Program : **B.Tech**

Subject Name: **Analog and Digital Communication**

Subject Code: **IT-404**

Semester: **4th**

Analog & Digital communication
IT-404

UNIT 4

Syllabus: Sampling of signal, sampling theorem for low pass and Band pass signal, Pulse amplitude modulation (PAM), and Time division multiplexing (TDM). Channel Bandwidth for PAM-TDM signal Type of sampling instantaneous, Natural and flat top, Aperture effect, Introduction to pulse position and pulse duration modulations, Digital signal, Quantization, Quantization error, Pulse code modulation, signal to noise ratio, Companding, Data rate and Baud rate, Bit rate, multiplexed PCM signal, Differential PCM (DPCM), Delta Modulation (DM) and Adaptive Delta Modulation (ADM), comparison of various systems.

4.1. Sampling of signal: The signals we use in the real world, such as our voices, are called "analog" signals. To process these signals in computers, we need to convert the signals to "digital" form. While an analog signal is continuous in both time and amplitude, a digital signal is discrete in both time and amplitude. To convert a signal from continuous time to discrete time, a process called sampling is used. The value of the signal is measured at certain intervals in time. Each measurement is referred to as a sample.

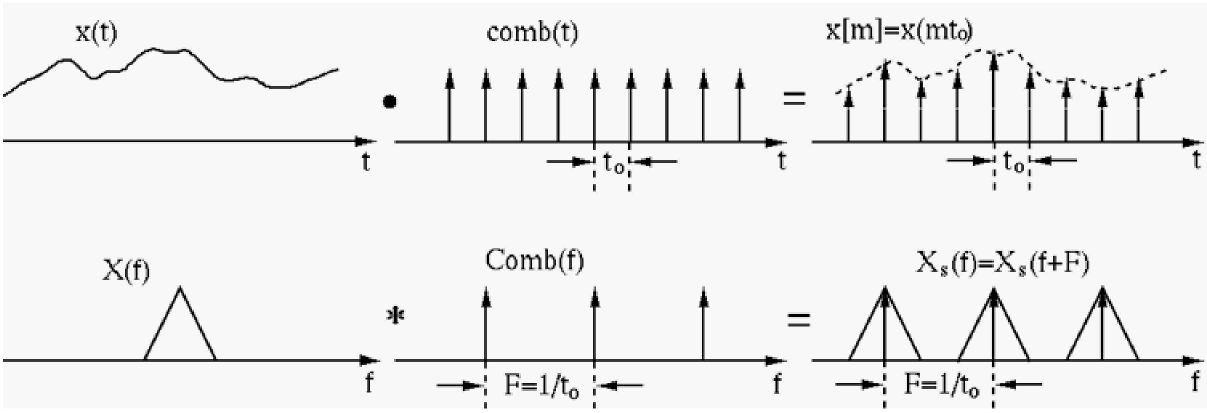


Figure.4.1. Sampling of signal and its frequency spectrum

A discrete-time signal can be obtained by uniformly sampling a continuous-time signal at $t_n = nT_S$, i.e.,$x[n] = x[nT_S]$. The values x[n] are samples of x(t), the time interval between samples is T_s, the sampling rate is $f_s = 1/T_S$. A system which performs the sampling operation is called a continuous-to-discrete (C-to-D) converter.

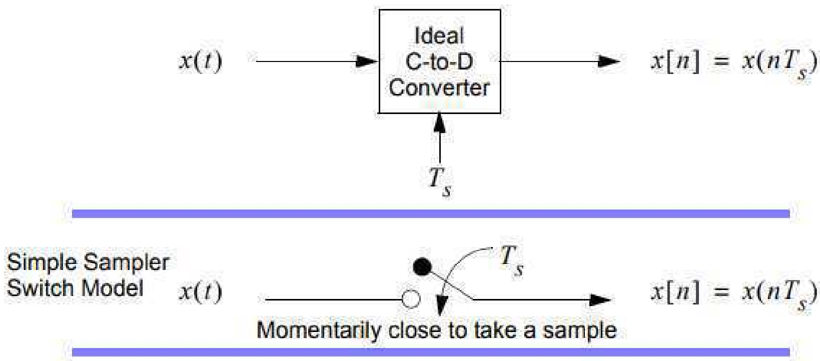


Figure.4.2.Sampler as switch

4.2. Sampling theorem for low pass signal

The lowpass sampling theorem states that, "a signal whose spectrum is band limited to B Hz, can be reproduced exactly from its samples if it is sampled at the rate fs which is greater than twice the maximum frequency ω of the signal to be sampled." Therefore minimum sampling frequency is

$$f_s = 2B\ Hz$$

If we have a continuous time signal x(t). The spectrum of x(t) is a band limited to f_m Hz i.e. the spectrum of x(t) is zero for $|\omega|>\omega_m$.

Sampling theorem statement 2: A continuous-time signal x(t) with frequencies no higher than f_m (Hz) can be reconstructed EXACTLY from its samples x[n] = x(nTs), if the samples are taken at a rate fs = 1/Ts that is greater than $2f_m$.

Sampling of input signal x(t) can be obtained by multiplying x(t) with an impulse train δ(t) of period T_s. The output of multiplier is a discrete signal called sampled signal which is represented with y(t) in the following diagrams:

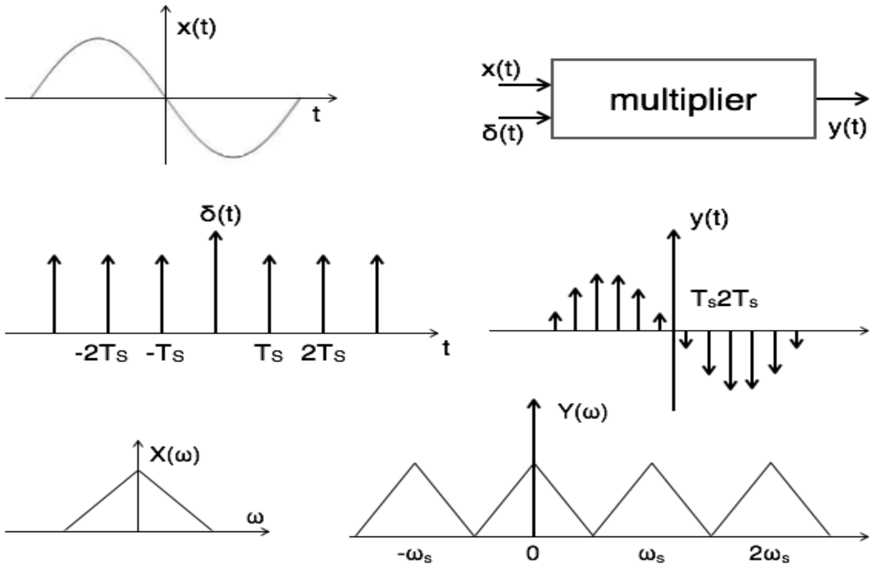


Figure.4.3. Sampling of signal x(t)

Here, you can observe that the sampled signal takes the period of impulse. The process of sampling can be understood as under.

The sampled signal is given by

$$y(t) = x(t).\delta(t) = x(t).\delta T_s(t)$$
$$= \sum_n x(nT_s).\delta(t-nT_s)$$

The impulse train $\delta T_s(t)$ is a periodic signal of period T_s, hence it can be expressed as a Fourier series

$$\delta T_s(t) = \frac{1}{T_s}[1 + 2\,cos\omega_s t + \ 2\,cos2\omega_s t + \ 2\,cos3\omega_s t + \cdots]$$

$$\text{Where } \omega_s = \frac{2\pi}{T_s} = 2\pi f_s$$

Substitute $\delta(t)$in equation 1.

$$\rightarrow y(t) = x(t).\delta(t)$$

$$y(t) = \frac{1}{T_s}[x(t) + 2x(t)\,cos\omega_s t + \ 2x(t)\,cos2\omega_s t + \ 2\,x(t)cos3\omega_s t + \cdots]$$

Now to find the $Y(\omega)$, we have to take the fourier transform of both the sides,

$$Y(\omega) = \frac{1}{T_s}[X(\omega) + X(\omega - \omega_s) + X(\omega + \omega_s) + X(\omega - 2\omega_s) + X(\omega + 2\omega_s) + \cdots]$$

$$Y(\omega) = \frac{1}{T_s}\sum_{n=-\infty}^{\infty} X(\omega - n\omega_s) \text{ Where } n = 0, \pm1, \pm2, \ldots$$

The Fourier spectrum $Y(\omega)$ is shown in figure. Now If we have to recover $x(t)$ from $y(t)$, we should be able to recover $X(\omega)$ from $Y(\omega)$, and it is possible only if there is no overlapping between the successive cycles of $Y(\omega)$, and for this condition

$$f_s > 2B\ Hz$$

Therefore the sampling interval

$$T_s < \frac{1}{2B}$$

Therefore as long as the sampling frequency f_s is greater than $2B, Y(\omega)$, will consist of non overlapping repetitions of $X(\omega)$, and $x(t)$ can be recovered from $y(t)$ by passing $y(t)$ by an ideal low pass filter with cut off frequency B Hz.

