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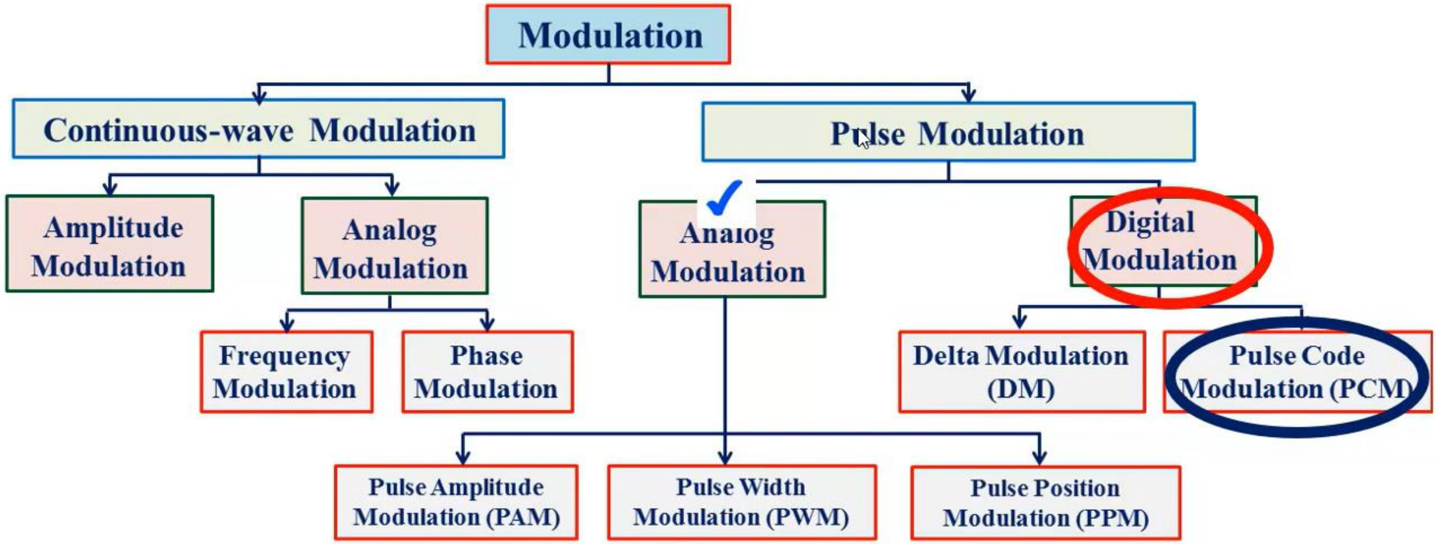
DIGITAL COMMUNICATION

- Prof. N. B. Kanirkar

- PCM – Pulse Code Modulation
- DM – Delta Modulation

Pulse Code Modulation

Types of Modulation



PCM

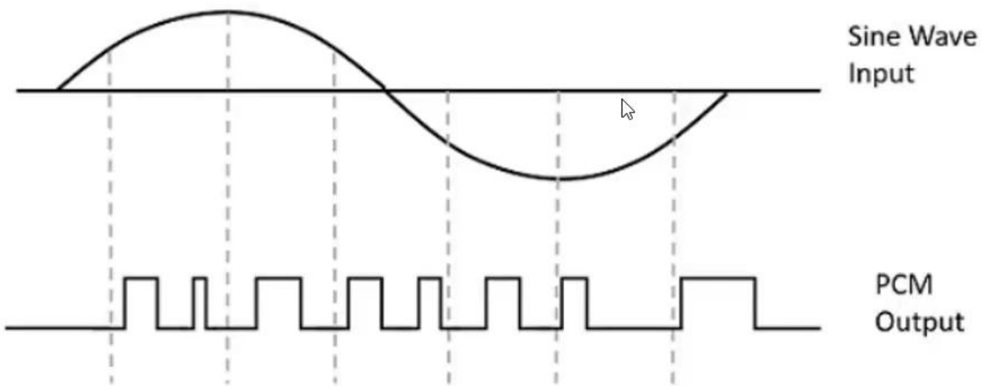
Pulse-code modulation (**PCM**) is a method used to digitally represent sampled analog signals.

It is the standard form of digital audio in computers, compact discs, digital telephony and other digital audio applications.

In a PCM stream, the amplitude of the analog signal is sampled regularly at uniform intervals, and each sample is quantized to the nearest value within a range of digital steps.

Pulse Code Modulation

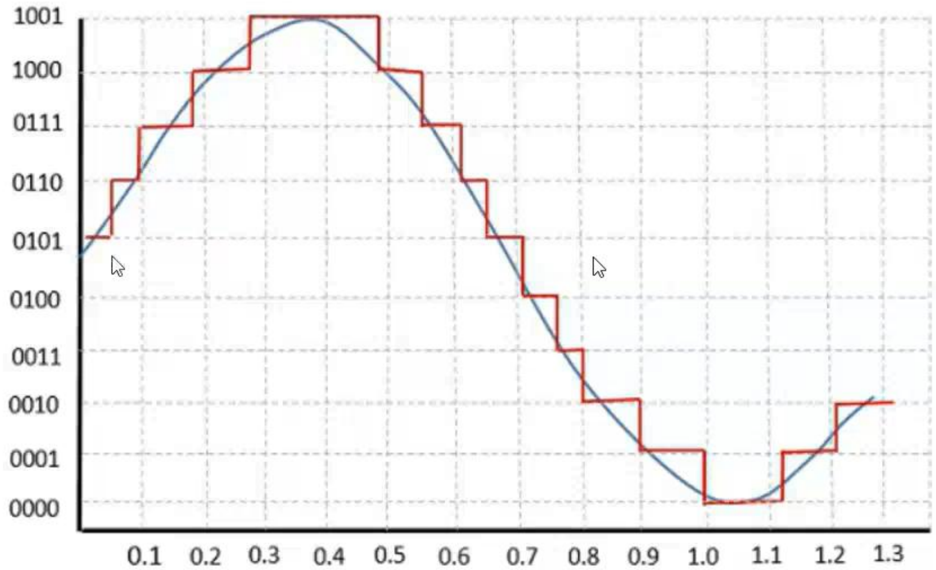
A signal is Pulse Code modulated to convert its analog information into a binary sequence, i.e., 1s and 0s. The output of a **Pulse Code Modulation (PCM)** will resemble a binary sequence. The following figure shows an example of PCM output with respect to instantaneous values of a given sine wave.



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 Pulse Code Modulation, 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.

The following figure shows how an analog signal gets quantized. The blue line represents analog signal while the red one represents the quantized signal.



