Module-2

Waveform Coding Techniques

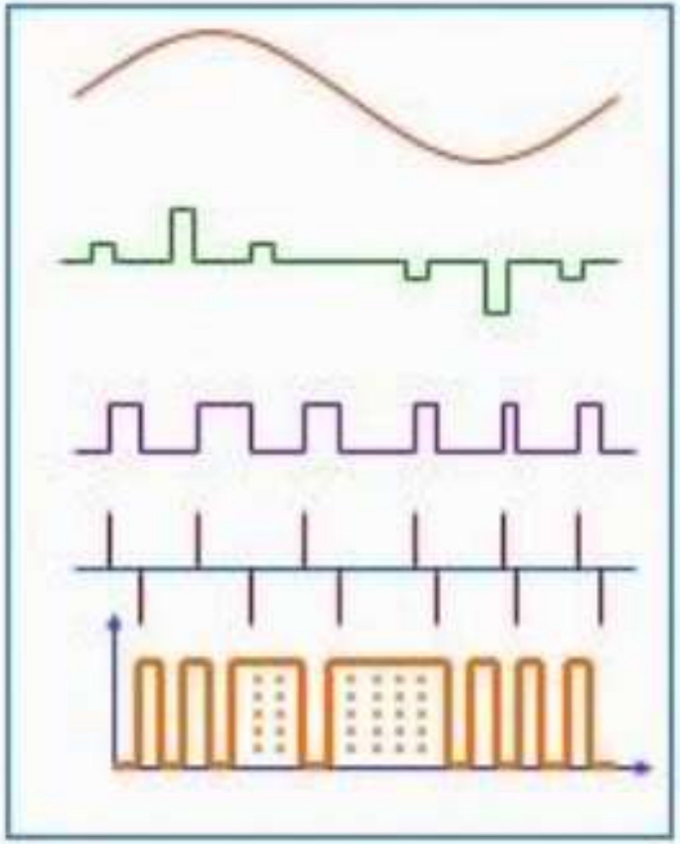
Pulse Code Modulation (PCM) - Uniform quantization, Quantization noise, Signal-to-Noise Ratio, Robust quantization. Differential pulse code modulation (DPCM), Delta Modulation (DM) - Quantization noise in DM, Adaptive Delta Modulation.

Types of Pulse Modulation

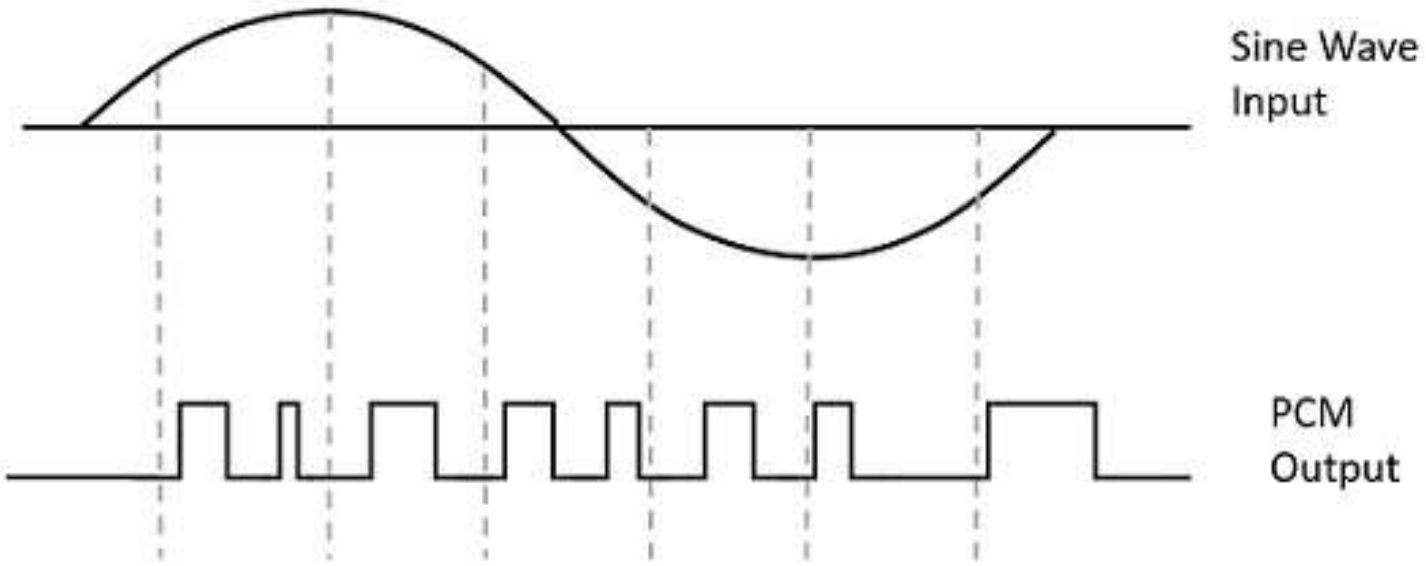
Pulse modulation is of two types

- o **Analog Pulse Modulation**
  - Pulse Amplitude Modulation (PAM)
  - Pulse width Modulation (PWM)
  - Pulse Position Modulation (PPM)
- o **Digital Pulse Modulation**
  - Pulse code Modulation (PCM)
  - Differential Pulse code Modulation(DPCM)
  - Delta Modulation (DM)
  - Adaptive Delta Modulation(ADM)

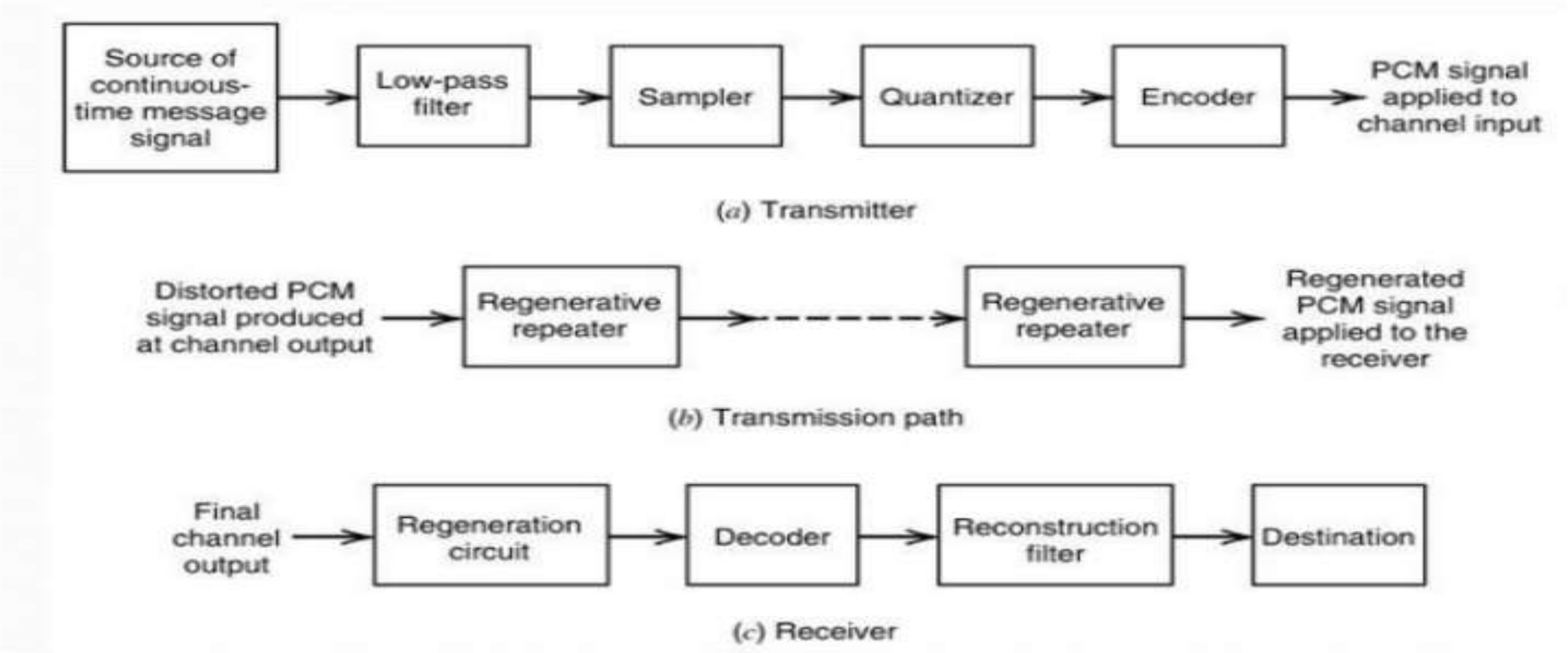
- Pulse Amplitude Modulation
- Pulse Width Modulation
- Pulse Position Modulation
- Pulse Code Modulation
- Delta Modulation