4.3. Sampling theorem for Band pass signal

In case of band pass signals, the spectrum of band pass signal X[ω] = 0 for the frequencies outside the range $f_1 \le f \le f_2$. The frequency f_1 is always greater than zero. Plus, there is no aliasing effect when $f_s > 2f_2$. But it has two disadvantages:

- The sampling rate is large in proportion with f_2. This has practical limitations.
- The sampled signal spectrum has spectral gaps.

To overcome this, the band pass theorem states that the input signal x(t) can be converted into its samples and can be recovered back without distortion when sampling frequency $f_s < 2f_2$.

4.4. Nyquist criteria and Signal reconstruction of sampled signal:

According to the Nyquist criterion, in order to recover the original signal from its samples, the minimum sampling frequency rate must be twice of the bandwidth of the message signal.

This minimum sampling rate f_s=2B required to recover x(t) from its samples y(t) is called the Nyquist rate and the corresponding sampling interval is called Nyquist Interval for y(t).

Effect of Sampling Rate on Y(ω)

Possibility of sampled frequency spectrum with different conditions is given by the following diagrams:

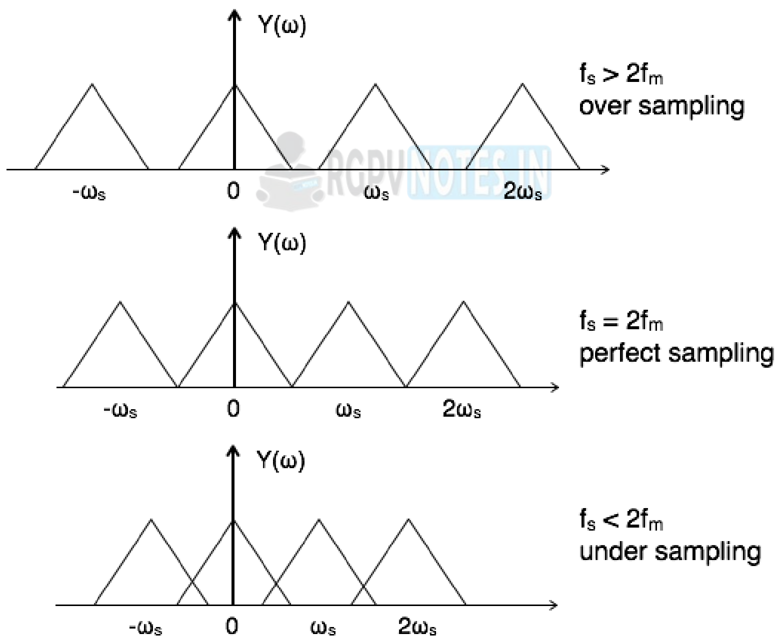


Figure.4.4. Sampling Conditions

Therefore, if we are sampling below the Nyquist rate then the recovery of the signal will not be possible.

4.5. Aliasing – Aliasing refers to the phenomenon of a high frequency component in the spectrum of a signal seemingly taking on the identity of a lower frequency in the spectrum of its sampled version (under sampled version of the message signal) Corrective measures for aliasing effects

1. Prior to sampling, a low-pass anti-aliasing filter is used to attenuate those high-frequency components of the signal that are not essential to the information being conveyed by the signal.

2. The filtered signal is sampled at a rate higher than the Nyquist rate.

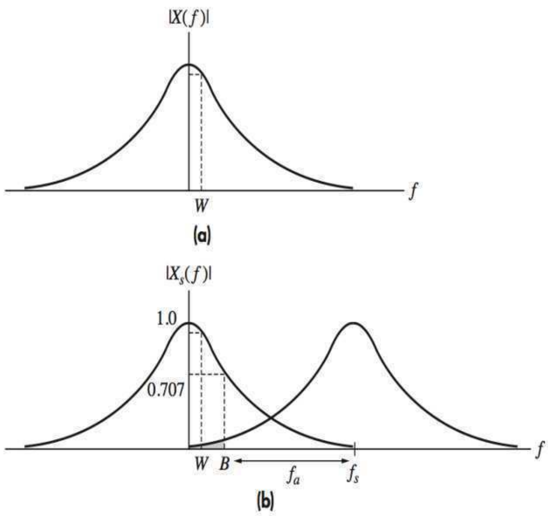
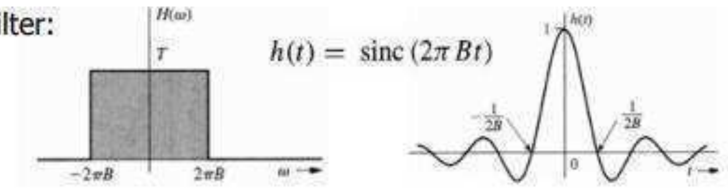


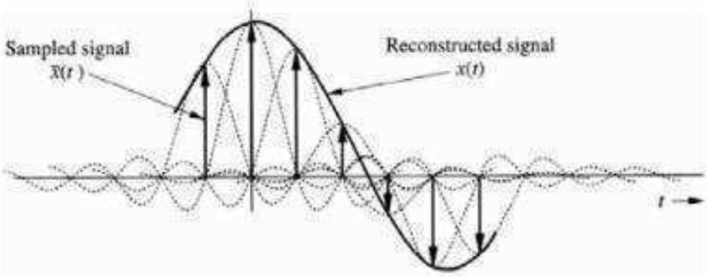
Figure.4.5. Aliasing

Ideal signal reconstruction:

- Use ideal lowpass filter:



- That's why the sinc function is also known as the **interpolation** function:



$$x(t) = \sum_n x(nT)h(t - nT)$$

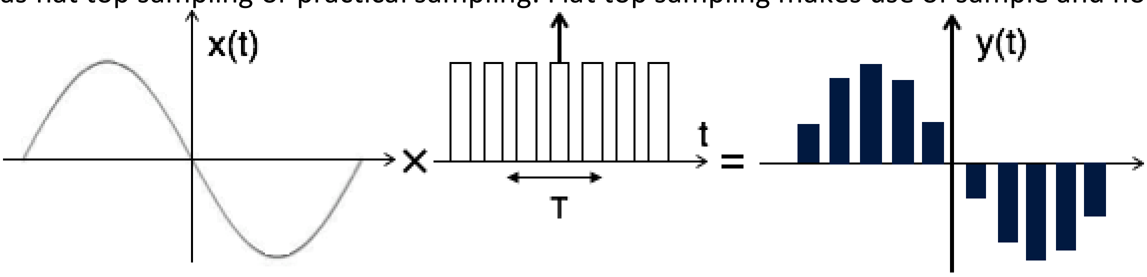
$$= \sum_n x(nT)\,\mathrm{sinc}\,(2\pi Bt - n\pi)$$

Figure.4.6. Ideal reconstruction system

4.5. Types of sampling:

4.5.1. Flat Top Sampling

During transmission, noise is introduced at top of the transmission pulse which can be easily removed if the pulse is in the form of flat top. Here, the top of the samples are flat i.e. they have constant amplitude. Hence, it is called as flat top sampling or practical sampling. Flat top sampling makes use of sample and hold circuit.



Figur.4.7. Flat Top Sampling

Theoretically, the sampled signal can be obtained by convolution of rectangular pulse p(t) with ideally sampled signal say $y_\delta(t)$ as shown in the diagram:

4.5.2. Natural Sampling

Natural sampling is similar to impulse sampling, except the impulse train is replaced by pulse train of period T. i.e. you multiply input signal x(t) to pulse train $\sum_{n=-\infty}^{\infty} P(t-nT)$ as shown below

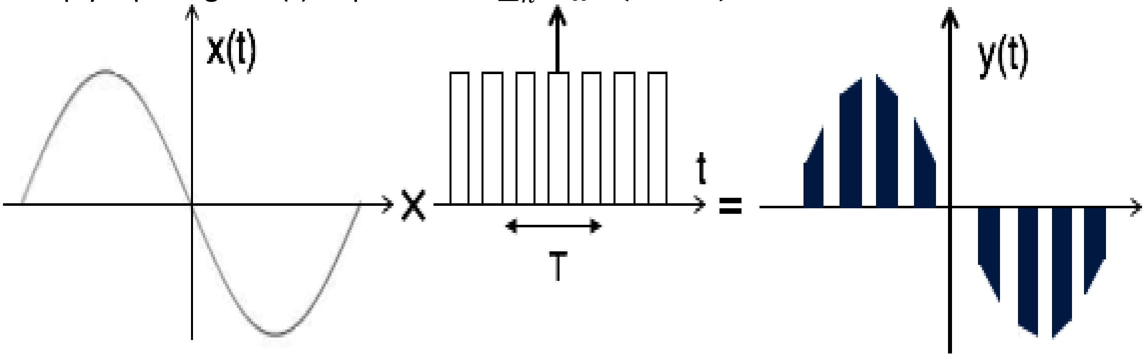


Figure.4.8.Natural Sampling

4.5.3. Impulse Sampling

Impulse sampling can be performed by multiplying input signal x(t) with impulse train Σδ(t−nT) of period 'T'. Here, the amplitude of impulse changes with respect to amplitude of input signal x(t). The output of sampler is given by