Both sampling and quantization results 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 two adjacent representation levels is called a **quantum** or **step-size**.

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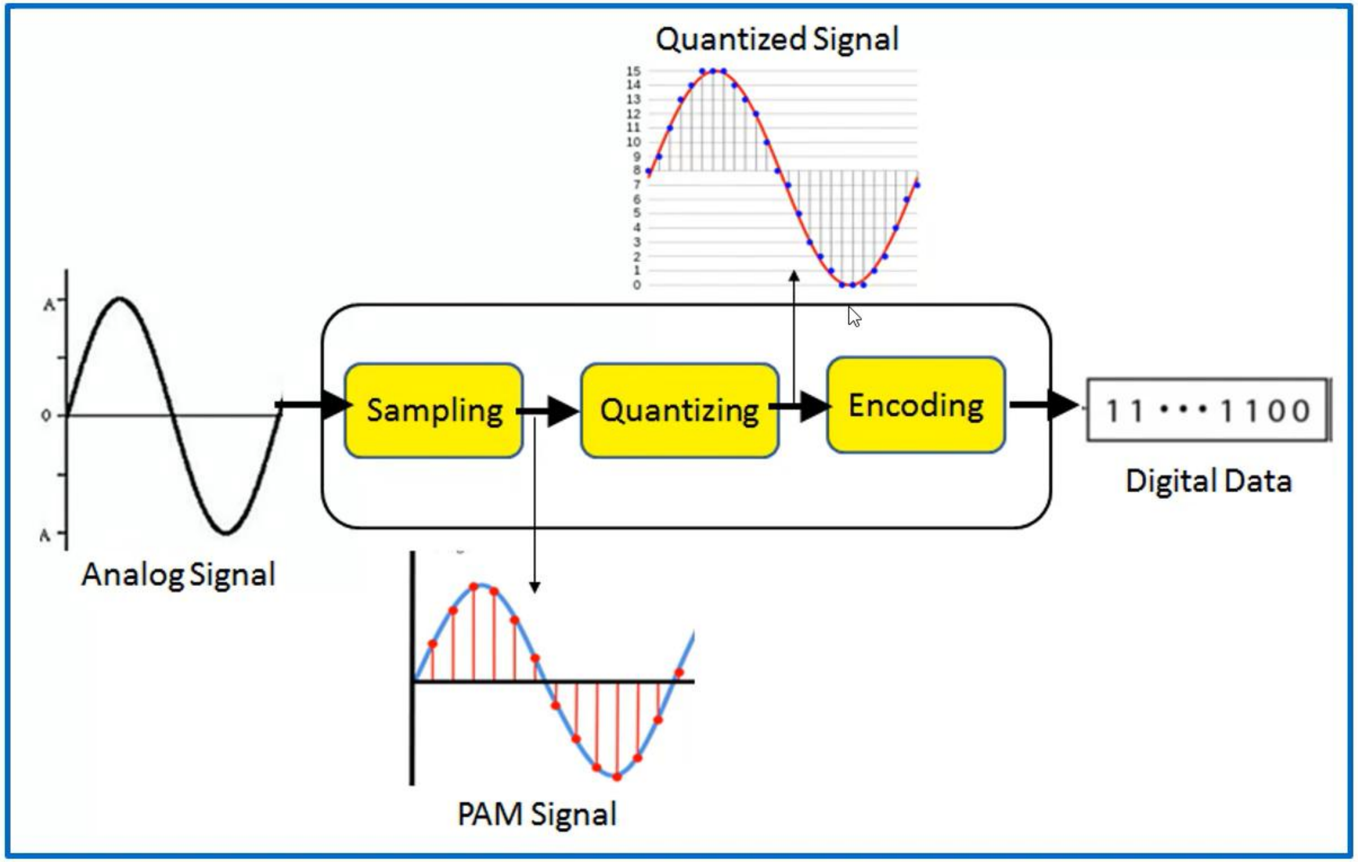
Pulse Code Modulation (PCM)

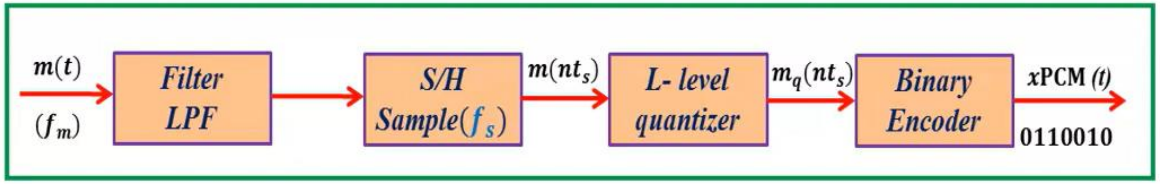


Pulse Code Modulation (PCM)

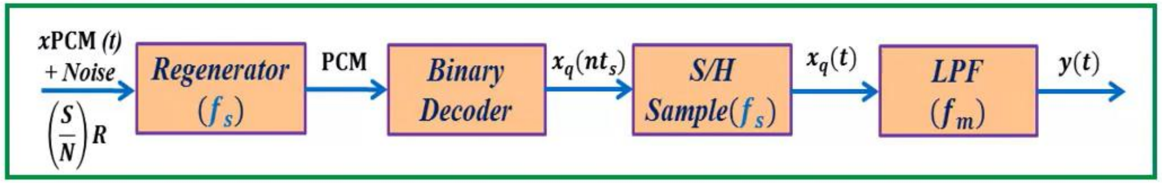
Most current digital audio systems (computers, compact discs, digital telephony etc.) use multi-bit Pulse Code Modulation (PCM) to represent the sound signal. Analog to PCM conversion consists of three steps: Sampling, Quantizing and Encoding.

Block Diagram of PCM system

PCM Transmitter



PCM Receiver



PCM Advantages and Disadvantages

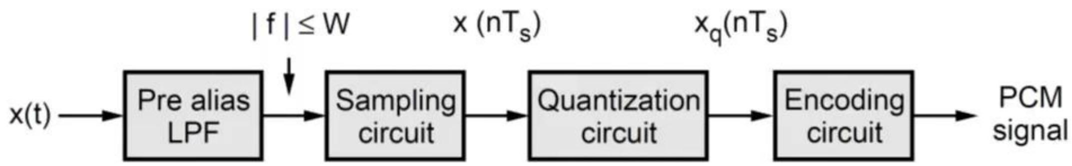
- **Advantages of PCM:**

1- Low channel noise and interference.
2- Easy multiplexing of various PCM signals.