PCM Waveform



Basic Elements 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 **W** of the message signal, in accordance with the sampling theorem.

- Quantizer

  Quantizing is a process of reducing the excessive bits and confining the data. The sampled output when given to Quantizer, reduces the redundant bits and compresses the value.

- Encoder

  The digitization of analog signal is done by the encoder. It designates each quantized level by a binary code. The sampling done here is the sample-and-hold process. These three sections LPF, Sampler, and Quantizer will act as an analog to digital converter. Encoding minimizes the bandwidth used.

- Regenerative Repeater

This section increases the signal strength. The output of the channel also has one regenerative repeater circuit, to compensate the signal loss 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 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.

Sampling:

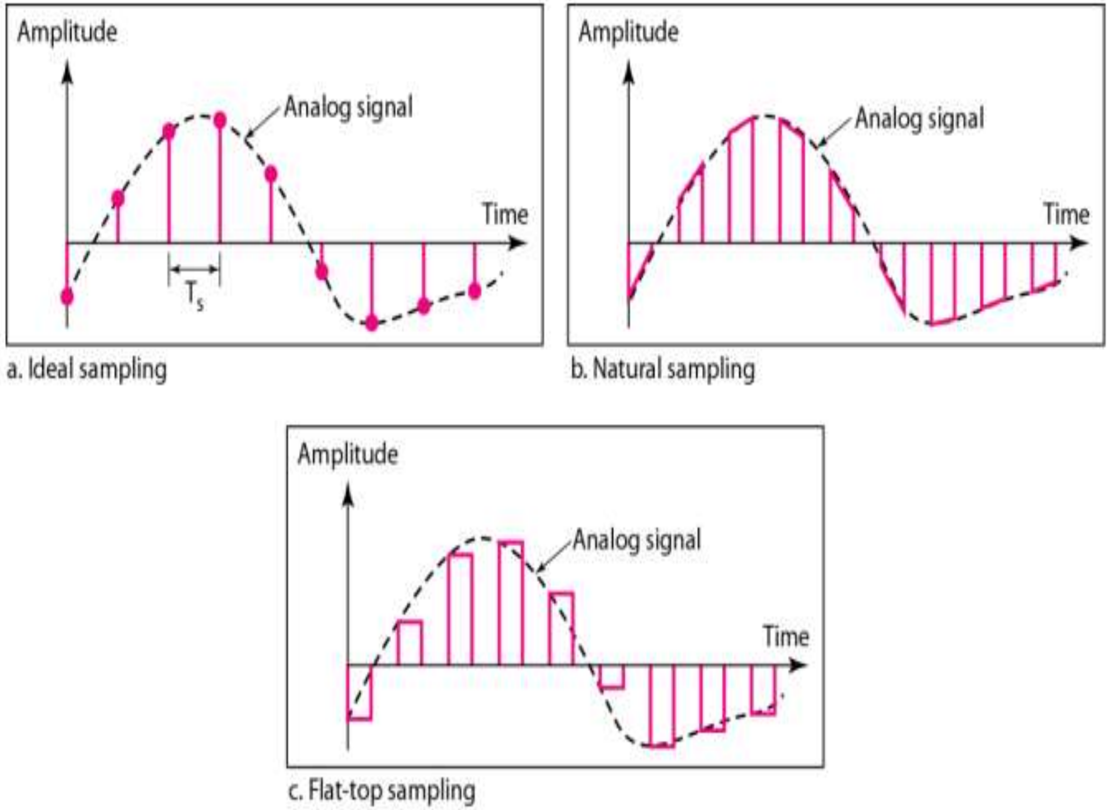
- Process of converting analog signal into discrete signal.
- Sampling is common in all pulse modulation techniques
- The signal is sampled at regular intervals such that each sample is proportional to amplitude of signal at that instant.
- Analog signal is sampled every $Ts\ Secs$, called sampling interval. fs=1/TS is called sampling rate or sampling frequency.

• For fs=2fm is Min. sampling rate called Nyquist rate. Sampled spectrum (ω) is repeating periodically without overlapping.

• Original spectrum is centered at ω=0 and having bandwidth of ωm. Spectrum can be recovered by passing through low pass filter with cut-off ωm. For fs<2fm sampled spectrum will overlap and cannot be recovered back. This is called aliasing.

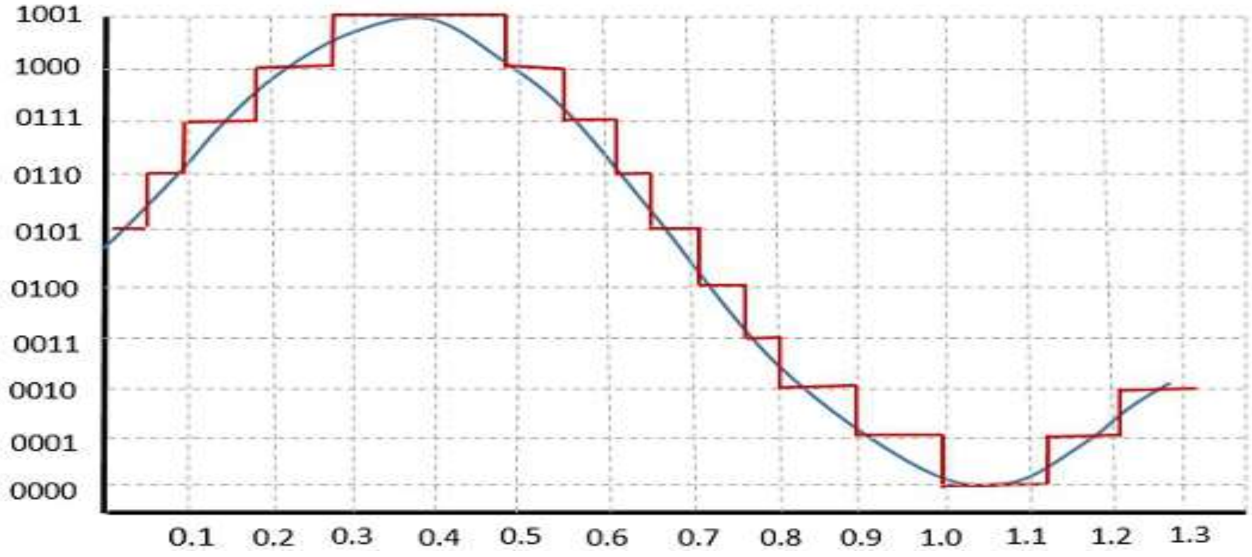
Sampling methods:

- Ideal – An impulse at each sampling instant.
- Natural – A pulse of Short width with varying amplitude.
- Flat Top – Uses sample and hold, like natural but with single amplitude value.



