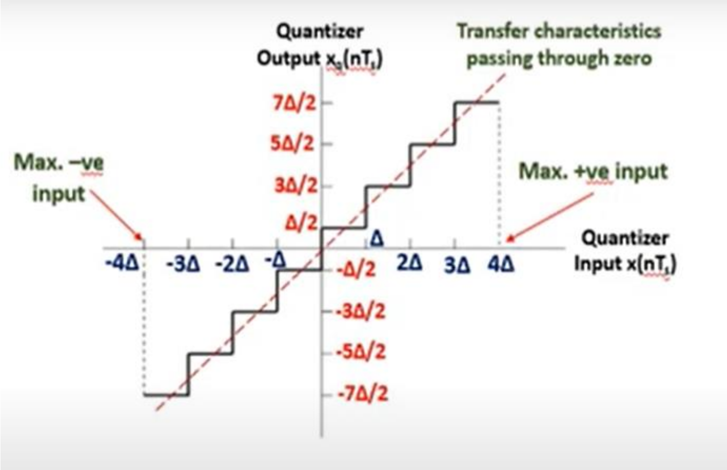
Where, n – No. of bits in PCM code

fm – signal bandwidth

Quantization Noise in PCM

Quantization error refers to the difference between the original analog signal and its quantized representation in a communication system

Quantization noise is introduced in the transmitter and is carried all the way along to the receiver output.it is signal dependent in the sense that it disappears when the message signal is switched off . It can be made negligibly small through the use of adequate number \representation levels in the quantizer and suitable companding technics



Let us consider total voltage range to be quantized is from – Vmax to Vmax and number of quantized levels or representation level be L

Here number of quantized levels $L = 2^n$, where n is number of bit used for encoding a sample

The total amplitude range Vpp = Vmax- (-V max)

= 2 V max

Therefore the step size delta (Δ) = 2 Vmax/ L

The maximum quantization noise /error (Qe) will be between (-Δ/2) and (Δ/2)

ie, (-Δ/2)<=Qe<=(Δ/2)

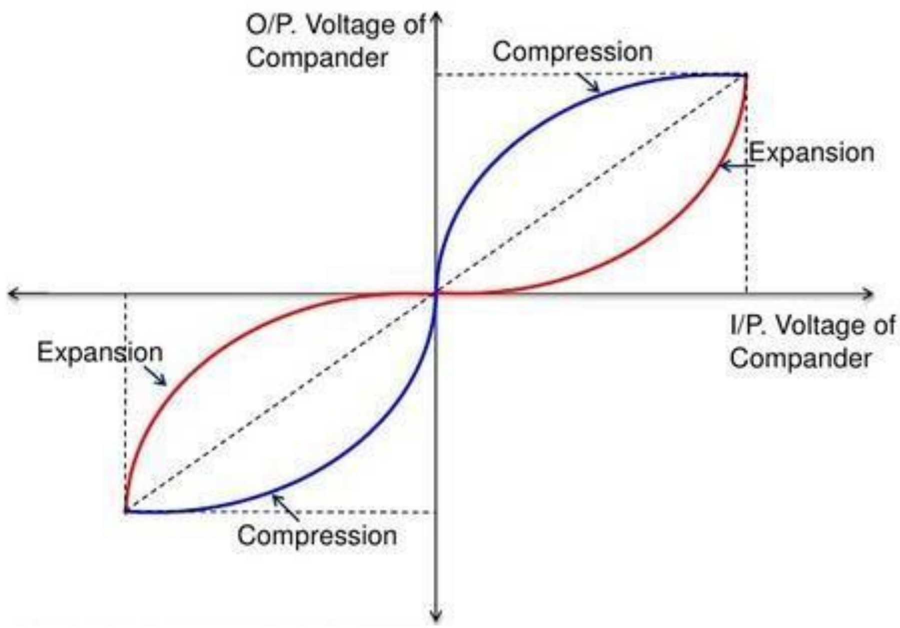
the average power of Qe can be expressed as $\Delta^2/12$

Companding in PCM

- In non uniform quantization the step size should be decreased for weak signal and it will increase for strong signal for getting a good SNR for all signal. But Non-uniform quantizers are difficult to make and expensive.So we find an alternative is to first pass the speech signal through nonlinearity before quantizing with a uniform quantizer, for this we using companding

The word **Companding** is a combination of Compressing and Expanding, which means that it does both. This is a non-linear technique used in PCM system , which compresses the data at the transmitter and expands the same data at the receiver. For improving SNR of weak signal.