- **Disadvantages of PCM:**

1- Complex systems.
2- Wide bandwidth is required.

- $x(t)$ is first passed through pre-alias low pass filter of cut off frequency 'W' Hz. Hence all the frequency components higher than 'W' Hz are blocked.



Block diagram of PCM generator

- The signal is then sampled by the sampling circuit. The sampling frequency $f_s \geq 2W$. And $T_s = \frac{1}{f_s}$.
- The quantization circuit quantizes the sampled signal to finite quantization levels. The quantized signal takes any one of the 'q' quantization levels.
- The quantized signal is then represented by finite number of digits. For example if there are 3-bits, then it will represent $2^3 = 8$ quantization levels. For example quantization level of '7' will be encoded as 111 by 3-bits. This encoded signal is called PCM signal.

PCM receiver

- The PCM signal is first given to regenerator circuit. The regenerator reshapes the pulses and removes the noise.
- The signal is then given to digital to analog converter. It is also called as decoder. It obtains and analog signal which has stepped nature.
- The lowpass filter finally smooths out the signal obtained from D/A converter.

Transmission Bandwidth in PCM

Let the quantizer use 'v' number of binary digits to represent each level. Then the number of levels that can be represented by 'v' digits will be,

$$q = 2^{v}$$

Here 'q' represents total number of digital levels of q-level quantizer. Each sample is converted to 'v' binary bits. i.e. Number of bits per sample = v

We know that, Number of samples per second = f_s

$\therefore$ Number of bits per second is given by,

$$\begin{aligned}(\text{Number of bits per second}) &= (\text{Number of bits per samples}) \\ &\quad \times (\text{Number of samples per second}) \\ &= v \text{ bits per sample} \times f_s \text{ samples per second}\end{aligned}$$

The number of bits per second is also called signaling rate of PCM and is denoted by 'r' i.e.,

$$\boxed{\textbf{Signaling rate in PCM : } r = v\, f_s}$$

Here $f_s \geq 2W$.

Bandwidth needed for PCM transmission will be given by half of the signaling rate i.e.,

Transmission bandwidth of PCM
$$\begin{cases} B_T \geq \frac{1}{2} r \\ B_T \geq \frac{1}{2} v\, f_s \quad \text{Since } f_s \geq 2\,W \\ B_T \geq v\,W \end{cases}$$

PCM Equations

1

$$R_b = b f_s \quad (b/s)$$

R_b : Bit rate (b/s)

b : Number of bits.

f_s : sampling frequency.

2

Transmission bandwidth B_{PCM} of PCM

$$B_{PCM} \geq \frac{1}{2} R_b \geq \frac{1}{2} b f_s \geq \frac{1}{2} b f_m$$

3

Information Capacity of PCM

$$C = B * Log_2(1 + \frac{S}{N})$$

$$C = B * Log_2(1 + \frac{12 P_x}{\Delta^2})$$

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Q (1):- A compact disc (CD) records audio signals digitally by using **PCM**. Assume the audio signal **15KHz**.

1- What is the ***Nyquist rate***.

2- If the samples are quantized into $L = 65536$ levels and then binary coded, determine the number of binary digits required to encoded a sample.

3- Determine the number of binary digits per second (***bit rate***).

Solution:-

1- $f_N = f_s = 2f_m$

$$f_N = 2 * 15\ K = 30\ KHz$$

2- $L = 2^b$

$$b = \frac{Log\ L}{Log\ 2} = \frac{Log\ (65536)}{Log\ 2} = 16\ bit$$

3- $R_b = bf_s$

$$R_b = 16 * 30\ K = 480\ K\ (b/s)$$

Q (2):- A television has band limited **4.5 MHz**. This signal is sampled, quantized and binary coded to obtain a **PCM** signal.

1- Determine the sample rate if the signal is to be sampled at a rate ***20%*** above the Nyquist rate .

2- If the samples are quantized into ***L = 1024*** levels and then binary coded, determine the number of binary digits required to encoded a sample.

3- Determine the number of binary digits per second (***bit rate***) and the minimum bandwidth required to transmit this signal.

Solution:-

1- $f_N = 2f_m = 2 * 4.5\ M = 9\ MHz$

$\therefore f_s > 20\%\ f_N$

$f_s > 20\% * 9\ MHz = 1.8\ MHz$

$f_s = 9\ M + 1.8\ M = 10.8\ MHz$

2- $L = 2^b$

$b = \frac{Log\ L}{Log\ 2} = \frac{Log\ (1024)}{Log\ 2} = 10\ bit$

3- $R_b = b f_s$

$R_b = 10 * 10.8\ M = 108\ M\ (b/s)$

$B_{PCM} \geq \frac{1}{2} R_b$

$B_{PCM} \geq \frac{1}{2} 108\ M$

$B_{PCM} \geq 54\ MHz$

So far we have gone through different modulation techniques. The one remaining is **digital modulation**, which falls under the classification of pulse modulation. Digital modulation has Pulse Code Modulation (PCM) as the main classification. It further gets processed to delta modulation and ADM

PCM (Pulse Code Modulation)

Your microphone is muted.