Quantization

- The digitization of analog signals involves the rounding off the values which are approximately equal to the analog values. The method of sampling chooses a few points on the analog signal and then these points are joined to round off the value to a near stabilized value. Such a process is called as **Quantization**.
- The quantizing of an analog signal is done by discretizing the signal with a number of quantization levels. **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** or **reconstruction levels**. The spacing between the two adjacent representation levels is called a **quantum** or **step-size**.

Types of Quantization

There are two types of Quantization –

Uniform Quantization and Non-uniform Quantization.

- The type of quantization in which the quantization levels are uniformly spaced is termed as a **Uniform Quantization**. The type of quantization in which the quantization levels are unequal and mostly the relation between them is logarithmic, is termed as a **Non-uniform Quantization**.
- There are two types of uniform quantization. They are Mid-Rise type and Mid-Tread type.

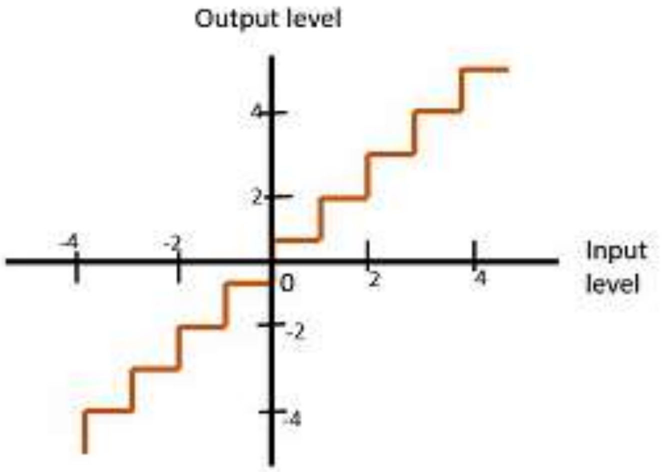


Fig 1 : Mid-Rise type Uniform Quantization

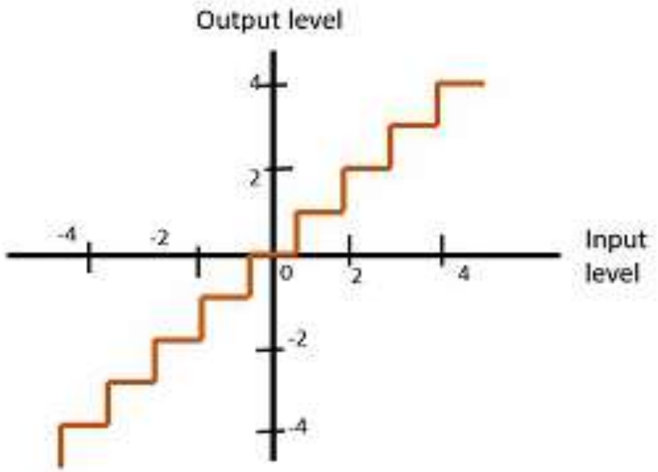


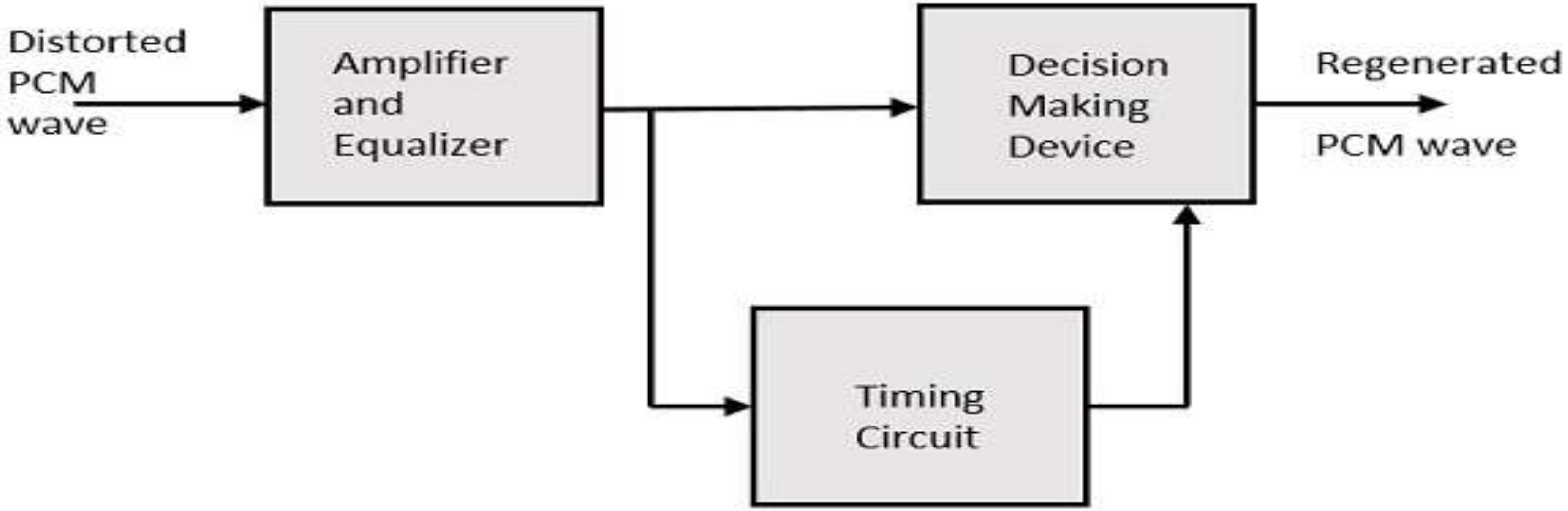
Fig 2 : Mid-Tread type Uniform Quantization

•The **Mid-Rise** type is so called because the origin lies in the middle of a raising part of the stair-case 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 quantizers are symmetric about the origin.

Regenerative repeater



Block diagram of a regenerative repeater

- Equalizer

The channel produces amplitude and phase distortions to the signals. This is due to the transmission characteristics of the channel. The Equalizer circuit compensates these losses by shaping the received pulses.

- Timing Circuit

To obtain a quality output, the sampling of the pulses should be done where the signal to noise ratio SNR is maximum.

To achieve this perfect sampling, a periodic pulse train has to be derived from the received pulses, which is done by the timing circuit.

Hence, the timing circuit, allots the timing interval for sampling at high SNR, through the received pulses.

Decision Device

- The timing circuit determines the sampling times. The decision device is enabled at these sampling times. The decision device decides its output based on whether the amplitude of the quantized pulse and the noise, exceeds a pre-determined value or not.

Robust quantization(Non Uniform Quantization)

- If the quantizer charateristics is nonlinear and the step size is not constant instead if it is variable,dependent on the amplitude of input signal the quantization is known as non uniform quantization.
- The step size is small at low input signal levels. Hence quantization error is also small at these input.The step size is higher at high input levels.