Here we are doing compression the signal and passing through a uniform quandizer and at receiver we will do the expansion hence weak signal SNR wiil be maintained .Compressor will amplify the weak signal where as expander at receiver will attenuate the weak signal hence we will get the original signal at the output without distrotion



There are two types of Companding techniques. They are –

A-law Companding Technique

Get Uniform quantization at **A = 1**, where the characteristic curve is linear and no compression is done.A-law has mid-rise at the origin. Hence, it contains a non-zero value.

- A-law Companding is used for PCM telephone systems.

µ-law Companding Technique

Get Uniform quantization at **µ = 0**, where the characteristic curve is linear and no compression is done.µ-law has mid-tread at the origin. Hence, it contains a zero value.

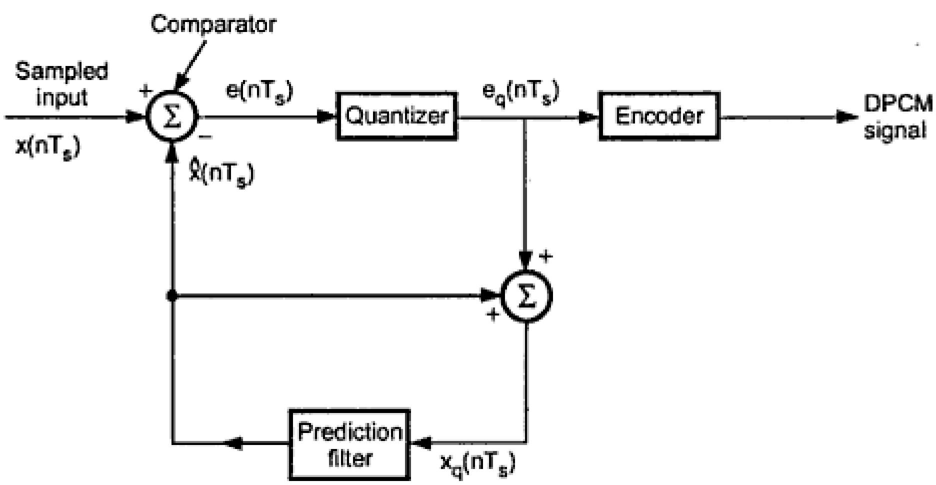
- µ-law companding is used for speech and music signals.

Differential PCM (DPCM)

For the samples that are highly correlated, when encoded by PCM technique, will produce huge redundant information.if this redundancy is reduced , then the overall bit rate will decreased and number of bits required to transmit one sample will also be reduced . To remove this redundant information and to have a better output, it is a wise decision to take a predicted sampled value, assumed from its previous sample and summarize them with the quantized values. Such a process is called as **Differential PCM (DPCM)** technique

DPCM Transmitter

The DPCM Transmitter consists of Quantizer and Predictor with two summer circuits. Following is the block diagram of DPCM transmitter quantizer stage.in this stage only there is difference from PCM block diagram



The DPSC works on the principle of prediction. The value of present sample is predicted from the past sample . the prediction will not be exact but it will be very close to the actual value. The comparator find out the difference between the actual sample x(nTs)and the predicted sample value x^(nTs) this is called error e(nTs). The predicted value is produced by a prediction filter . The quantized output signal $e_q(nT_s)$and previous prediction is added and given as input to the prediction filter $x_q(nT_s)$. This make the prediction more perfect. we can see that the quantized output signal $e_q(nT_s)$is very small and can be encoded with small number of bits hence number of bits per sample are reduced in DPCM

DPCM Receiver

The block diagram of DPCM Receiver consists of a decoder, a predictor, and a summer circuit alone

Reconstruction of DPCM Signal

Fig. shows the block diagram of DPCM receiver.