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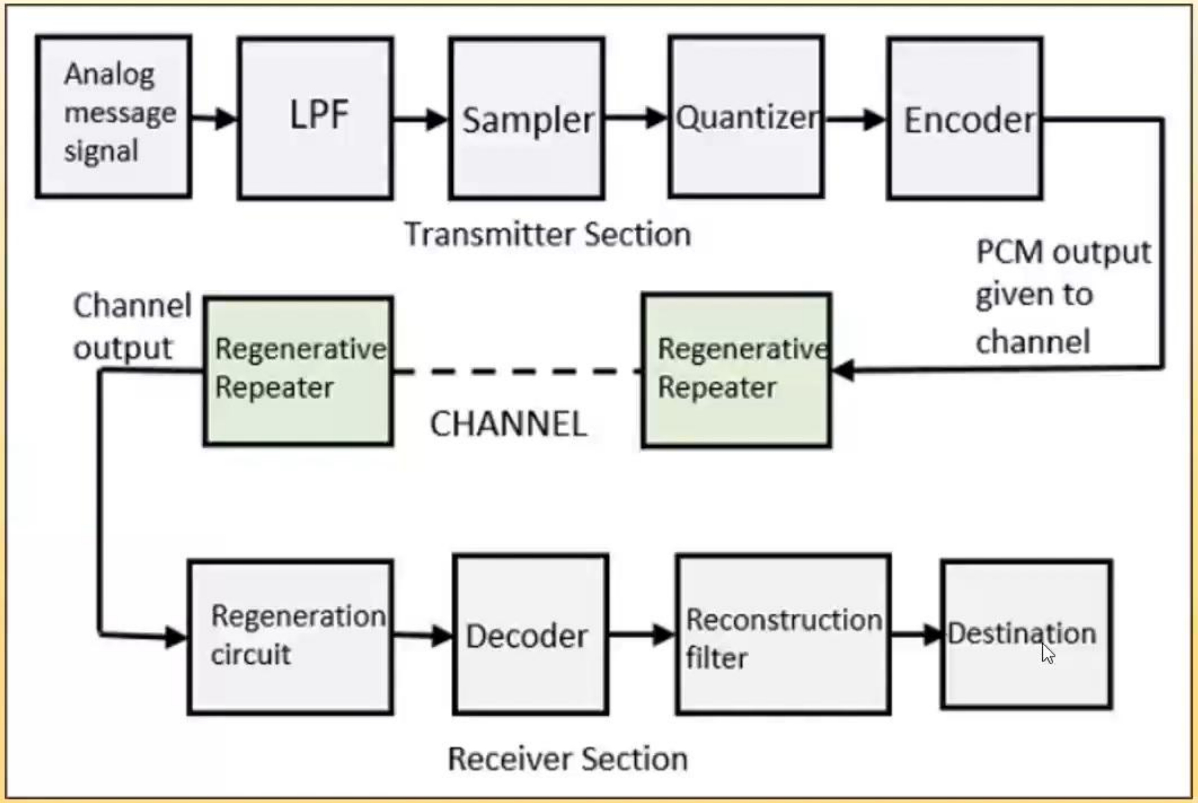
Basic Elements of PCM

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. The following figure is the block diagram of PCM which represents the basic elements of both the transmitter and the receiver sections.

PCM Transmitter & Receiver



Low Pass Filter (LPF)

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 circuit which uses the technique that helps to collect the sample data at instantaneous values of the 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 will act as an analog to the digital converter. Encoding minimizes the bandwidth used.

Regenerative Repeater

The output of the channel has one regenerative repeater circuit to compensate the signal loss and reconstruct the signal. It also increases the strength of the signal.

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 analog signal given, codes it, and samples it. It then transmits in an analog form. This whole process is repeated in a reverse pattern to obtain the original signal.

Quantization

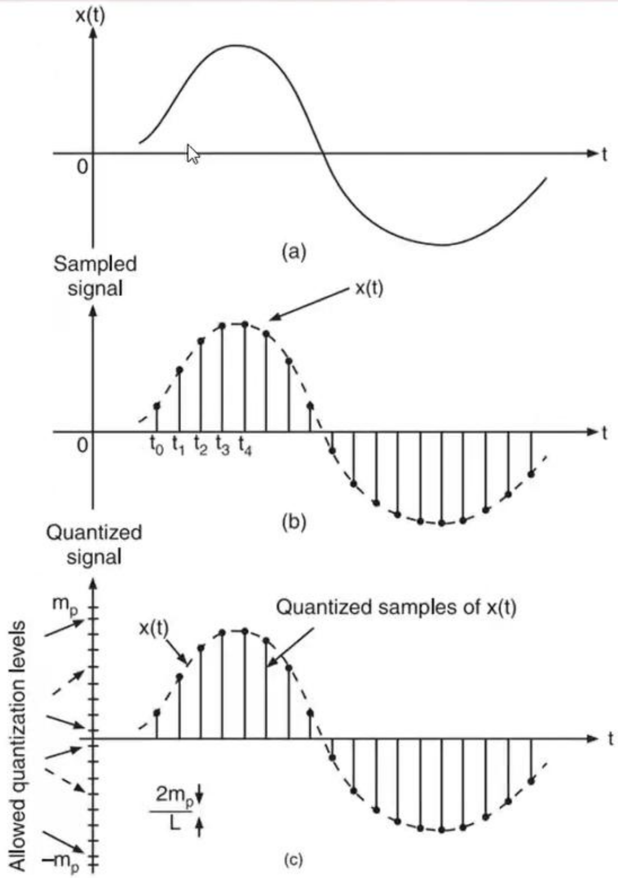
There are few modulation techniques which are followed to construct a PCM signal. These techniques like **sampling, quantization,** and **companding** help to create an effective PCM signal, which can exactly reproduce the original signal.

Quantization

The digitization of analog signals involves the rounding off of the values which are approximately equal to the analog values. The method of sampling chooses 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**.

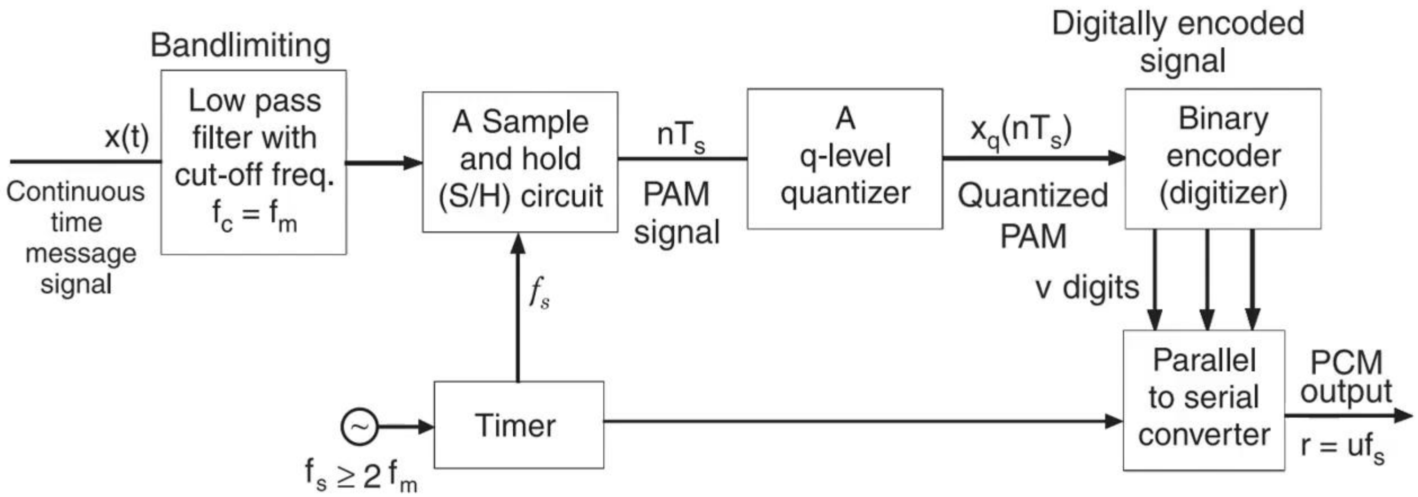
The quantizing of an analog signal is done by discretizing the signal with a number of quantization levels. Quantization is representing the sampled values of the amplitude by a finite set of levels, which means converting a **continuous-amplitude sample** into a **discrete-time signal**.