Necessity of Non uniform quantization

In uniform quantization,the step size remains same throughout the range of quantizer. Therefore over the complete range of inputs,the maximum error also remains same.

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

The step size is defined by, $\Delta = \frac{2x_{max}}{q}$

If x(t) is normalized, its maximum value i.e., $x_{max}=1$

The number of levels q will be,

$Q=2^v$;v= number of bits

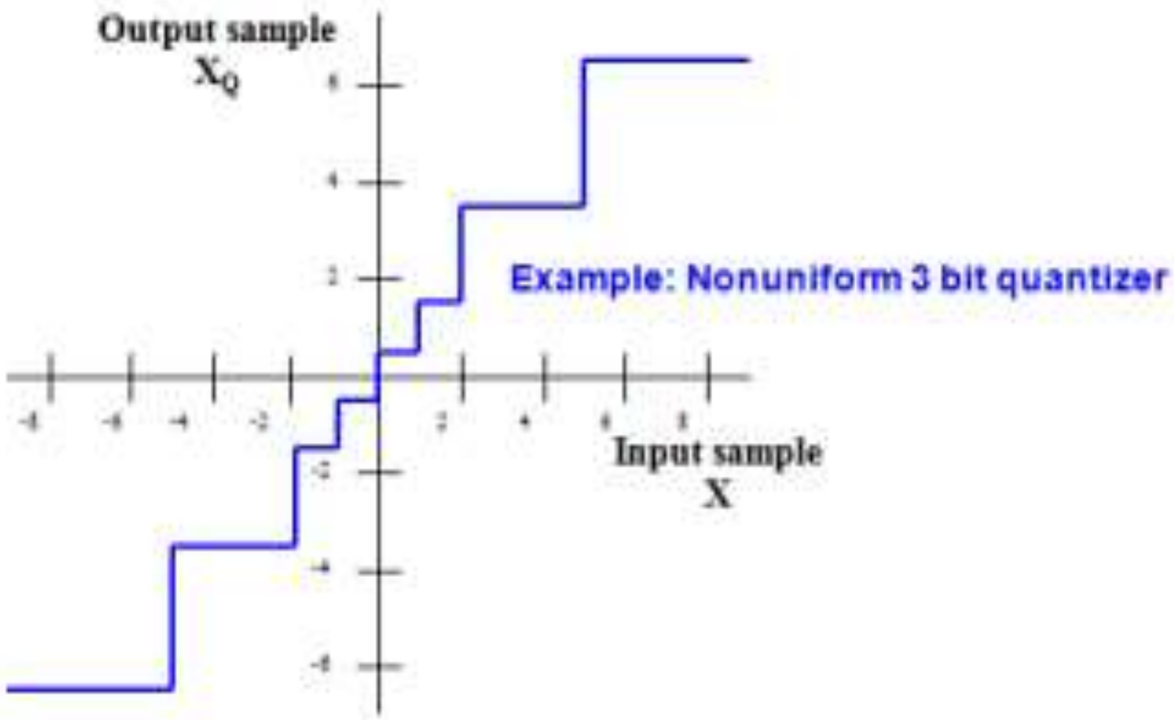
For example ; If v=4bits

$Q=2^4=16$levels

$\Delta=\frac{1}{8}$; $\varepsilon_{max}=\left|\frac{\Delta}{2}\right|=\frac{1}{16}$

Hence note that here the quantization error is $\frac{1}{16}^{th}$ part of the full voltage range. For simplicity we assume that full range voltage is 16volts.Then maximum quantization error will be 1volts.However for the low signal amplitudes like 2volts,3volts,the maximum quantization error of 1volt is quite large.This means that for signal amplitude which are close to 15volts,16volts,the quantization error of 1volt can be considered to be small.This problem in uniform quantization will be overcome by nonuniform quantization

Non uniform Quantizer



Companding PCM System

- Non-uniform quantizers are difficult to make and expensive.
- The use of a non – uniform quantizer is equivalent to passing the baseband signal through a compressor and then applying the compressed signal to a uniform quantizer.
- The strong signals are attenuated and the weak signals are amplified before applying them to a uniform quantizer.
- The input to the quantizer will have a more uniform distribution.
- Expander is used to restore the signal samples to their correct relative level.
- The compression of signal at the transmitter and expansion at the receiver is combined to constitute companding.

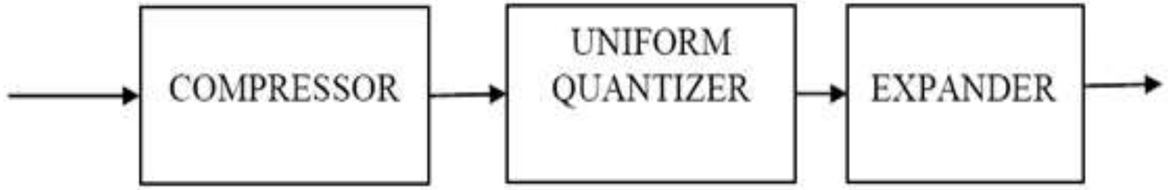


Fig: Model of Non uniform quantization

Companding Law

There are two popular standards for non-linear quantization known as

(a) The μ - law companding

(b) The A – law companding.

(a) μ - law companding

In the μ law companding, the compressor characteristics c(x) is continuous .It is approximately linear for smaller values of input levels and the logarithmic for high input levels. It is described by

$$\frac{c|x|}{x_{max}} = \frac{\ln\left(1+\frac{\mu|x|}{x_{max}}\right)}{\ln(1+\mu)} \qquad 0 \leq \frac{|x|}{x_{max}} \leq 1$$

c(x)---represents the output
x---is the input to the compressor

Also $\frac{|x|}{x_{max}}$----- represents the normalized value of input and x_{max} is peak amplitude value.

- Special case of uniform quantization μ=0
- Practical value of μ=255
- The μ law is used for PCM telephone systems in the united states, Canada and Japan
- μ law companding is used for speech and music signals

(b)A-law Companding

In the A law companding, the compressor characteristics is piecewise, made up of a linear segment for low-level inputs and a logarithmic segment for high level inputs, It is described by

$$\frac{c|x|}{x_{max}} = \frac{\frac{A\,|x|}{x_{max}})}{1+lnA} \qquad 0 \leq \frac{|x|}{x_{max}} \leq \frac{1}{A}$$

$$\frac{c|x|}{x_{max}} = \frac{(1+\ln\left(\frac{A|x|}{x_{max}}\right)}{1+\ln A} \quad \frac{1}{A} \leq \frac{|x|}{x_{max}} \leq 1$$

- Special case of uniform quantization A=1
- Practical value A=87.5
- A-law is used for PCM telephone systems in Europe.

Advantages of PCM

- Robustness to channel noise and interference
- Efficient regeneration of the coded signal along the transmission path.
- High signal –to-noise ratio (SNR).
- A uniform format for the transmission of difference kinds of baseband signals causes their integration with other forms of digital data in a communication network.
- Secure communication through the use f special modulation and encryption.

Limitations of PCM

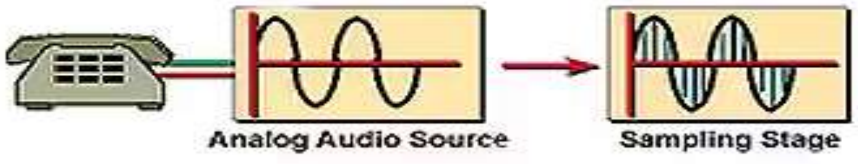
- 1. Increased system complexity.
- 2. Increased channel bandwidth.