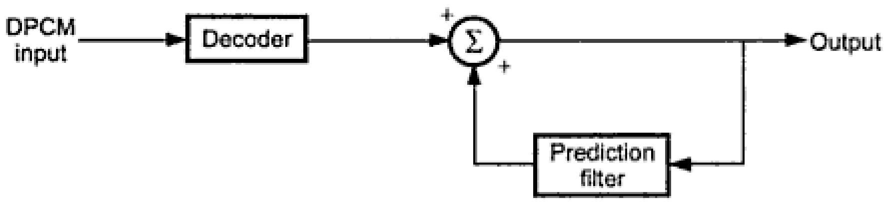


Fig. DPCM receiver

The decoder first reconstructs the quantized error signal from incoming binary signal. The prediction filter output and quantized 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 quantization error $q(nT_s)$, which is introduced permanently in the reconstructed signal.

Advantages of Dpcm:

1) Bandwidth Requirement Of Dpcm Is Less Compared To PCM
2) Quantization Error Is Reduced Because Of Prediction Filter.
3) Numbers Of Bits Used To Represent .One Sample Value Are Also Reduced Compared To Pcm.

Delta Modulation(DM)

PCM transmits all the bits which are used to code the sample. Hence signaling rate and transmission channel bandwidth are large in PCM. To overcome this problem Delta Modulation is used.

Operating Principle of DM

Delta modulation transmits only one bit per sample. That is the present sample value is compared with the previous sample value and the indication, whether the amplitude is increased or decreased is sent. Input signal is approximated to step signal by the delta modulator. In delta modulator step size(Δ) is fixed .If the difference between present sample value and previous sample is positive, then approximated signal is increased by one step ie.+ Δ , if the difference is negative, then approximated signal is reduced by -Δ.When the step is reduced, 0' is transmitted and if the step is increased, "1' is transmitted. Thus for each sample, only one binary bit is transmitted.

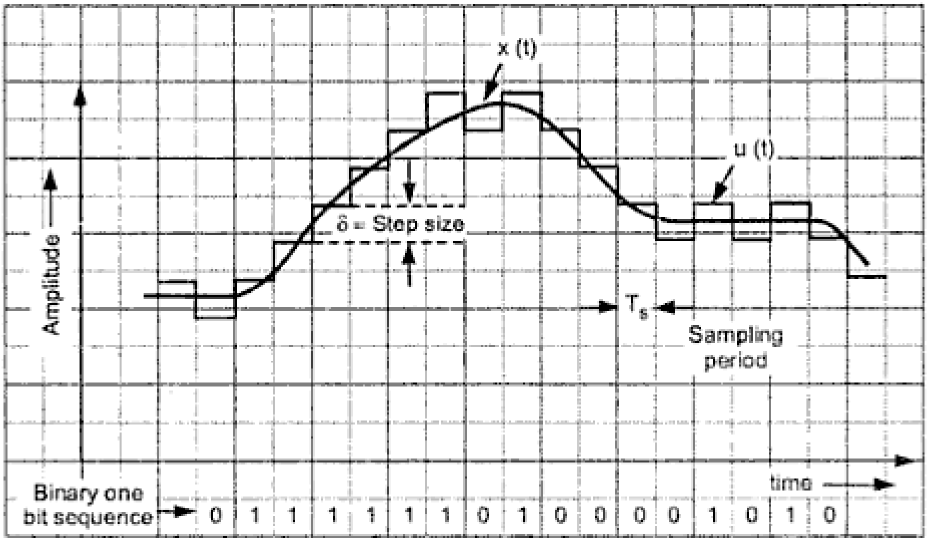


Fig. Delta modulation waveform

DELTA TRANSMITTER AND RECEIVER

The summer in the accumulator adds quantizer output ($\pm\delta$) with the previous sample approximation. This gives present sample approximation. i.e.,