Sampling & Quantization



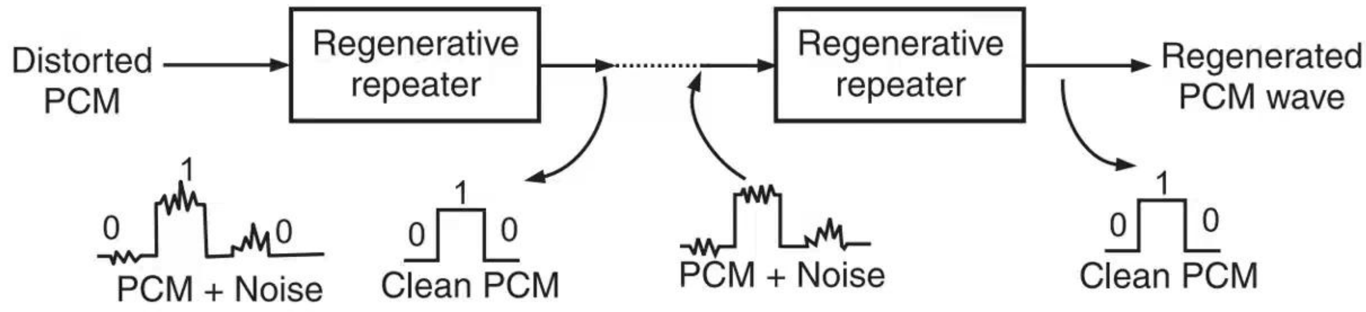
(a) An analog signal, (b) Samples of analog signal, (c) Quantization

PCM Generator



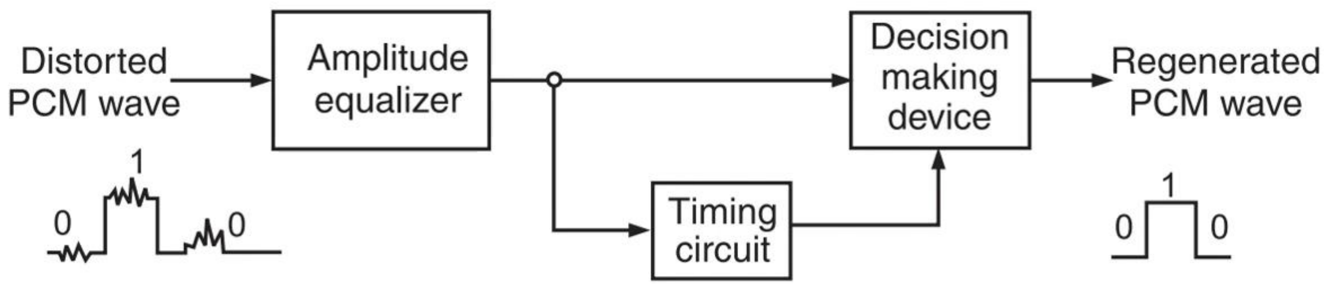
A practical PCM generator

PCM Transmission Path



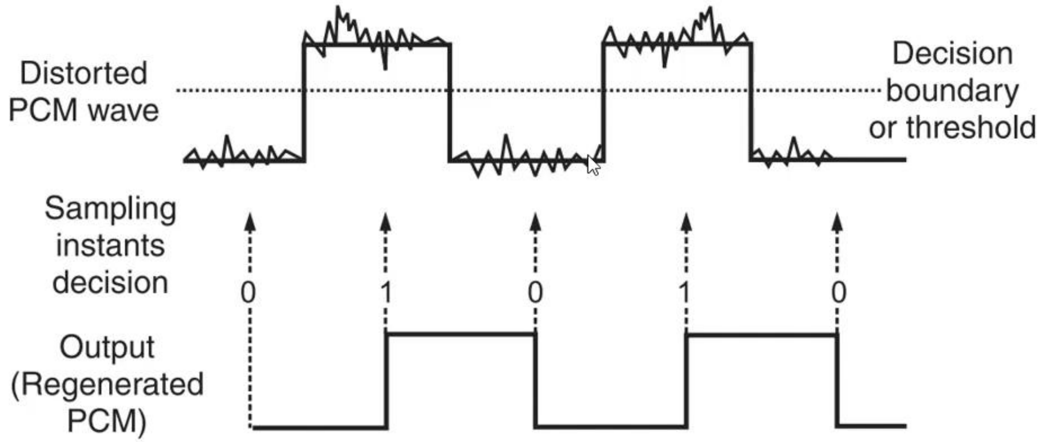
PCM transmission path

PCM – Regenerative Repeater



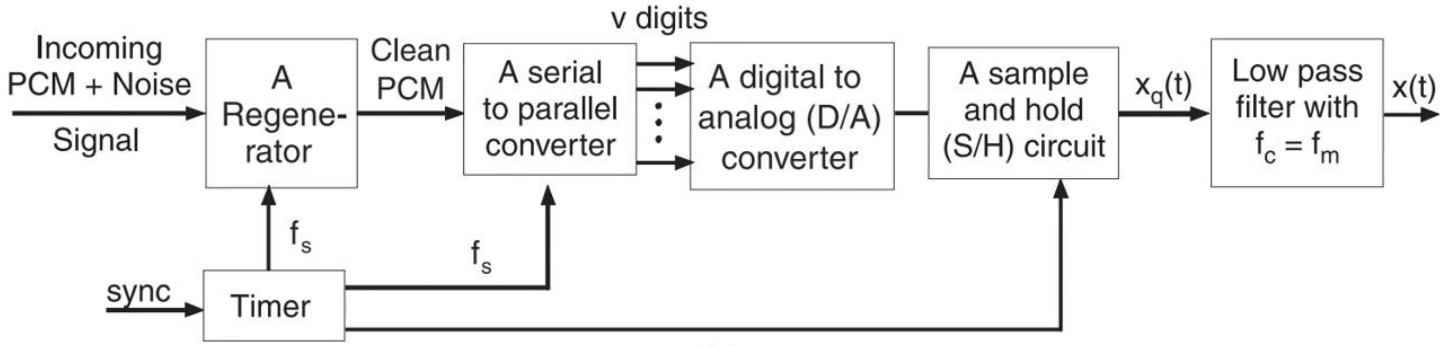
Block diagram of a regenerative repeater