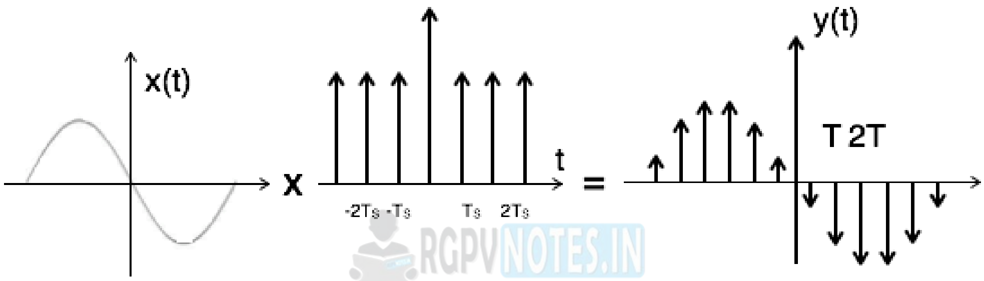


Figure.4.9.Impulse Sampling

y(t)=x(t)×y(t)=x(t)× impulse train

=x(t)×$\sum_{n=-\infty}^{\infty} \delta(t-nT)$

y(t)=yδ(t)= $\sum_{n=-\infty}^{\infty} \mathrm{x(nt)}\, \delta(t-nT)$

To get the spectrum of sampled signal, consider Fourier transform of equation on both sides

Y(ω)=1$\frac{1}{T}\sum_{n=-\infty}^{\infty} \mathrm{X}(\omega-\mathrm{n}\omega_s)$

This is called ideal sampling or impulse sampling. You cannot use this practically because pulse width cannot be zero and the generation of impulse train is not possible practically.

4.6. Time Division Multiplexing:

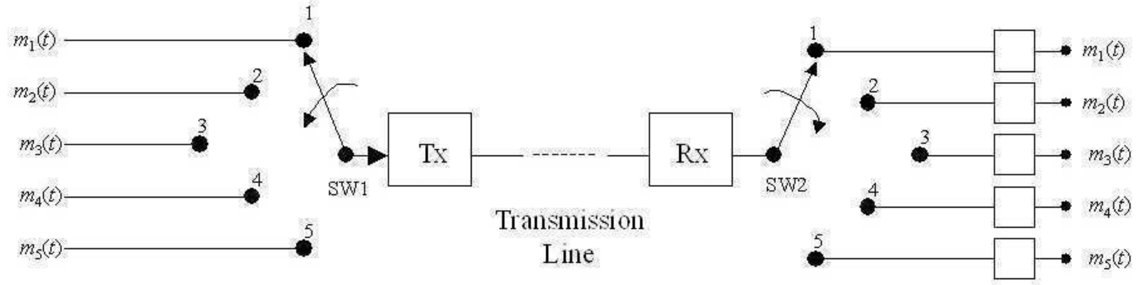
In TDM, the data flow of each input stream is divided into units. One unit may be 1 bit, 1 byte, or a block of few bytes. Each input unit is allotted an input time slot. One input unit corresponds to one output unit and is allotted an output time slot. During transmission, one unit of each of the input streams is allotted one-time slot, periodically, in a sequence, on a rotational basis. TDM allows each channel the full band width of the transmission medium whenever its signal is transmitted, although each channel is not continuously on the system.

Advantages of TDM

– high reliability and efficient operation as the circuitry required is digital.

– Relatively small inter channel cross-talk arising from nonlinearities in the amplifiers that handle the signals in the transmitter and receiver.

Disadvantages of TDM – timing jitter



Switches SW1 and SW2 rotate in synchronism, and in effect sample each message input in a sequence $m_1(t)$, $m_2(t)$, $m_3(t)$, $m_4(t)$, $m_5(t)$, $m_1(t)$, $m_2(t)$,…

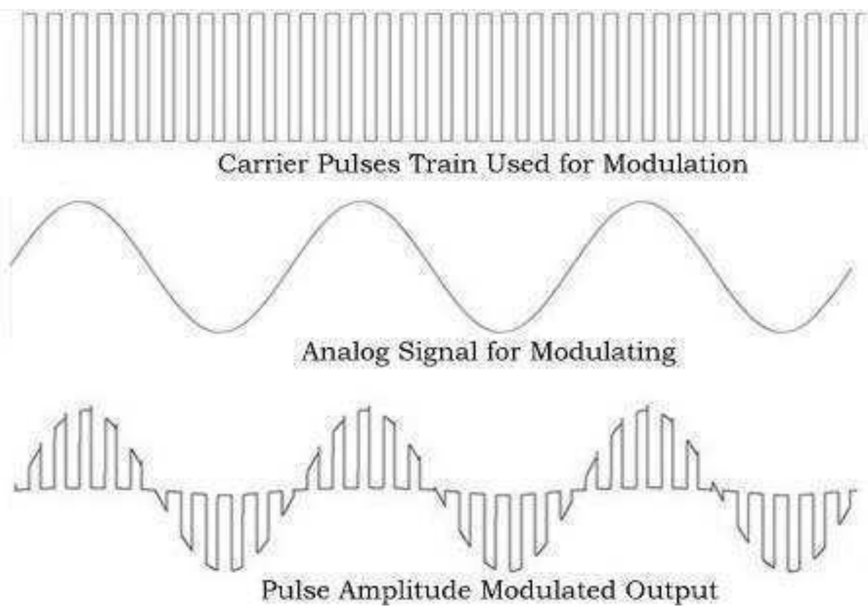
Figure.4.10. TDM system

4.7. Pulse Amplitude Modulation:

In pulse amplitude modulation, the amplitude of regular interval of periodic pulses or electromagnetic pulses is varied in proposition to the sample of modulating signal or message signal. This is an analog type of modulation. In the pulse amplitude modulation, the message signal is sampled at regular periodic or time intervals and this each sample is made proportional to the magnitude of the message signal. These sample pulses can be transmitted directly using wired media or we can use a carrier signal for transmitting through wireless. There are two types of sampling techniques for transmitting messages using pulse amplitude modulation, they are

- **FLAT TOP PAM:** The amplitude of each pulse is directly proportional to instantaneous modulating signal amplitude at the time of pulse occurrence and then keeps the amplitude of the pulse for the rest of the half cycle.
- **Natural PAM:** The amplitude of each pulse is directly proportional to the instantaneous modulating signal amplitude at the time of pulse occurrence and then follows the amplitude of the modulating signal for the rest of the half cycle.

Flat top PAM is the best for transmission because we can easily remove the noise and we can also easily recognize the noise. When we compare the difference between the flat top PAM and natural PAM, flat top PAM principle of sampling uses sample and hold circuit. In natural principle of sampling, noise interference is minimum. But in flat top PAM noise interference maximum. Flat top PAM and natural PAM are practical and sampling rate satisfies the sampling criteria.



Figur.4.11. Pulse Amplitude Modulation

4.8. PAM TDM system:

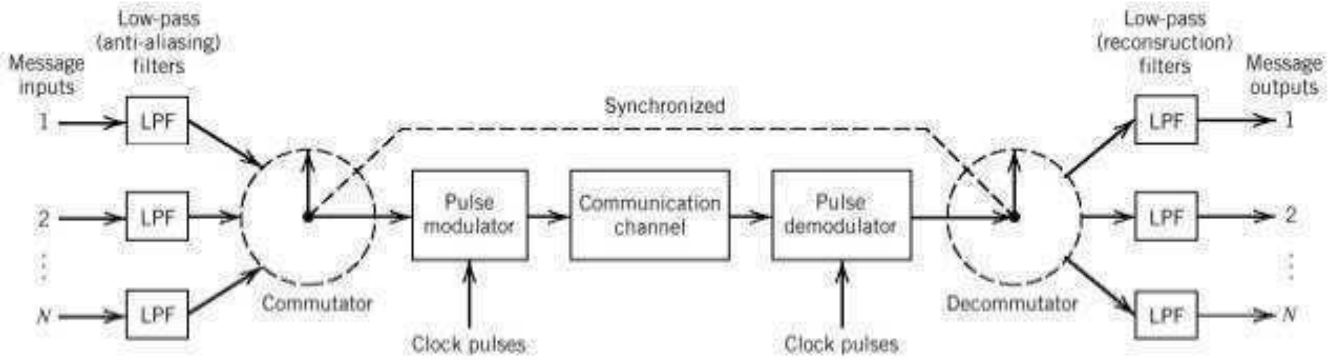


Figure.4.12. PAM TDM conceptual circuit

4.9. Pulse Duration Modulation (PDM) or Pulse Width Modulation (PWM):

It is a type of analog modulation. In pulse width modulation or pulse duration modulation, the width of the pulse carrier is varied in accordance with the sample values of message signal or modulating signal or modulating voltage. In pulse width modulation, the amplitude is made constant and width of pulse and position of pulse is made proportional to the amplitude of the signal. We can vary the pulse width in three ways:

1. By keeping the leading edge constant and vary the pulse width with respect to leading edge
2. By keeping the tailing constant.
3. By keeping the center of the pulse constant.