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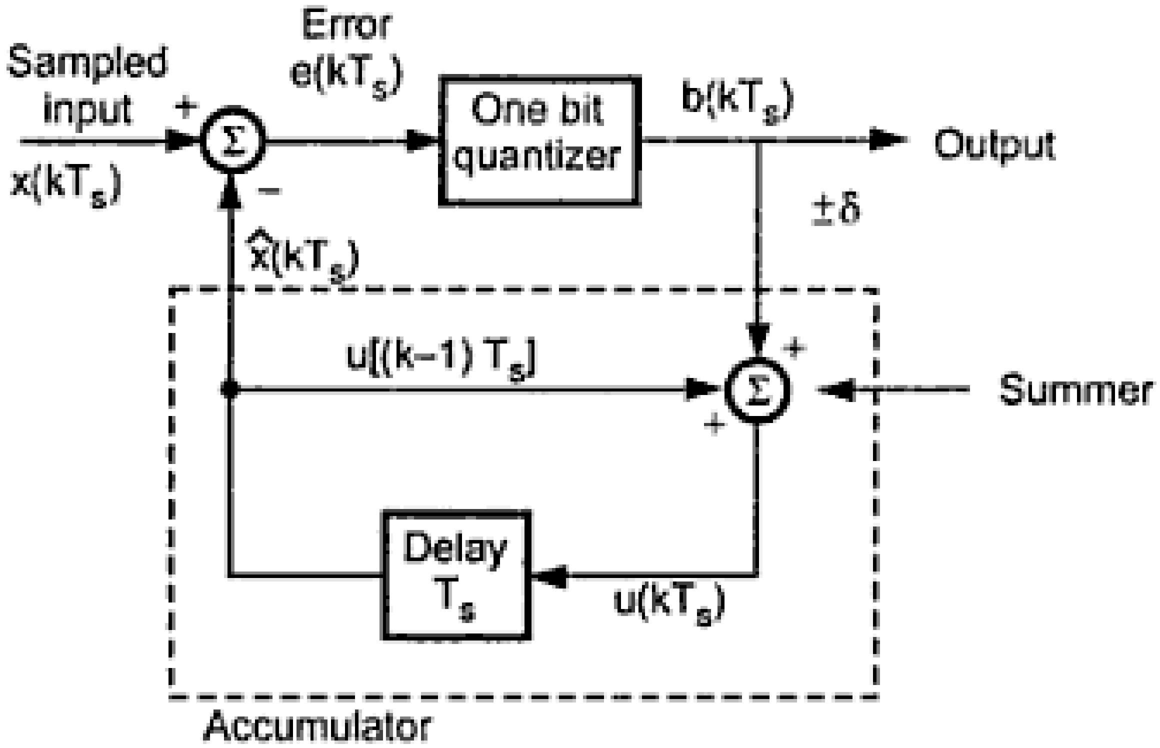
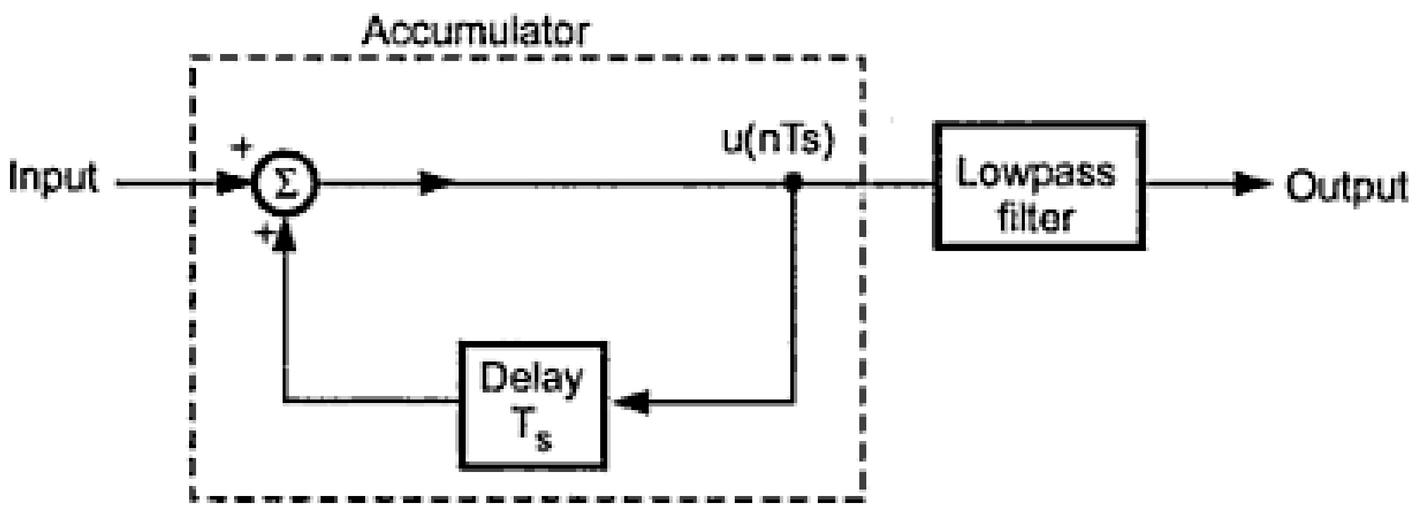
The previous sample approximation $u[(n-1)T_s]$ is restored by delaying one sample period T_s. The sampled input signal $x(nT_s)$ and staircase approximated signal $\hat{x}(nT_s)$ are subtracted to get error signal $e(nT_s)$.