PCM Regenerative Repeater Waveform



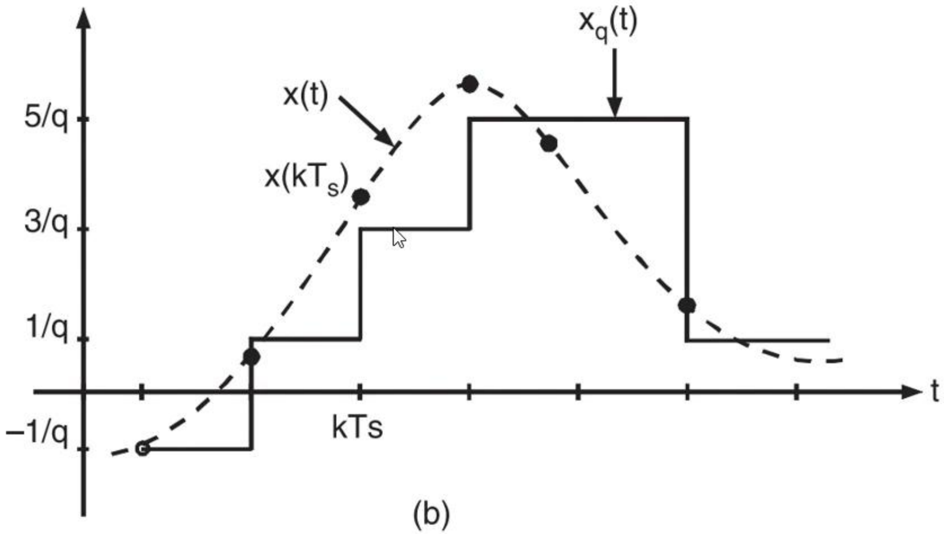
Waveforms of a regenerative repeater

PCM Receiver



(a)

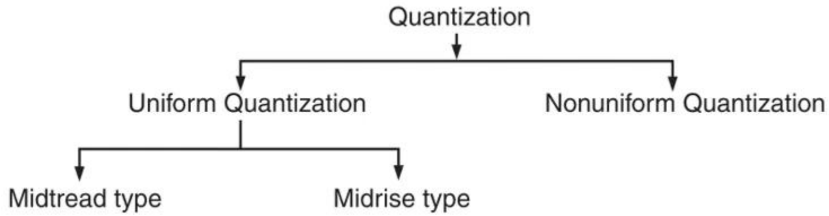
PCM Reconstructed Waveform



(a) PCM receiver, (b) Reconstructed waveform

Classification of Quantization Process

Basically, quantization process may be classified as follows :



The quantization process can be classified into two types as under:

(i) Uniform quantization

(ii) Non-uniform quantization.

This classification is based on the step size as defined earlier.