We can generate pulse width using different circuitry. In practical, we use 555 Timer which is the best way for generating the pulse width modulation signals. By configuring the 555 timer as mono stable or a stable multi vibrator, we can generate the PWM signals. We can use PIC, 8051, AVR, ARM, etc. microcontrollers to generate the PWM signals. PWM signal generation has n number of ways. In demodulation, we need PWM detector and its related circuitry for demodulating the PWM signal. The waveforms are shown in the figure.

4.10. Pulse Position Modulation (PPM): In the pulse position modulation, the position of each pulse in a signal by taking the reference signal is varied according to the sample value of message or modulating signal instantaneously. In the pulse position modulation, width and amplitude is kept constant. It is a technique that uses pulses of the same breath and height but is displaced in time from some base position according to the amplitude of the signal at the time of sampling. The position of the pulse is 1:1 which is propositional to the width of the pulse and also propositional to the instantaneous amplitude of sampled modulating signal. The position of pulse position modulation is easy when compared to other modulation. It requires pulse width generator and mono stable multi vibrator.

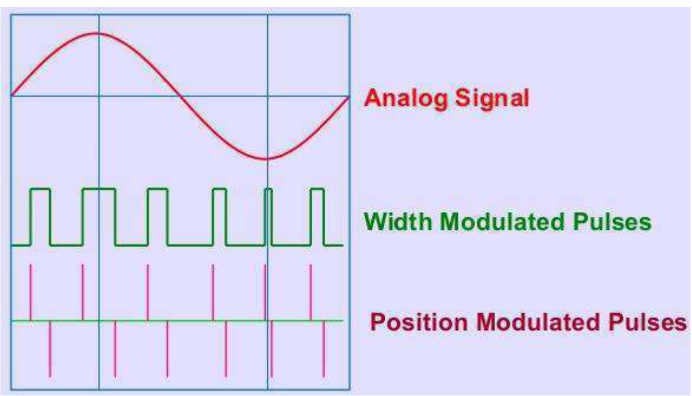


Figure.4.13. PWM and PPM

Pulse width generator is used for generating pulse width modulation signal which will help to trigger the mono stable multi vibrator; here trial edge of the PWM signal is used for triggering the mono stable multi vibrator. After triggering the mono stable multi vibrator, PWM signal is converted into pulse position modulation signal. For demodulation, it requires reference pulse generator, flip-flop and pulse width modulation demodulator.

4.11. Quantization:

In the process of quantization we create a new signal $m_q(t)$, which is an approximation to $m(t)$. The quantized signal $m_q(t)$, has the great merit that it is separable from the additive noise.

The operation of quantization is represented in figure 4.14. Here we have a signal m(t), whose amplitude varies in the range from V_H to V_L as shown in the figure.

We have divided the total range in to M equal intervals each of size S, called the step size and given by

$$S = \Delta = \frac{(V_H - V_L)}{M}$$

In our example M=8. In the centre of each of this step we located quantization levels $m0, m1, m2, \dots m7$. The $m_q(t)$ is generated in the following manner-

Whenever the signal $m(t)$ is in the range Δ_0, the signal $m_q(t)$ maintains a constant level m_0, whenever the signal $m(t)$ is in the range Δ_1, the signal $m_q(t)$ maintains a constant level m_1 and so on. Hence the signal $m_q(t)$ will found all times to one of the levels $m0, m1, m2, \dots m7$. The transition in $m_q(t)$ from $m0$ to $m1$ is made abruptly when $m(t)$ passes the transition level L_{01}, which is mid way between m_0 and m_1 and so on.

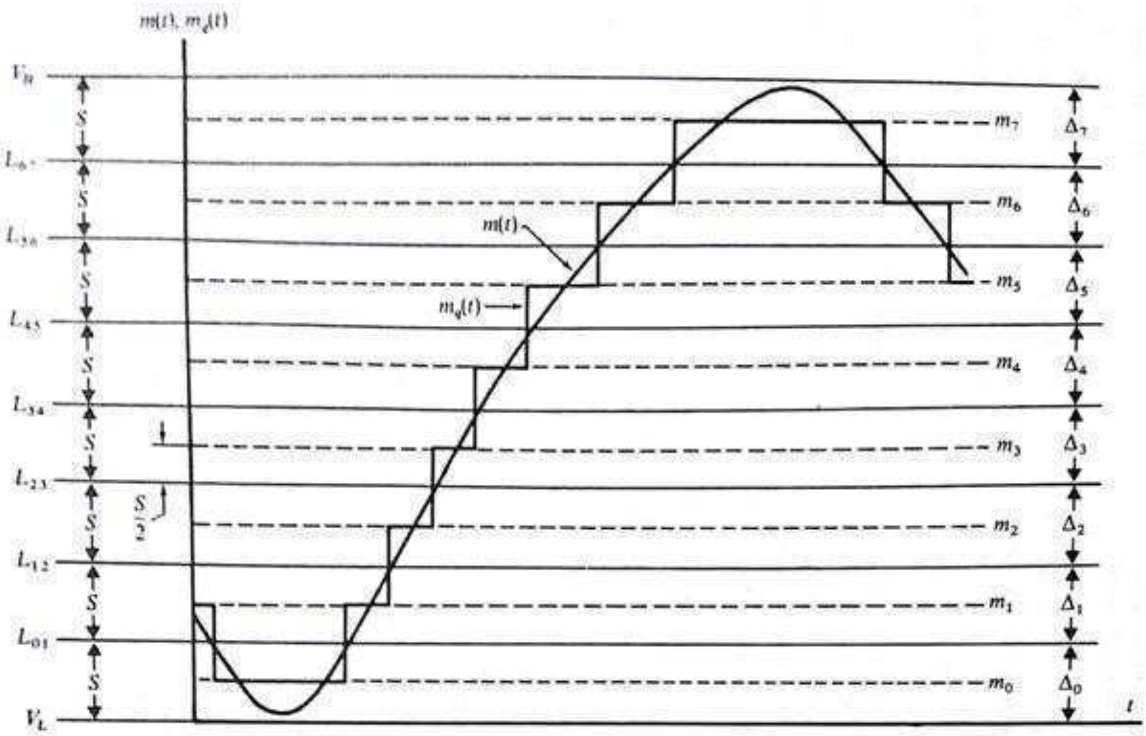


Figure.4.14. Quantization Process

Using quantization of signals, the effect of noise can be reduced significantly. The difference between $m(t)$ and $m_q(t)$ can be regarded as noise and is called quantization noise.

$$quantization\ noise = m(t) - m_q(t)$$

Also the quantized signal and original signal differs from one another in a ransom manner. This difference or error due to quantization process is called quantization error and is given by

$$e = m(t) - m_k$$

When $m(t)$ happens to be close to quantization level m_k, quantizer output will be m_k.

We can define the variable Δv to be the height of the each of the L levels of the quantizer as shown above. This gives a value of Δv equal to

$$\Delta v = \frac{2m_k}{L}$$

Therefore, for a set of quantizers with the same m_k, the larger the number of levels of a quantizer, the smaller the size of each quantization interval, and for a set of quantizers with the same number of quantization intervals, the larger m_p is the larger the quantization interval length to accommodate all the quantization range. The process of transforming sampled amplitude values of a message signal into a discrete amplitude value is referred to as Quantization. The quantization Process has a two-fold effect:

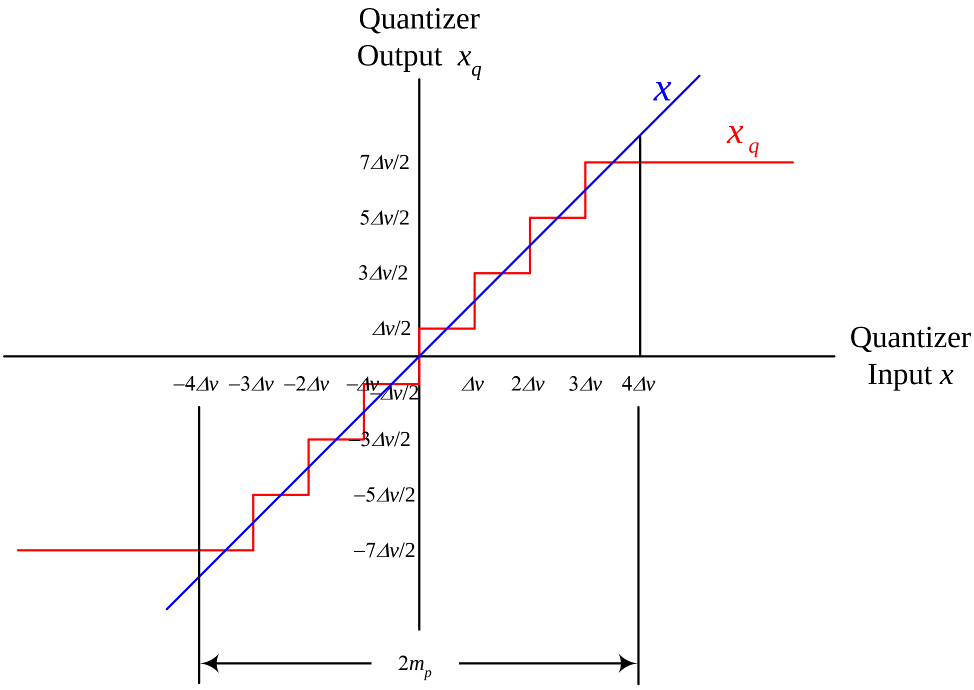
1. The peak-to-peak range of the input sample values is subdivided into a finite set of decision levels or decision thresholds that are aligned with the risers of the staircase, and
2. The output is assigned a discrete value selected from a finite set of representation levels that are aligned with the treads of the staircase.

A quantizer is memory less in that the quantizer output is determined only by the value of a corresponding input sample, independently of earlier analog samples applied to the input.

4.11.1 Quantization error:

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.

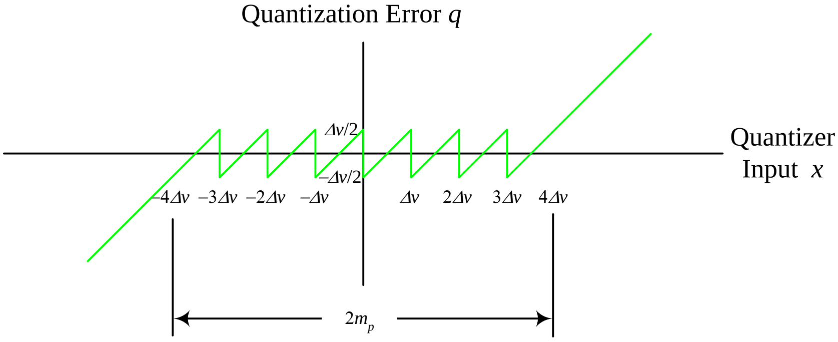
Now if we look at the input output characteristics of the quantizer, it will be similar to the red line in the following figure. Note that as long as the input is within the quantization range of the quantizer, the output of the quantizer represented by the red line follows the input of the quantizer. When the input of the quantizer exceeds the range of $-m_p$ to m_p, the output of the quantizer starts to deviate from the input and the quantization error (difference between an input and the corresponding output sample) increases significantly.



Now let us define the quantization error represented by the difference between the input sample and the corresponding output sample to be q, or

$$q = x - x_q$$

Plotting this quantization error versus the input signal of a quantizer is seen next. Notice that the plot of the quantization error is obtained by taking the difference between the blow and red lines in the above figure.



It is seen from this figure that the quantization error of any sample is restricted between $-\Delta v/2$ and $\Delta v/2$ except when the input signal exceeds the range of quantization of $-m_p$ to m_p.

4.12. Pulse Code Modulation (PCM) Transmitter:

A signal which is to be quantized before transmission is sampled as well. The quantization is used to reduce the effect of noise and the sampling allows us to do the time division multiplexing. The combined operation of sampling and quantization generate a quantized PAM waveform i.e. a train of pulses whose amplitude is restricted to a number of discrete levels.

Rather than transmitting the sampled values itself, we may represent each quantization level by a code number and transmit the code number. Most frequently the code number is converted in to binary equivalent before transmission. Then the digits of the binary representation of the code are transmitted as pulses. This system of transmission is called binary Pulse Code Modulation. The whole process can be understood by the following diagram.

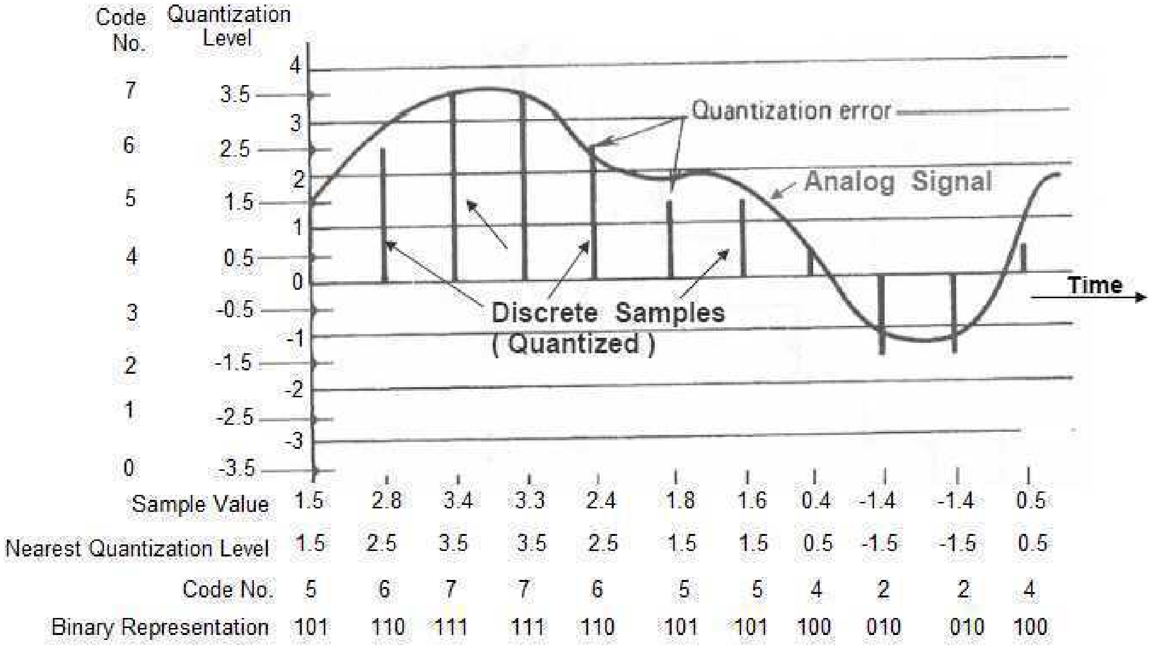


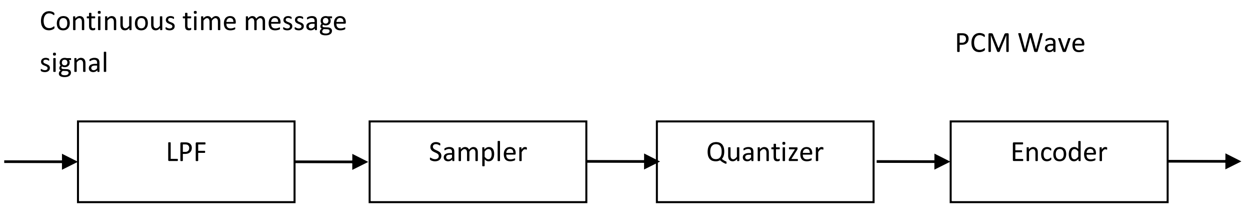
Figure.4.15. PCM Process

Basic Blocks:

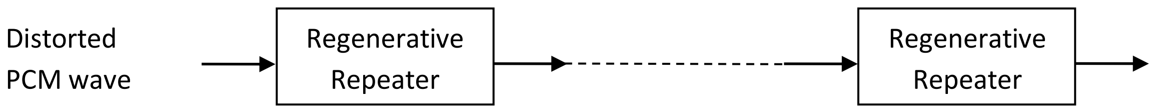
1. Anti aliasing Filter, 2. Sampler, 3. Quantizer, 4. Encoder

The block diagram of a PCM transmitter is shown in figure 4.16. An anti-aliasing filter is basically a filter used to ensure that the input signal to sampler is free from the unwanted frequency components. For most of the applications these are low-pass filters. It removes the frequency components of the signal which are above the cutoff frequency of the filter. The cutoff frequency of the filter is chosen such it is very close to the highest frequency component of the signal.