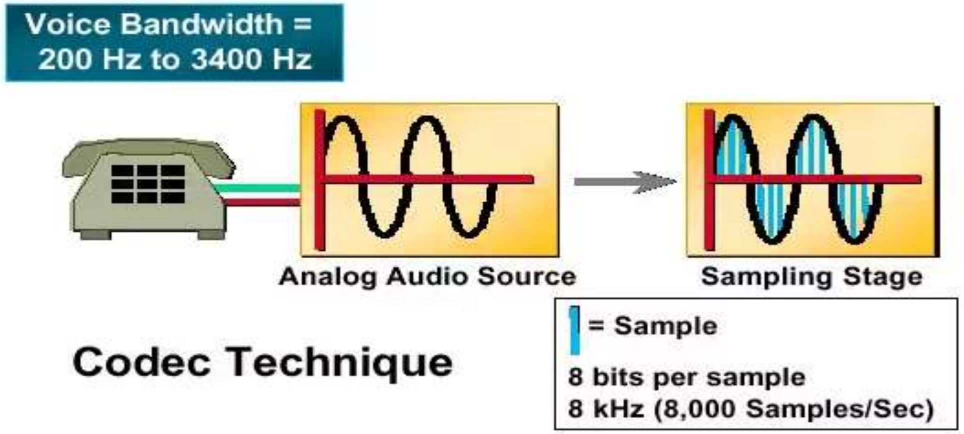
Channel bandwidth requirement can be reduced by reducing the bit rate. The use of sophisticated data compression techniques remove the redundancy inherently present in a PCM signal and thereby bit rate of transmitted data is reduced without serious degradation in system performance.

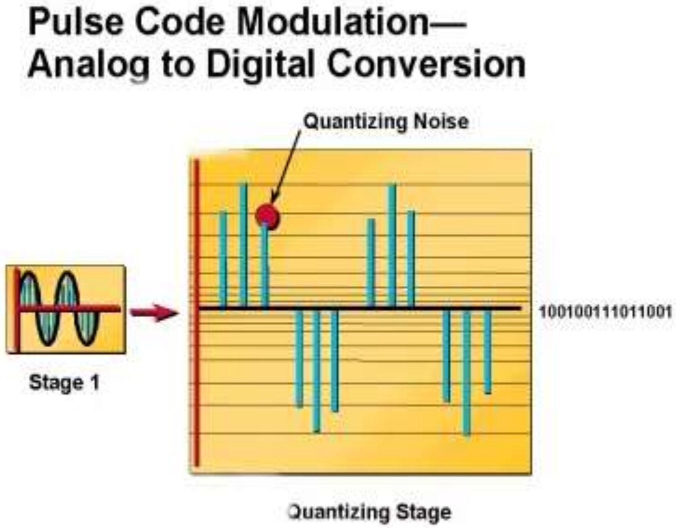
Applications

1. In telephony (with the advent of fiber optic cables)
2. In the space communication, space craft transmits signals to earth.
3. The compact disc (CD) is a recent application of PCM.

PCM in ITU-G.711 specification







- **Digitize Voice**

  After filter and sample (using PAM) an input analog voice signal, the next step is to digitize these samples in preparation for transmission over a Telephony network. The process of digitizing analog voice signals is called PCM. The only difference between PAM and PCM is that PCM takes the process one step further. PCM decodes each analog sample using binary code words. PCM has an analog-to-digital converter on the source side and a digital-to-analog converter on the destination side. PCM uses a technique called quantization to encode these samples.

- **Quantization and Coding**

Quantization is the process of converting each analog sample value into a discrete value that can be assigned a unique digital code word.

As the input signal samples enter the quantization phase, they are assigned to a quantization interval. All quantization intervals are equally spaced (uniform quantization) throughout the dynamic range of the input analog signal. Each quantization interval is assigned a discrete value in the form of a binary code word.

The standard word size used is eight bits. If an input analog signal is sampled 8000 times per second and each sample is given a code word that is eight bits long, then the maximum transmission bit rate for Telephony systems using PCM is 64,000 bits per second.

- Each input sample is assigned a quantization interval that is closest to its amplitude height. If an input sample is not assigned a quantization interval that matches its actual height, then an error is introduced into the PCM process. This error is called quantization noise. Quantization noise is equivalent to the random noise that impacts the signal-to-noise ratio (SNR) of a voice signal. SNR is a measure of signal strength relative to background noise.
- The ratio is usually measured in decibels (dB). If the incoming signal strength in microvolts is Vs, and the noise level, also in microvolts, is Vn, then the signal-to-noise ratio, S/N, in decibels is given by the formula $S/N = 20 \log10(Vs/Vn)$. SNR is measured in decibels (dB).
- The higher the SNR, the better the voice quality. Quantization noise reduces the SNR of a signal. Therefore, an increase in quantization noise degrades the quality of a voice signal.

- One way to reduce quantization noise is to increase the amount of quantization intervals. The difference between the input signal amplitude height and the quantization interval decreases as the quantization intervals are increased **(increases in the intervals decrease the quantization noise**). However, the amount of code words also need to be increased in proportion to the increase in quantization intervals. This process introduces additional problems that deal with the capacity of a PCM system to handle more code words.
- SNR (including quantization noise) is the single most important factor that affects voice quality in uniform quantization. Uniform quantization uses equal quantization levels throughout the entire dynamic range of an input analog signal.
- Therefore, low signals have a small SNR (low-signal-level voice quality) and high signals have a large SNR (high-signal-level voice quality). Since most voice signals generated are of the low kind, having better voice quality at higher signal levels is a very inefficient way of digitizing voice signals. To improve voice quality at lower signal levels, uniform quantization (uniform PCM) is replaced by a non uniform quantization process called companding.

Refer notes for SNR

DPCM(Differential Pulse code Modulation)

- For the samples that are highly correlated, when encoded by PCM technique, leave redundant information behind.
- To process this redundant information and to have a better output, it is a wise decision to take a predicted sampled value, assumed from its previous output and summarize them with the quantized values. Such a process is called as Differential PCM (DPCM) technique.

- x(t) is a continuous time signal,it is sampled using the flat top sampling technique with the highest nyquist rate.The samples are encoded by 3 bit PCM.
- The signals are encoded to the nearest digital level (shown in circle).The encoded bits are written on top of samples.
- The encoded bits at 4Ts,5Ts,6Ts are having same value(110).this information can be transmit by one sample.But three samples carrying same information ,this is redundant information.
- Samples taken at 9Ts,10Ts.The difference between these samples is only due to last bit and first two bits are reduntant.