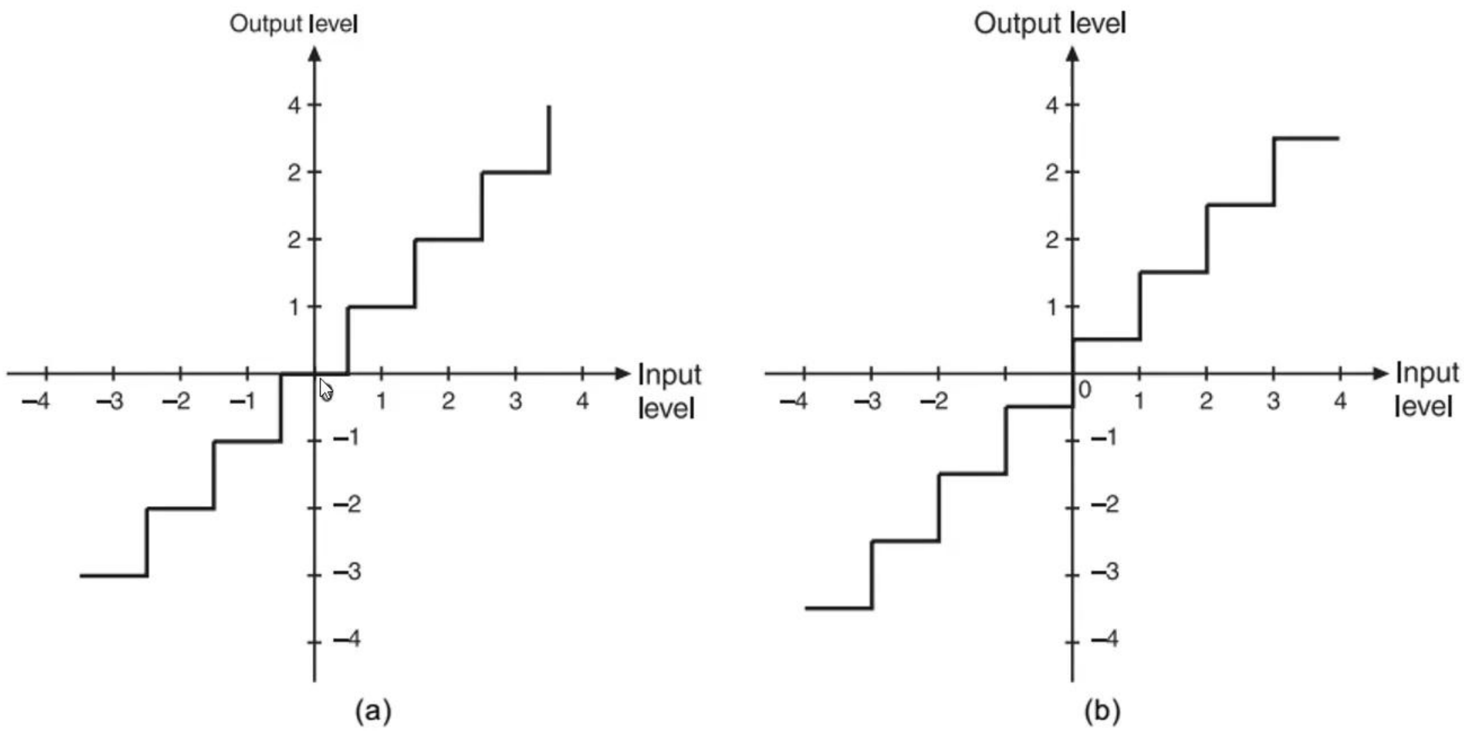
(i) Uniform Quantizer

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

(ii) Non-uniform Quantizer

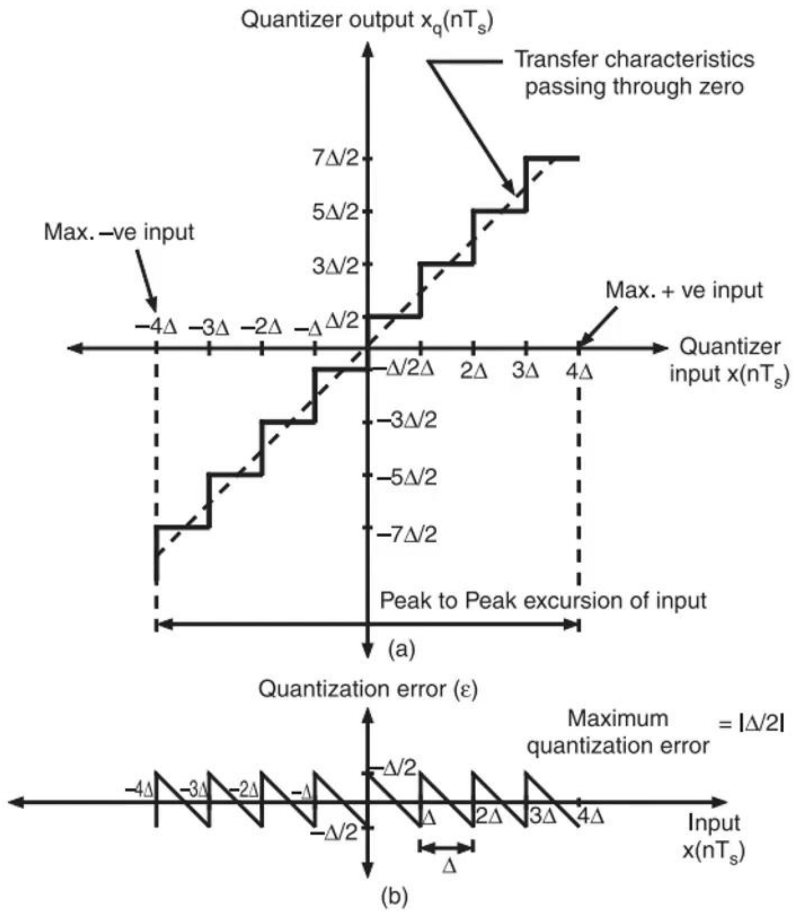
A non-uniform quantizer is that type of quantizer in which the 'step-size' varies according to the input signal values.

Uniform Quantization



Two types of Uniform quantization: (a) Midtread, and (b) Midrise

Transfer characteristic of Quantizer



Transfer characteristic of a quantizer (b) Variation of quantization

Companding in PCM

The word **Companding** is a combination of **Com**pressing and Ex**panding**, which means that it does both. This is a non-linear technique used in PCM which compresses the data at the transmitter and expands the same data at the receiver. The effects of noise and crosstalk are reduced by using this technique.

There are two types of Companding techniques.

Your microphone is muted.

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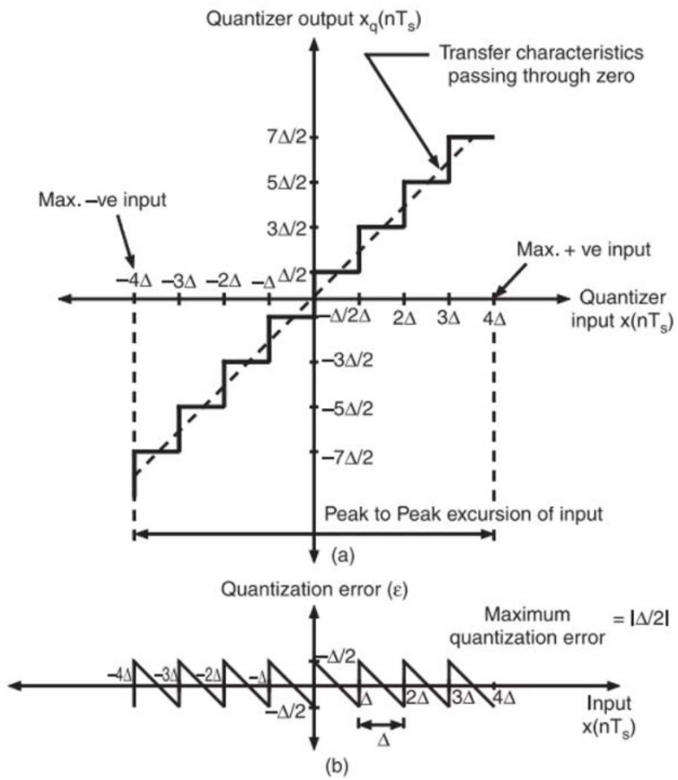
Transfer characteristic of Quantizer



Transfer characteristic of a quantizer (b) Variation of quantization

Companding in PCM & Types

Companding in PCM

The word **Companding** is a combination of **Com**pressing and Ex**panding**, which means that it does both. This is a non-linear technique used in PCM which compresses the data at the transmitter and expands the same data at the receiver. The effects of noise and crosstalk are reduced by using this technique.

There are two types of Companding techniques.

A-law Companding Technique

- Uniform quantization is achieved at **A = 1**, where the characteristic curve is linear and there is no compression.
- A-law has mid-rise at the origin. Hence, it contains a non-zero value.
- A-law companding is used for PCM telephone systems.
- A-law is used in many parts of the world.

μ-law Companding Technique

- Uniform quantization is achieved at **μ = 0**, where the characteristic curve is linear and there is no compression.
- μ-law has mid-tread at the origin. Hence, it contains a zero value.
- μ-law companding is used for speech and music signals.
- μ-law is used in North America and Japan.

Comparison

The sampling rate of a signal should be higher than the Nyquist rate, to achieve better sampling. If this sampling interval in a Differential PCM (DPCM) is reduced considerably, the sample-to-sample amplitude difference is very small, as if the difference is **1-bit quantization**, then the step-size is very small i.e., **Δ** (delta).

Differential PCM

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 predicted sampled values, assumed from its previous outputs and summarize them with the quantized values.

Such a process is named as **Differential PCM** technique.

APPLICATIONS OF PCM

Some of the applications of PCM may be listed as under:

(i) With the advent of fibre optic cables, PCM is used in telephony.

(ii) In space communication, space craft transmits signals to earth. Here, the transmitted power is quite small (i.e., 10 or 15 W) and the distances are very large (i.e., a few million km). However, due to the high noise immunity, only PCM systems can be used in such applications.

ADVANTAGES OF PCM : SALIENT FEATURES OF PCM

Following are the advantages of a PCM system:

(i) PCM provides high noise immunity.