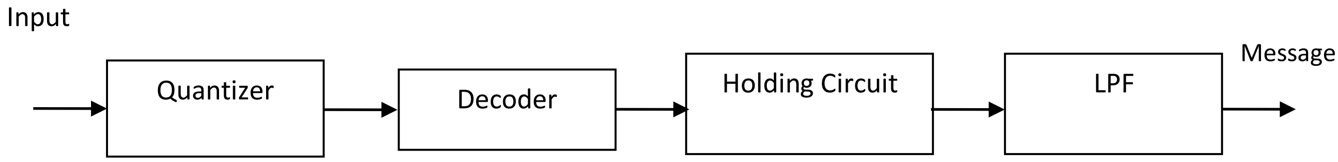
The message signal is sampled at the Nyquist rate by the sampler. The sampled pulses are then quantized by the quantizer. The encoder encodes these quantized pulses in to binary equivalent, which are then transmitted over the channel. During the channel the regenerative repeaters are used to maintain the signal to noise ratio.



(A) Transmitter



(b) Transmission Path



(c) Receiver

Figure.4.16. PCM System Basic Block Diagram

Figure (c) shows the receiver. The first block is again the quantizer, but this quantizer is different from the transmitter quantizer as it has to take the decision regarding the presence or absence of the pulse only. Thus there are only two quantization levels. The output of the quantizer goes to the decoder which is an D/A converter that performs the inverse operation of the encoder. The decoder output is a sequence of quantized pulses. The original signal is reconstructed in the holding circuit and the LPF.

4.13. Types of Quantizer: Uniform and Non-uniform quantization

4.13.1. Uniform Quantizer: In Uniform type, the quantization levels are uniformly spaced, where as in non-uniform type the spacing between the levels will be unequal and mostly the relation is logarithmic.

Types of Uniform Quantizer: (based on I/P - O/P Characteristics)

1. Mid-Rise type Quantizer
2. Mid-Tread type Quantizer

In the stair case like graph, the origin lies the middle of the tread portion in Mid –Tread type where as the origin lies in the middle of the rise portion in the Mid-Rise type.

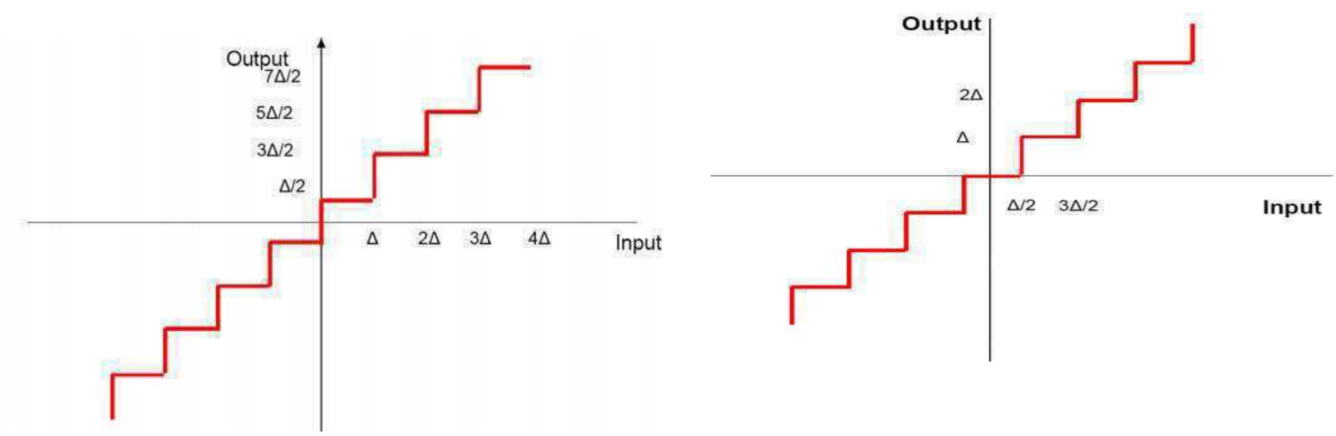


Figure.4.17 (a) Mid – Rise type: Quantization levels: even number (b) Mid – tread type :Quantization levels: odd number

4.13.2. Non-Uniform Quantizer 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 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. In Non - Uniform Quantizer the step size varies. 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 resultant signal is then transmitted.

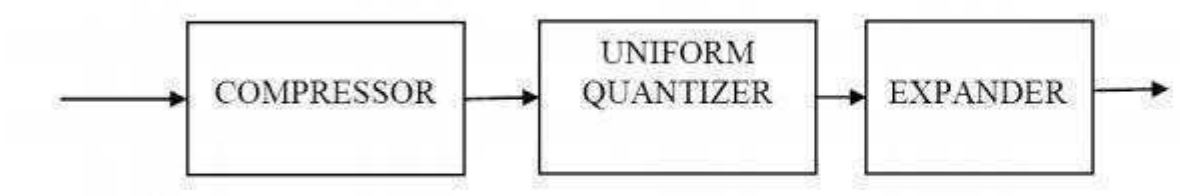


Figure.4.18. Non-uniform quantizer

At the receiver, a device with a characteristic complementary to the compressor called Expander is used to restore the signal samples to their correct relative level. The Compressor and expander take together constitute a Compander.

- The compressor will compress the dynamic range of the signal so that high dynamic range signal can be passed through components of low dynamic range capability, the uniform quantizer will undergo the quantization process of the compressed signal and the lastly the expander will undergo expansion and invert the compression function to reconstruct the original signal.
- The expander has complementary characteristics as that of compressor so that the compressor input is equal to expander output in order to reproduce the signal at the receiver.
- The Figure.4.19 below illustrates the input-output characteristics and curves of the companding process, and it can be seen that companding has linear characteristics.

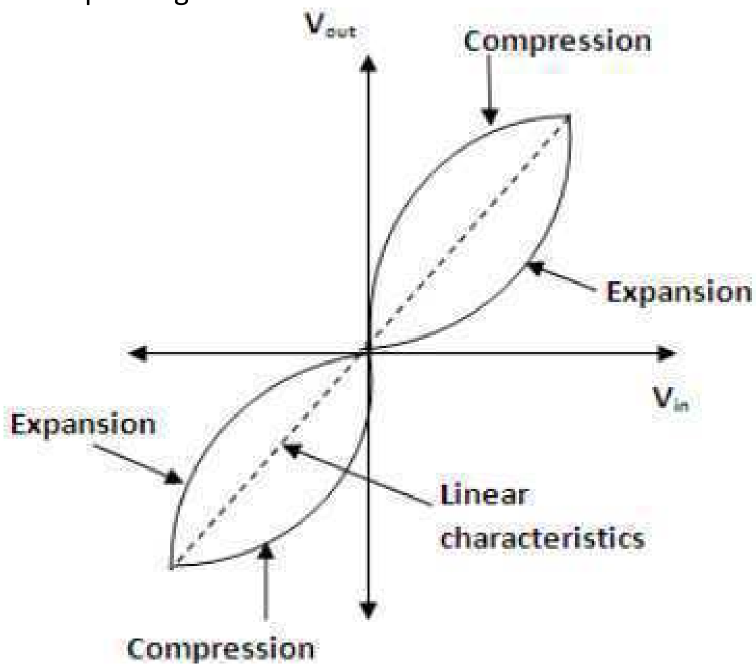


Figure.4.19. Input-output characteristics of Companding

Advantages of Non- Uniform Quantization:

1. Higher average signal to quantization noise power ratio than the uniform quantizer when the signal is non uniform which is the case in many practical situations.

2. RMS value of the quantizer noise power of a non – uniform quantizer is substantially proportional to the sampled value and hence the effect of the quantizer noise is reduced.

There are two types of Companding techniques.

- μ-law: North America and Japan. For $\mu = 255$ (for 8-bit codes),

$$y = \text{sgn}(x)\frac{1}{\ln(1+\mu)}\ln(1+\mu|x|)\ , \quad (0 < x < 1)$$

- A-law: Europe, rest of world.

$$y = \begin{cases} \text{sgn}(x)\dfrac{A|x|}{1+\ln(A)} & |x| < \dfrac{1}{A} \\ \text{sgn}(x)\dfrac{1+\ln(A|x|)}{1+\ln(A)} & \dfrac{1}{A} < |x| < 1 \end{cases}$$

The standard value is $A = 87.7$.

For both laws, the input to the compressor is

$$x = \frac{m(t)}{m_p}$$

where $-m_p \le m(t) \le m_p$.

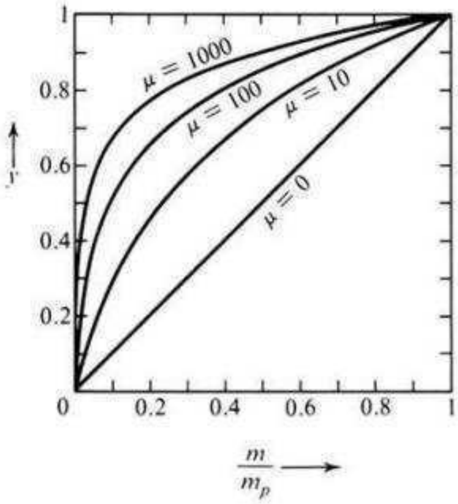
A-law Companding Technique

- 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

µ-law provides slightly larger dynamic range than A-law. A-law has smaller proportional distortion for small signals. A-law is used for international connections if at least one country uses it.