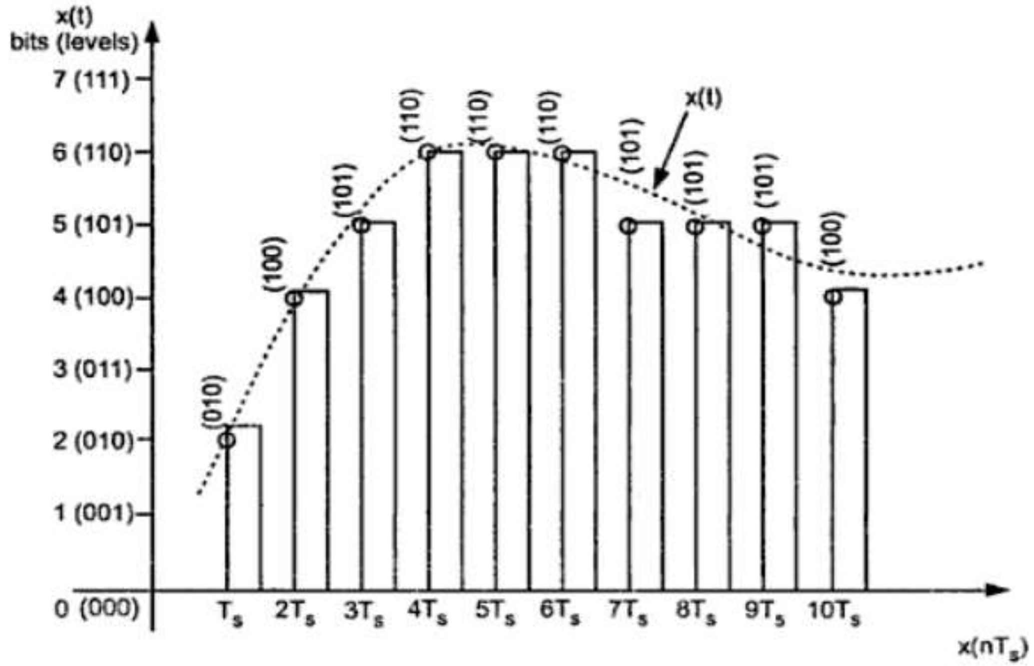


Fig. Redundant information in PCM

- If redundancy is reduced, overall bit rate will decreased and number of bits required to transmit one sample is also reduced. This type of digital modulation technique is called Differential Pulse code modulation.
- This is works on the principle of prediction filter. The value of present sample is predicted from the past sample. Prediction is not exact but it is close to the sample.

DPCM Transmitter

- The DPCM Transmitter consists of Quantizer and Predictor with two summer circuits.

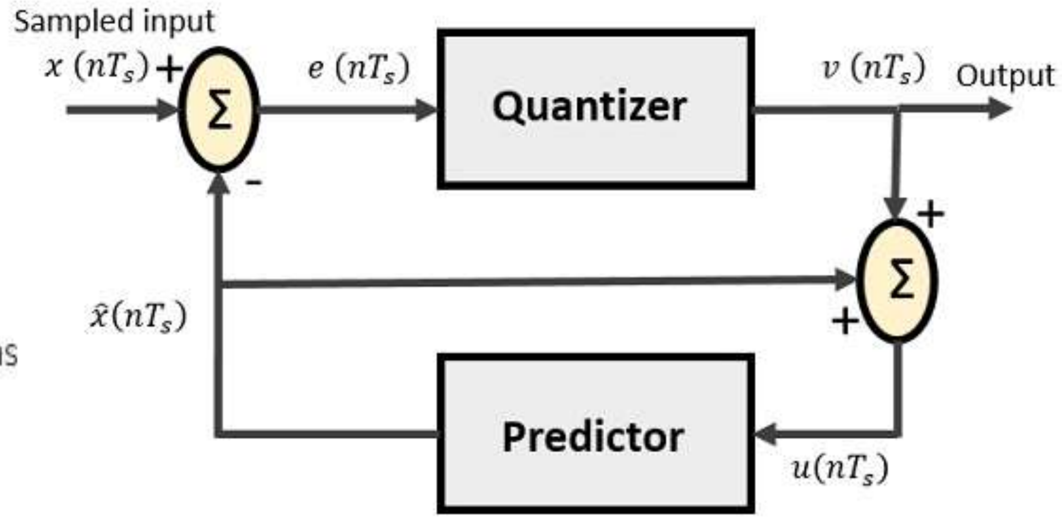
$x(nT_s)$ is the sampled input

$\hat{x}(nT_s)$ is the predicted sample

$e(nT_s)$ is the difference of sampled input and predicted output, often called as prediction error

$v(nT_s)$ is the quantized output

$u(nT_s)$ is the predictor input which is actually the summer output of the predictor output and the quantizer output



The predictor produces the assumed samples from the previous outputs of the transmitter circuit. The input to this predictor is the quantized versions of the input signal $x(nT_s)$.

Quantizer Output is represented as –

$$v(nT_s) = Q[e(nT_s)]$$

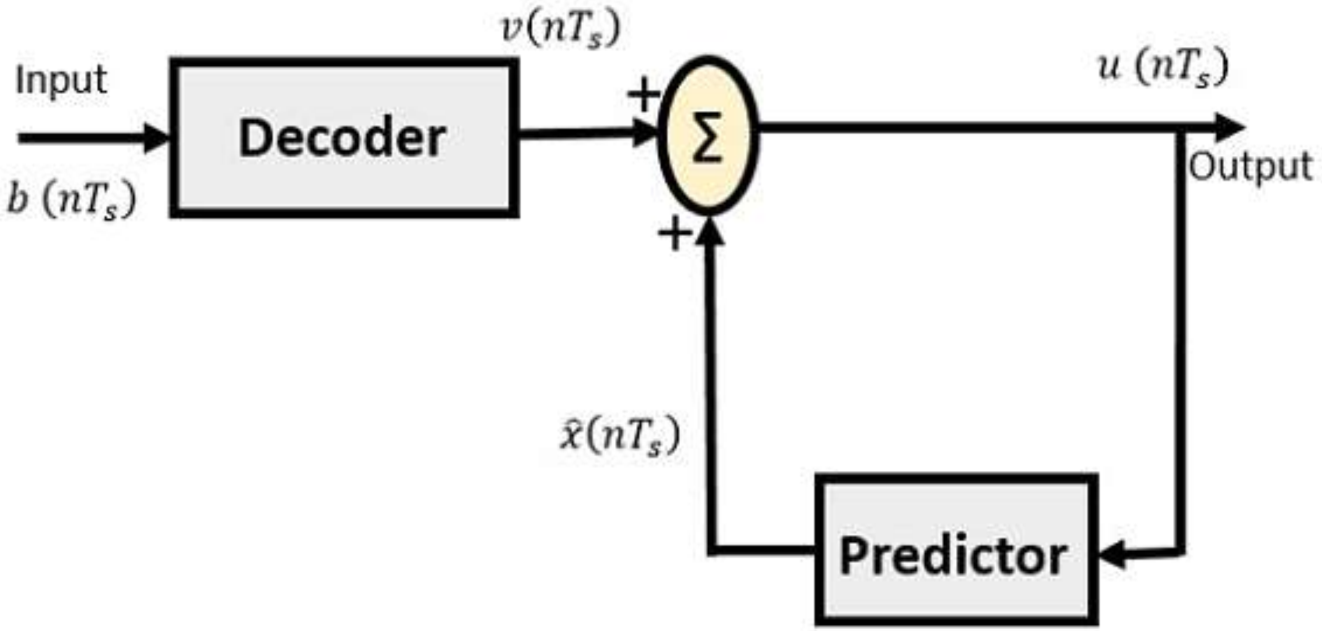
$$= e(nT_s) + q(nT_s)$$ Where **q (nT$_s$)** is the quantization error

Predictor input is the sum of quantizer output and predictor output,

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

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

DPCM-Receiver



SNR

The output signal –to- quantization noise ratio of a signal coder ,

$$SNR_o = \frac{\sigma x2}{\sigma Q2}$$

σ_x^2=Variance of the original input x(n T_s)
σ_Q^2=Variance of the quantization error q(n T_s)

$$SNR_o = (\sigma_x^2/\sigma_E^2)(\sigma_E^2/\sigma_Q^2)$$
$$= G_p(SNR)_p$$

Where
$(SNR)_p$=Prediction error-to-quantization noise ratio $(SNR)_p=(\sigma_E^2/\sigma_Q^2)$
G_p=Prediction gain $G_p==(\sigma_x^2/\sigma_E^2)$

Quantity G_p,When greater than unity,represents the gain in signal-to-noise ratio that is due to the differential quantization scheme.For a given base-band signal,the variance σ_x^2 is fixed,so that G_p is maximized by minimizing the variance σ_E^2 of the prediction error e(n T_s).