(ii) Due to digital nature of the signal, we can place repeaters between the transmitter and the receivers. Infact, the repeaters regenerate the received PCM signal. This can not be possible in analog systems. Repeaters further reduce the effect of noise.

(iii) We can store the PCM signal due to its digital nature.

(iv) We can use various coding techniques so that only the desired person can decode the received signal.

Delta Modulation

What is Delta Modulation?

The type of modulation, where the sampling rate is much higher and in which the stepsize after quantization is of smaller value **Δ**, such a modulation is termed as **delta modulation**.

Features of Delta Modulation

- An over-sampled input is taken to make full use of a signal correlation.
- The quantization design is simple.
- The input sequence is much higher than Nyquist rate.
- The quality is moderate.
- The design of the modulator and the demodulator is simple.
- The stair-case approximation of output waveform.
- The step-size is very small, i.e., **Δ** (delta).
- The bit rate can be decided by the user.
- It requires simpler implementation.

Delta Modulation is a simplified form of DPCM technique, also viewed as 1-bit DPCM scheme. As the sampling interval is reduced, the signal correlation will be higher.

(i) Reason to use Delta Modulation

We have observed 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.

(ii) Working Principle

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 transmitted. Input signal x(t) is approximated to step signal by the delta modulator. This step size is kept fixed. The difference between the input signal x(t) and staircase approximated signal is confined to two levels, i.e., $+\Delta$ and $-\Delta$. Now, if the difference is positive, then approximated signal is increased by one step, i.e., '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. Hence, for each sample, only one binary bit is transmitted. Figure shows the analog signal x(t) and its staircase approximated signal by the delta modulator.

(iii) Mathematical Expressions

Thus, the principle of delta modulation can be explained with the help of few equations as under:

The error between the sampled value of x(t) and last approximated sample is given as,

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

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

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

$\hat{x}(nT_s)$ = last sample approximation of the staircase waveform.

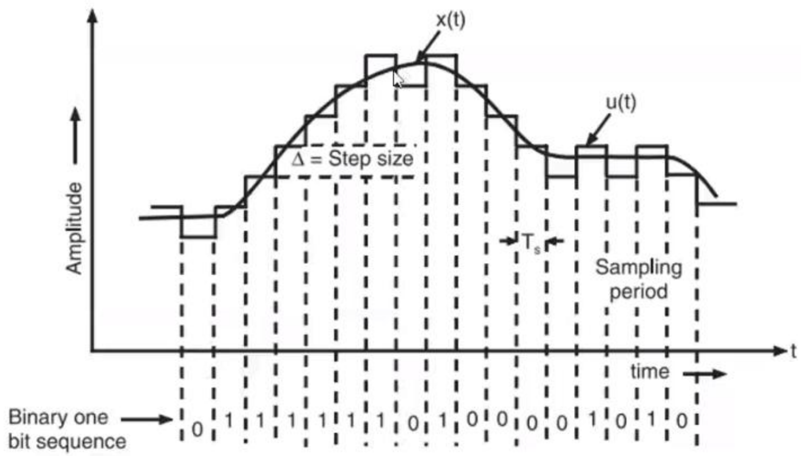
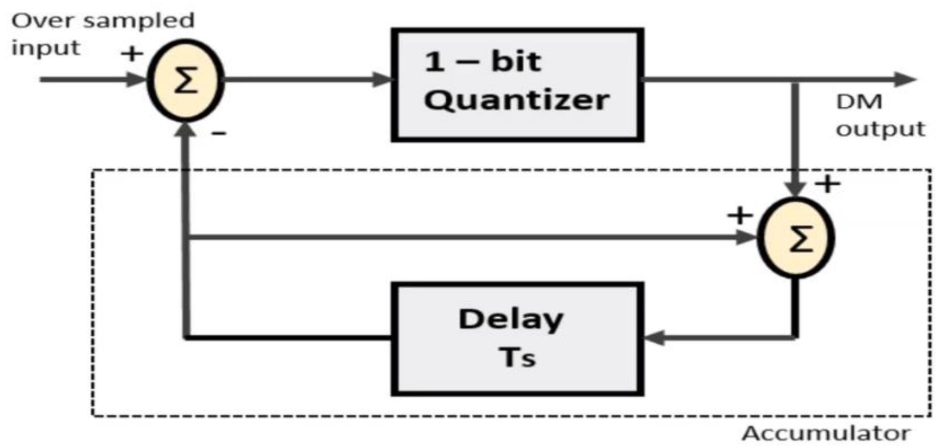


Fig. Delta modulation waveform

DM – Block Diagram

The **Delta Modulator** comprises of a 1-bit quantizer and a delay circuit along with two summer circuits. Following is the block diagram of a delta modulator.



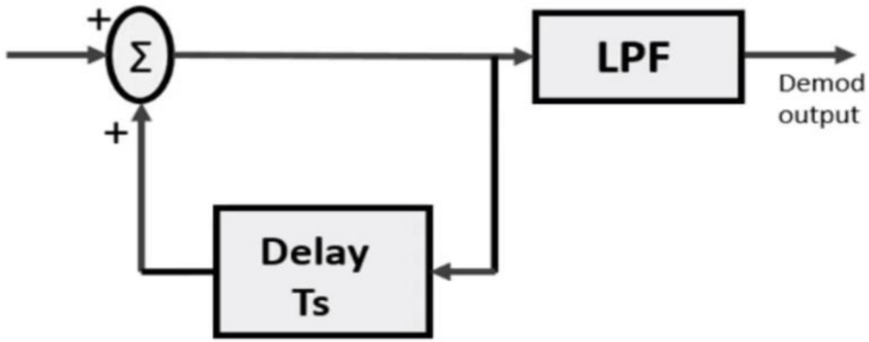
A stair-case approximated waveform will be the output of the delta modulator with the step-size as delta (**Δ**). The output quality of the waveform is moderate.

Delta Demodulator

Delta Demodulator

The delta demodulator comprises of a low pass filter, a summer, and a delay circuit. The predictor circuit is eliminated here and hence no assumed input is given to the demodulator.

Following is the block diagram for delta demodulator.



Low pass filter is used for many reasons, but the prominent one is noise elimination for out-of-band signals. The step-size error that may occur at the transmitter is called **granular noise**, which is eliminated here. If there is no noise present, then the modulator output equals the demodulator input.

Advantages of DM over DPCM

- 1-bit quantizer
- Very easy design of modulator & demodulator

However, there exists some **noise in DM** and following are the types of noise.

- Slope Over load distortion (when Δ is small)
- Granular noise (when Δ is large)