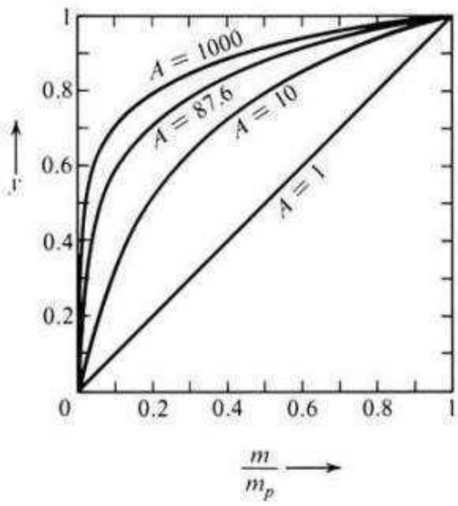


Figure.4.20. (a) μ-law curve (b) A-law curve

4.14. PCM Signal to noise ratio:

The signal to quantization noise ratio is given as

SNR = $\frac{S}{N_q} = \frac{Normalized\ signal\ power}{Normalized\ noise\ power}$

$$= \frac{Normalized\ signal\ power}{\frac{\Delta^2}{12}}$$

The number of quantization value is equal to: q=2^v

$$\Delta = \frac{2X_{max}}{2^v}$$

$$\frac{S}{N_q} = \frac{\text{Normalized signal power}}{\left[\frac{2X_{max}}{2^v}\right]^2 * \frac{1}{12}}$$

Let the normalized signal power is equal to P then the signal to quantization noise will be given by:

$$\boxed{\frac{S}{N_q} = \frac{3P * 2^{2v}}{X_{max}^2}}$$

4.15. Data rate: Bit rate is typically seen in terms of the actual data rate.

Bit Rate: The speed of the data is expressed in bits per second (bits/s or bps). The data rate R is a function of the duration of the bit or bit time (T_B). R = 1/T_B .Rate is also called channel capacity C.

Baud rate: Baud rate refers to the number of signal or symbol changes that occur per second. A symbol is one of several voltage, frequency, or phase changes. If N is the number of bits per symbol, then the number of required symbols is S = 2^N. Thus, the gross bit rate is:

R = baud rate x $\log_2 S$ = baud rate x 3.32 $\log_{10} S$

Multiplexed PCM signal:

When a large number of PCM signals are to be transmitted over a common channel, multiplexing of these PCM signals is required. Figure shows the basic time division multiplexing scheme, called as the PCM multiplexed digital system**.** This system has been designed to accommodate N voice and each signal is band limited to F_m kHz, and the sampling is done at a standard rate of $2F_m$ kHz. This is Nyquist rate. The sampling is done by the commutator switch SW1. These voice signals are selected one by one and connected to a PCM transmitter by the commutator switch SW1. Each sampled signal is then applied to the PCM transmitter which converts it into a digital signal by the process of A to D conversion and companding. The resulting digital waveform is transmitted over a co-axial cable. Periodically, after every constant distance, the PCM-TDM signal is regenerated by amplifiers called "repeaters". They eliminate the distortion introduced by the channel and remove the superimposed noise and regenerate a clean PCM-TDM signal at their output. This ensures that the received signal is free from the distortions and noise. At the destination the signal is companded, decoded and demultiplexed, using a PCM receiver. The PCM receiver output is connected to different low pass filters via commutator switch SW2. Synchronization between the transmitter and receiver commutator SW1 and SW2 is essential in order to ensure proper communication.

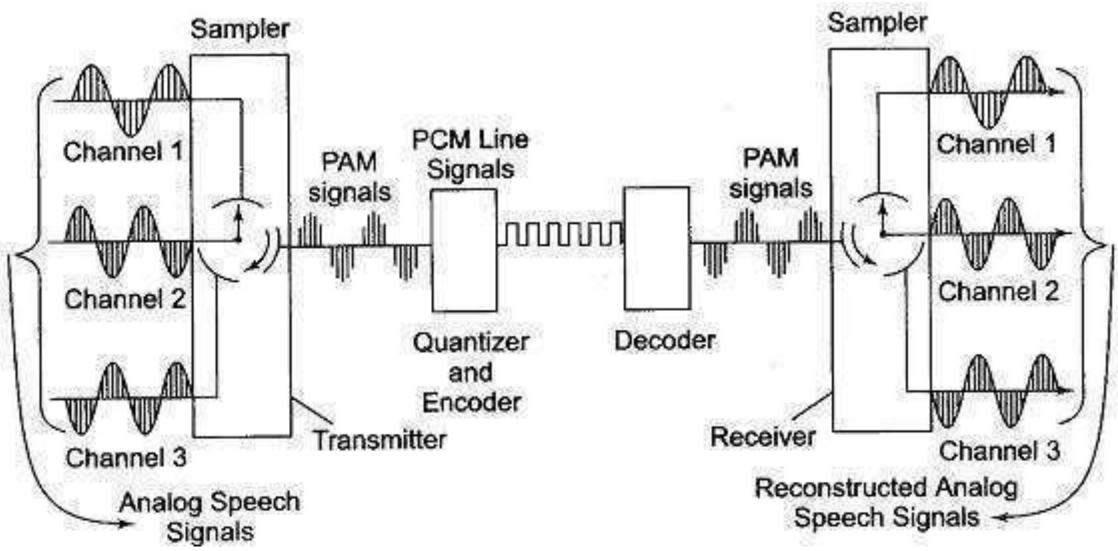


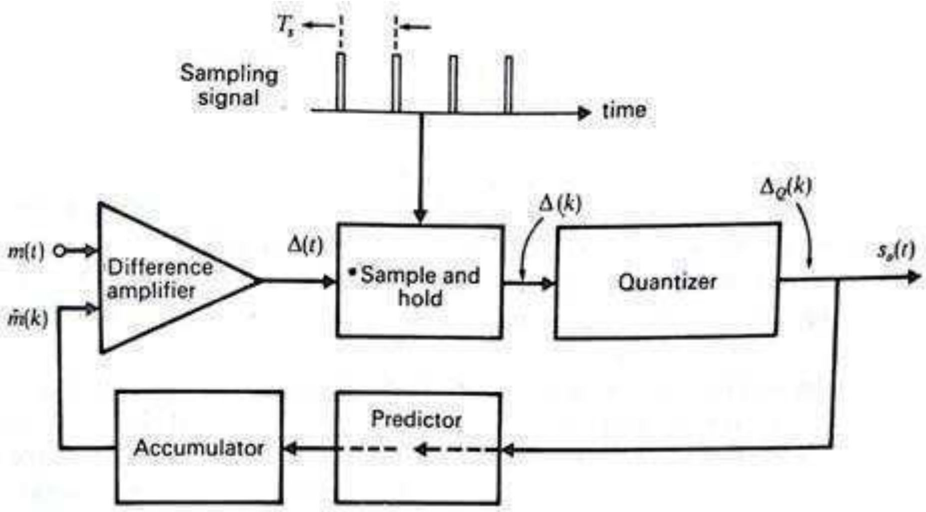
Figure.4.20. Multiplexed PCM TDM system

4.16. Differential Pulse Code Modulation (DPCM):

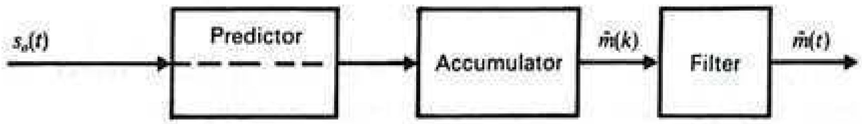
In DPCM instead of transmitting the sampled values itself at each sampling time; we can transmit the difference between the two successive samples. If such changes are transmitted then at the receiving end we can generate a waveform identical to the m(t) by simply adding up these changes.

The DPCM has the special merit that when these differences are transmitted by PCM. The differences $m(k) - m(k-1)$ will be smaller than the sample values them and fewer levels will be required to quantize $m(k)$, and corresponding fewer bits will be needed to encode the signal. The basic principle of DPCM is shown in figure.

The receiver consists of an accumulator which adds-up the receiver quantized differences $\Delta_Q(k)$ and a filter which smoothes out the quantization noise. The output of accumulator is the signal approximation $\widehat{m}(k)$ which becomes $\widehat{m}(t)$ at the filter output.