Depending on the sign of $e(nT_s)$ one bit quantizer produces an output step of $+\delta$ or $-\delta$. If the step size is $+\delta$, then binary '1' is transmitted and if it is $-\delta$, then binary '0' is transmitted.



(a)



(b)

Fig. (a) Delta modulation transmitter and
(b) Delta modulation receiver

DM Receiver

At the receiver shown in Fig. (b), the accumulator and low-pass filter 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 $+\delta$ step to the previous output (which is delayed). If input is binary '0' then one step 'δ' is subtracted from the delayed signal. The low-pass filter has the cutoff frequency equal to highest frequency in $x(t)$. This filter smoothen the staircase signal to reconstruct $x(t)$.

NOISE IN DELTA MODULATION

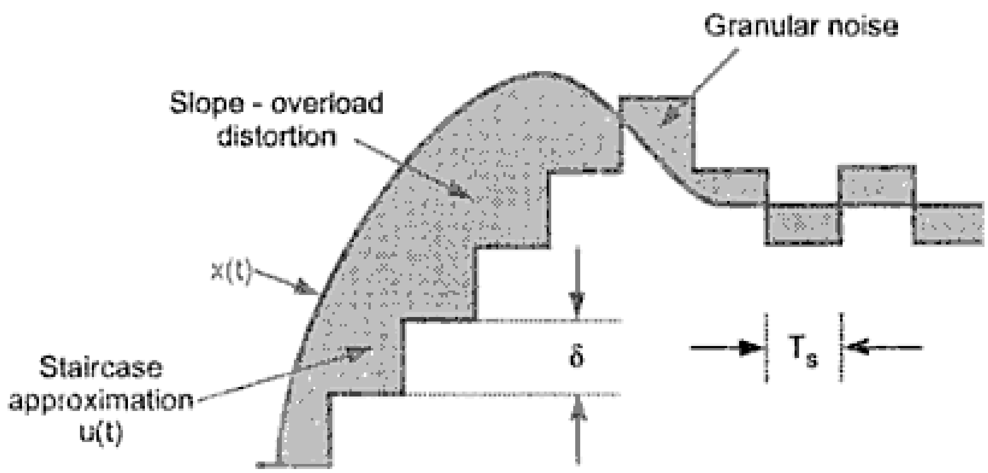


Fig. Quantization errors in delta modulation

Slope Overload Distortion (Startup Error)

This distortion arises because of the large dynamic range of the input signal. As can be seen from Fig. the rate of rise of input signal x(t) is so high that the staircase 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 the original input signal x(t). This error is called slope overload distortion. To reduce this error, the step size should be increased Since the step size of delta modulator remains fixed, its maximum or minimum slopes occur along straight lines. Therefore this modulator is also called Linear DeltaModulator (LDM).

Granular Noise

Granular noise occurs when the step size is too large compared to small variations in the input signal. That is for very small variations in the input signal, the staircase signal is changed by large amount, because of large step size. Fig shows that when the 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

errors.

Adaptive Delta Modulation(ADM).