Consider the input samples x(nTs)={2.1,2.2,2.3,2.6,2.7,2.8}.Explain how encoding and decoding done in DPCM,Assume first order prediction filter $\hat{x}(nT_s)$ =xq(n-1).

DM(Delta Modulation)

- Delta Modulation is the one-bit (or two-level) versions of DPCM. DM provides a staircase approximation to the over sampled version of an input base band signal.
- The difference between the input and the approximation is quantized into only two levels, namely, $\pm\delta$ corresponding to positive and negative differences, respectively

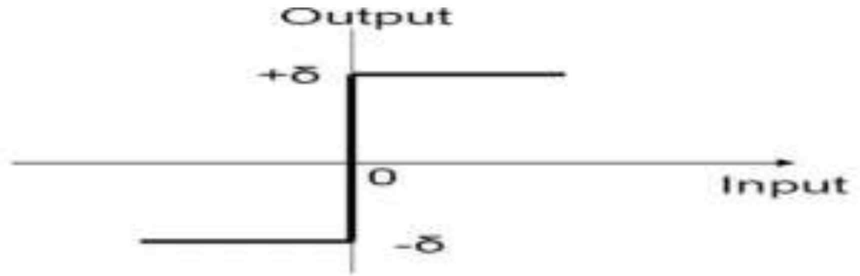


Fig Input-output characteristics of Delta Modulation

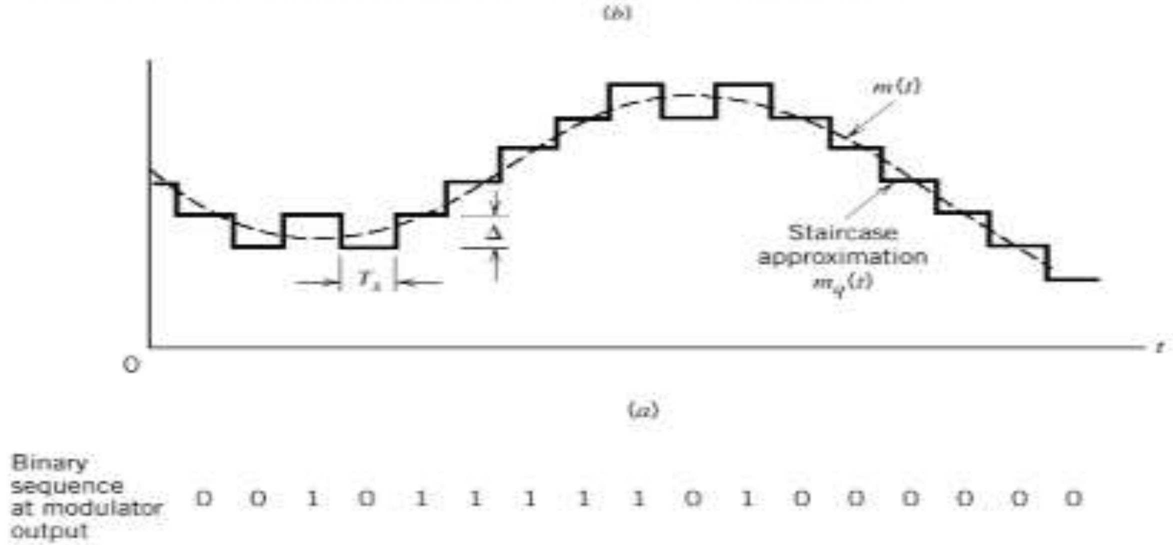
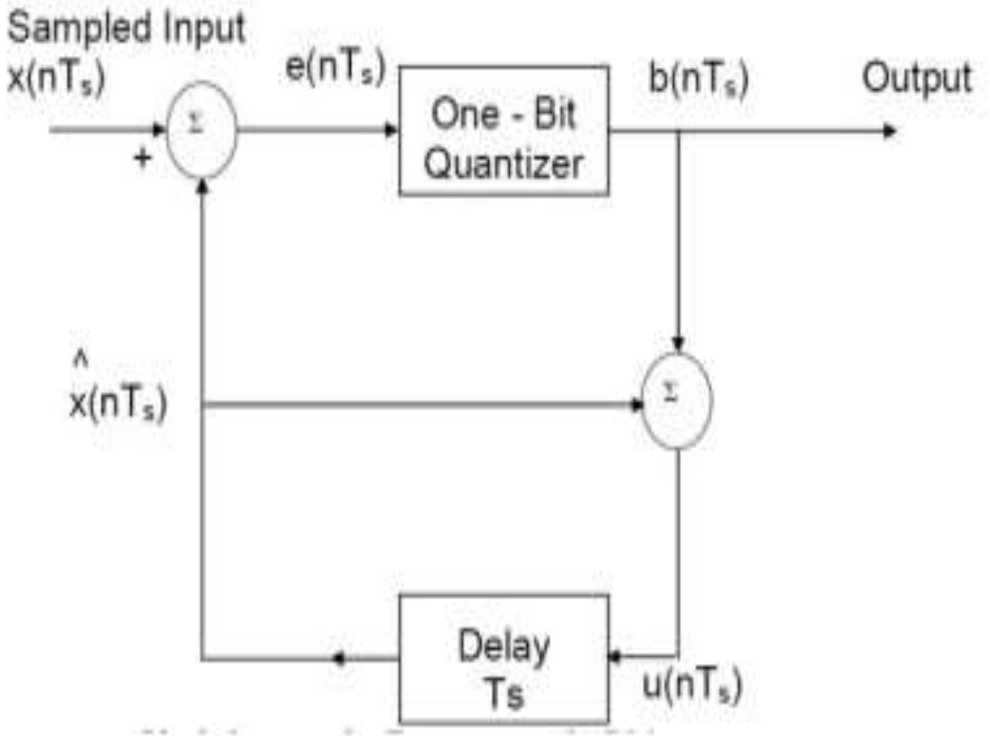