Thus large step size is required to accommodate wide dynamic range of the input signal (to reduce slope overload distortion) and small steps are required to reduce granular noise. Adaptive delta modulation is the modification to overcome these

Operating Principle

To overcome the quantization errors due to slope overload and granular noise, the step size (Δ) is made adaptive to variations in the input signal x (t). Particularly in the steep segment of the signal x(t), the step size is increased. When the input is varying slowly, the step size is reduced. Then the method is called Adaptive Delta Modulation**(ADM).**

Transmitter and Receiver

Fig (a) shows the transmitter and (b) shows receiver of adaptive deltamodulator. The logic for step size control is added in the diagram. The step size when slope of signal of x(t) is high.increases or decreases according to certain rule depending on one bit quantizer output.

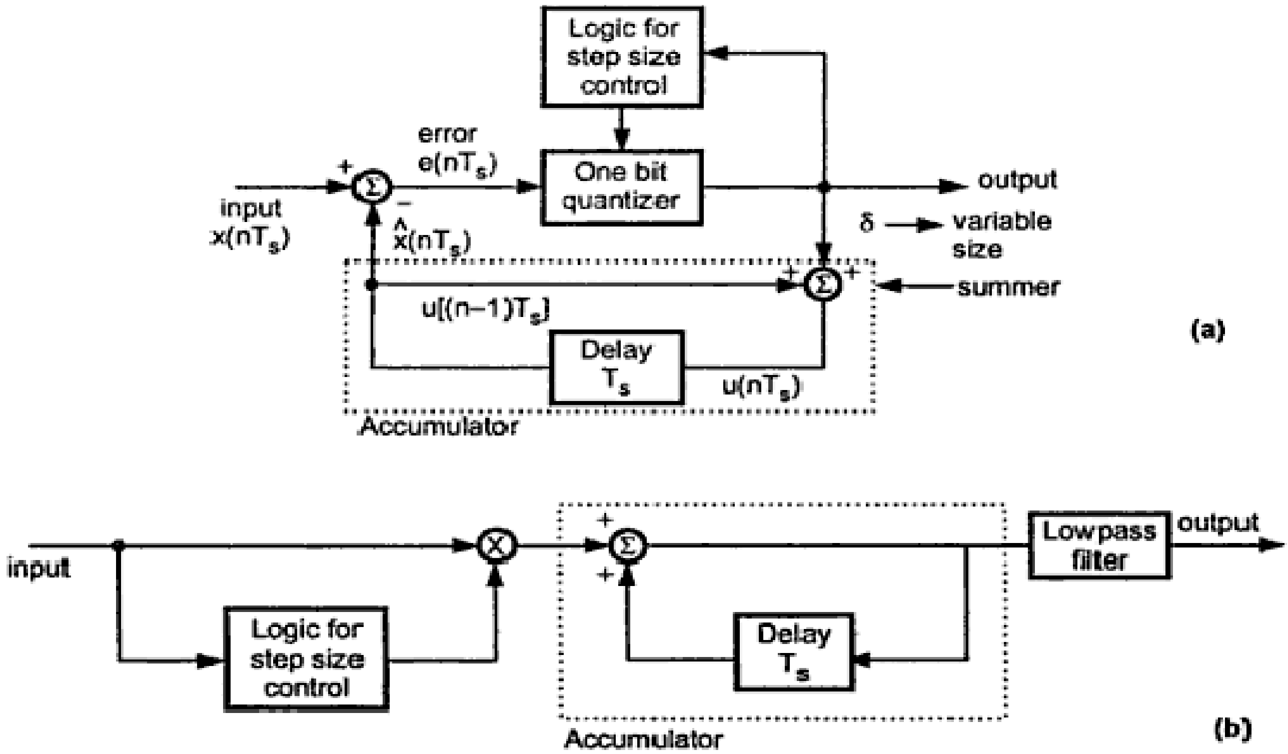


Fig. Adaptive delta modulator (a) Transmitter (b) Receiver

For example if one bit quantizer output is high (1), then step size may be doubled for next sample. If one bit quantizer output is low, then step size may be reduced by one step. Fig- shows the waveforms of adaptive delta modulator and sequence of bits transmitted.