Fig Delta Modulation waveform

DM Transmitter

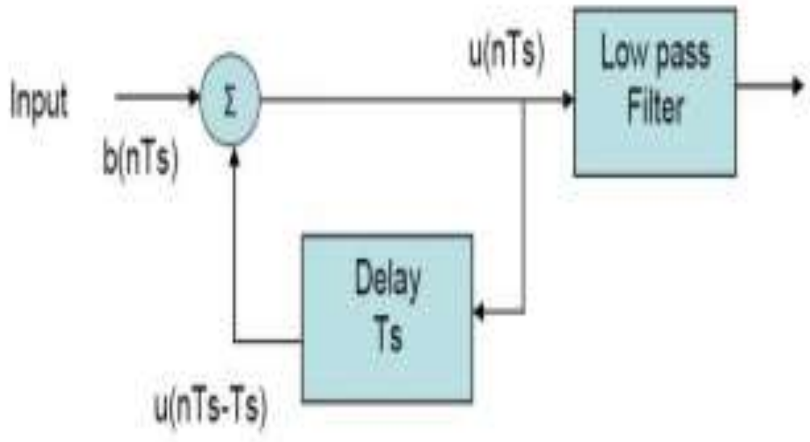
$$e(nT_s)=x(nT_s)-\hat{x}(nT_s)$$
$$=x(nT_s)-u(nT_s-T_s) \qquad (1)$$
$$b(nT_s)=\delta\, sgn[e(nT_s)] \qquad (2)$$
$$u(nT_s)=u(nT_s-T_s)+b(nT_s) \qquad (3)$$

$$u(nT_s)=\delta \sum_{i=1}^{n} sgn[e(iT_s)]$$
$$= \sum_{i=1}^{n} b[(iT_s)] \qquad (4)$$



DM Receiver

- In the receiver the stair case approximation u(t) is reconstructed by passing the incoming sequence of positive and negative pulses through an accumulator in a manner similar to that used in the transmitter.
- The out-of –band quantization noise in the high frequency staircase waveform u(t) is rejected by passing it through a low-pass filter with a band-width equal to the original signal bandwidth.



Quantization Noise in DM

QUANTIZATION NOISE

Delta modulation systems are subject to two types of quantization error:

(1) Slope overload distortion (2) Granular noise

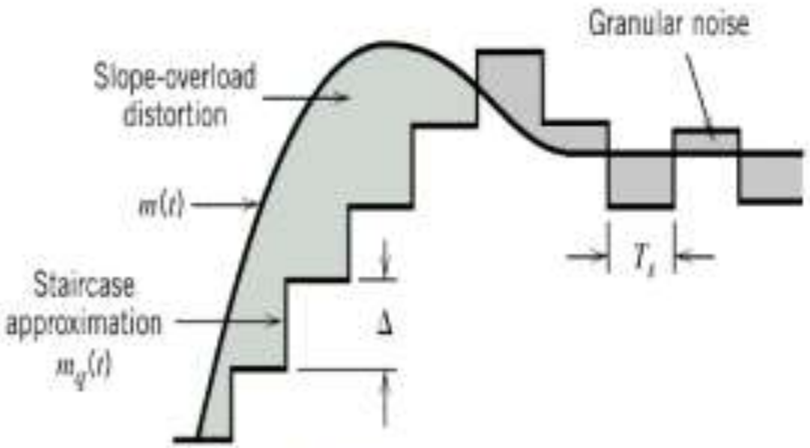


Fig Quantization error in DM

Slope Overload Noise

- The step size is too small for the staircase approximation u(t) to follow a steep segment of the input waveform x(t),with the result that u(t) falls behind x(t) .This condition is called slope-overload, and the resulting quantization error is called slope-overload distortion(noise).
- Since the maximum slope of the staircase approximation u(t) is fixed by the step size , increases and decreases in u(t) tend to occur along straight lines.

Granular Noise

• The granular noise occurs when the step size is too large relative to the local slope characteristics of the input wave form x(t), thereby causing the staircase approximation u(t) to hunt around a relatively flat segment of the input waveform; The granular noise is analogous to quantization noise in a PCM system.

Advantages of DM

- Since ,the delta modulation transmits only one bit for one sample, therefore the signaling rate and transmission channel bandwidth is quite small for delta modulation compared to PCM.
- The transmitter and receiver implementation is very much simple for delta modulation. There is no analog to digital converter requires in DM.

Disadvantages of DM

- Slope overload distortion
- Granular noise or idle noise

Refer notes for SNR

ADM(Adaptive delta Modulation)

- The choice of optimum step size that minimize the mean square value of quantization error.
- To satisfy such a requirement, the modulator must be made (adaptive) in the sense that the step size is made to vary in accordance with the input signal.
- Adaptive delta modulation is a delta modulation where the step size is automatically varied depending on the amplitude characteristics of the analog input signal.
- The performance of a delta modulator can be improved significantly by making the step size of the modulator assume a time-varying form.
- In particular, during a steep segment of the input signal the step size is increased. Conversely, when the input signal is varying slowly, the step size is reduced.
- In this way, the size is adapted to the level of the input signal. The resulting method is called adaptive delta modulation (ADM). There are several types of ADM, depending on the type of scheme used for adjusting the step size. In this ADM, a discrete set of values is provided for the step size.

ADM Transmitter and Receiver

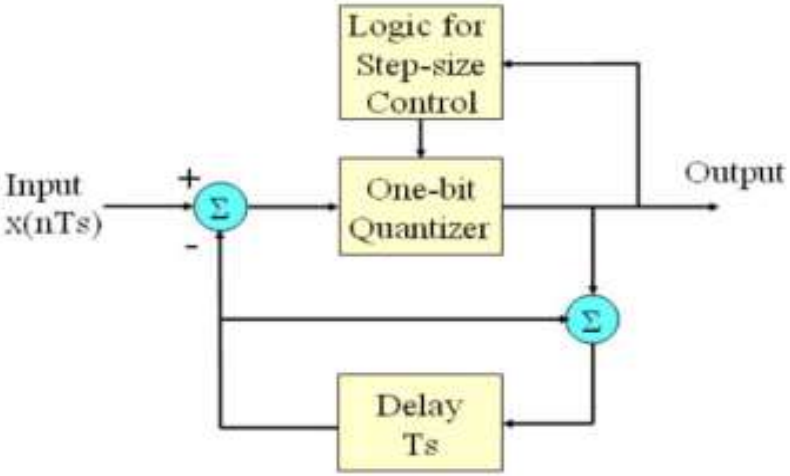


Fig:ADM Transmitter

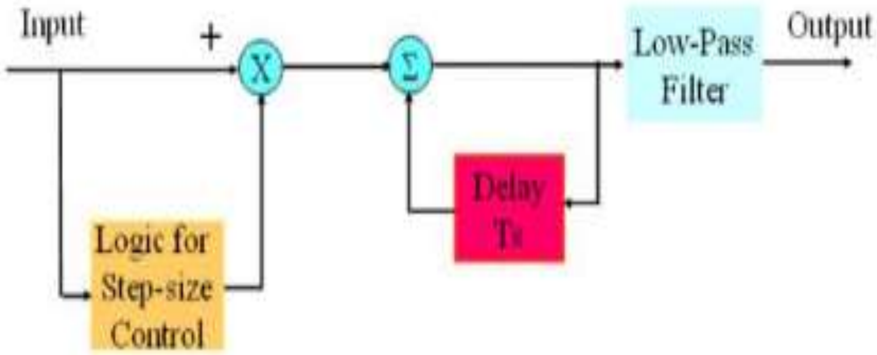


Fig :ADM Receiver

Advantages of Adaptive Delta Modulation

1.The signal to noise ratio becomes better than ordinary delta modulation because of the reduction in slope overload distortion and idle noise.

2.Because of the variable step size, the dynamic range of ADM is wider than simple DM

3.Utilization of bandwidth is better than DM.

Practical Applications of Adaptive Delta Modulation

- The following are some of the uses of the adaptive modulation method
- This modulation is utilized in systems that demand higher wireless voice quality as well as faster data transmission.
- This modulation method is utilized in television signal transmission.
- In speech coding, this modulation approach is utilized.
- NASA also uses this modulation as a standard for all communications between mission control and spacecraft.