In the receiver of adaptive delta modulator shown in Fig. (b) 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 decides the step size. It is then given to an accumulator which builds up staircase waveform. The low-pass filter then smoothens out the staircase waveform to reconstruct the smooth signal

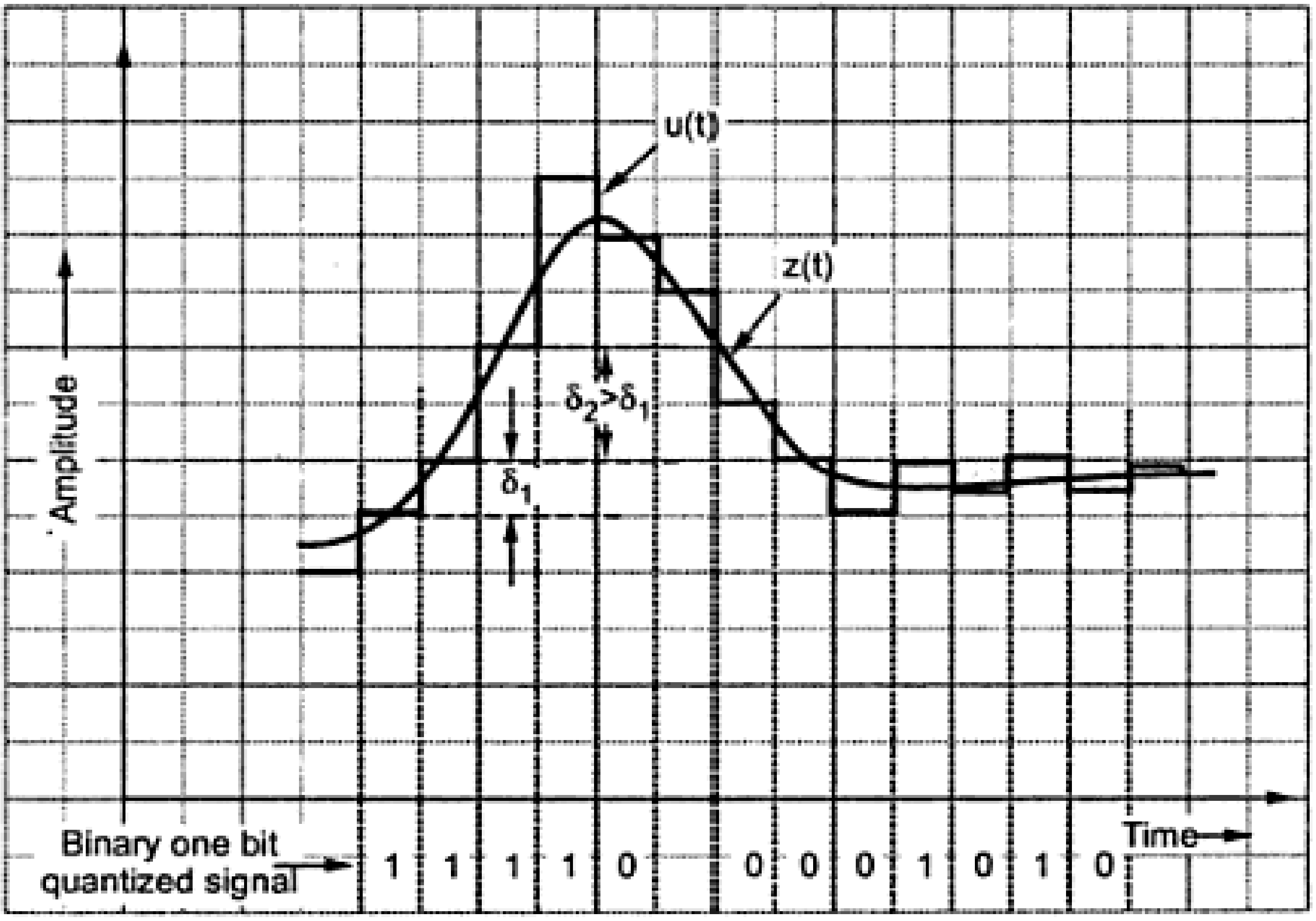


Fig. Waveforms of adaptive delta modulation