(a) Transmitter



(b) Receiver

Figure.4.21. Differential PCM

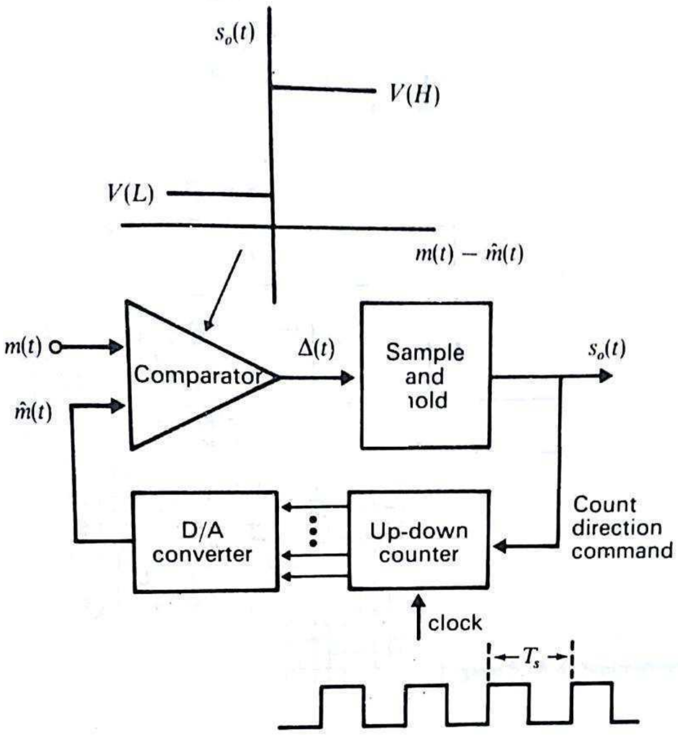
At the transmitter we need to know whether the $\widehat{m}(t)$ is larger or smaller than $m(t)$ and by how much amount. We may than determine whether the next difference $\Delta_Q(k)$ needs to be positive or negative and of what amplitude in order to bring $\widehat{m}(t)$ as close as possible to $m(t)$. For this reason we have a duplicate accumulator at transmitter.

At each sampling time the transmitter difference amplifier compares $m(t)$ and $\widehat{m}(t)$, and the sample and hold circuit holds the result of that comparison $\Delta(t)$, for the duration of interval between sampling times. The quantizer generates the signal $S_0(t) = \Delta_Q(k)$ both for the transmission to the receiver and to provide the input to the receiver accumulator in the transmitter. The basic limitation of the DPCM scheme is that the transmitted differences are quantized and are of limited values.

4.17. Delta Modulation (DM):

Delta Modulation is a DPCM scheme in which the difference signal $\Delta(t)$ is encoded into just a single bit. The single bit providing just for two possibilities is used to increase or decrease the estimate $\widehat{m}(t)[m_q(t)]$. The baseband signal $m(t)$ and its quantized approximation $\widehat{m}(t)$ are applied as input to a comparator. The comparator has one fixed output V(H) when $m(t) > m_q(t)$ and a difference output V(L) when $m(t) < m_q(t)$. Ideally the transition between V(H) and V(L) is arbitrarily abrupt as $m(t) - m_q(t)$ passes through zero. The up-down counter increments or decrements its count by 1 at each active edge of the clock waveform. The count direction i.e. incrementing or decrementing is determined by the voltage levels at the "Count direction command" input to the counter. When this binary input is at level V(H), the counter counts up and when this binary input is at level V(L), the counter counts down.

The digital output of the counter is converted into analog quantized approximation $m_q(t)$ by a D/A converter. The waveforms for the delta modulator is shown in figure (b), assuming that the active clock edge is falling edge. The Linear Delta Modulator is shown in figure (a).



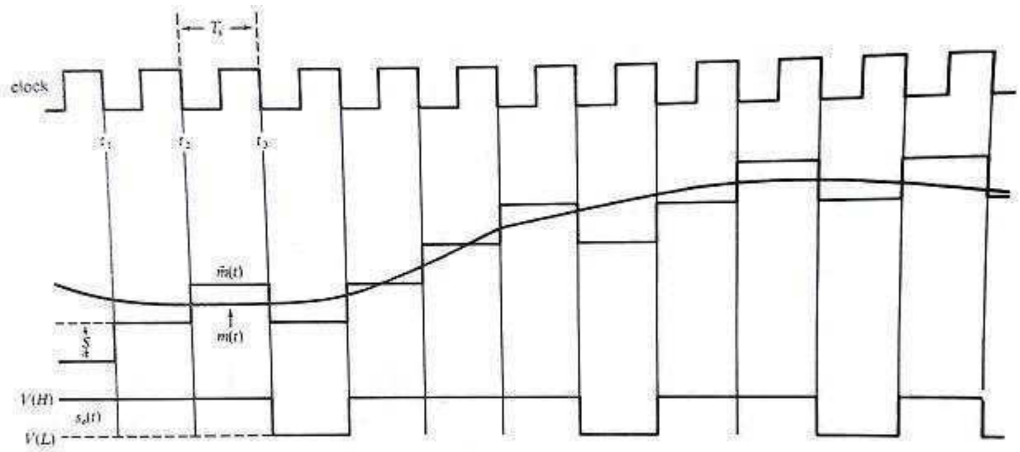
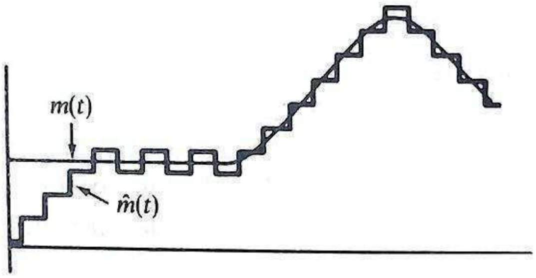


Figure.4.22. (a) Delta Modulator (b) The response of the delta modulator to a baseband signal m(t)

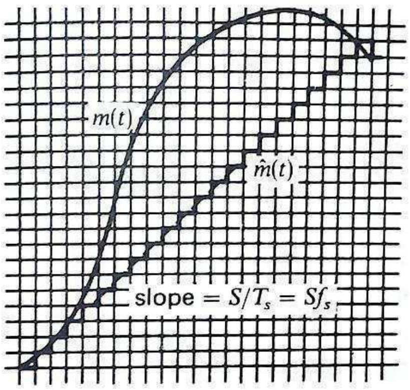
4.17.1. Delta Modulation and the slope overload error.

It may be noted that at startup there is a brief interval when the quantized signal may be a poor approximation to the baseband signal as shown in figure (a).

The initial large discrepancy between $m(t)$ and $m_q(t)$ and stepwise approach of $m_q(t)$ to $m(t)$ is shown in figure (b).



(a) Startup response of DM



(b) Slope Overload in a linear DM

Figure.4.23. Response of DM and slope overload distortion

It should be noted that when $m_q(t)$ has caught up $m(t)$ and even though $m(t)$ remains constant, $m_q(t)$ hunts, swinging up and down to $m(t)$.

4.17.2. Slope Overload distortion

The excessive disparity between $m(t)$ and $m_q(t)$ is described as a slope overload error and occurs whenever $m(t)$ has a slope larger than the slope S/T_s which can be sustained by the waveform $m_q(t)$. The slope overload as shown in figure (b) is developed due to the small size of S. To overcome the overload we have to increase the sampling rate above the rate initially selected to satisfy the Nyquist criterion. The sampling rate f_s must satisfy the following condition

$$sf_s = 2f_s = \pi f A$$

4.17.3. Features of DM: Following are some of the features of delta modulation.

- An over-sampled input is taken to make full use of the signal correlation.
- The quantization design is simple.
- The input sequence is much higher than the 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.
- This involves simpler implementation.

4.17.4. Advantages of DM Over DPCM

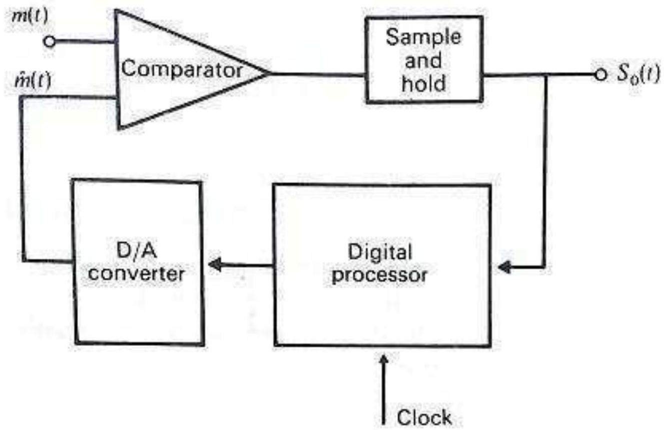
- 1-bit quantizer
- Very easy design of the modulator and the demodulator. However, there exists some noise in DM.
- Slope Over load distortion (when Δ is small) & Granular noise (when Δ is large)

4.18. Adaptive Delta Modulation (ADM).

In digital modulation, we have come across certain problem of determining the step-size, which influences the quality of the output wave.

A larger step-size is needed in the step slope of modulating signal and a smaller step size is needed where the message has a small slope. The minute details get missed in the process. So, it would be better if we can control the adjustment of step-size, according to our requirement in order to obtain the sampling in a desired fashion. This is the concept of Adaptive Delta Modulation.

Following is the



a: Adaptive Delta Modulation (ADM)

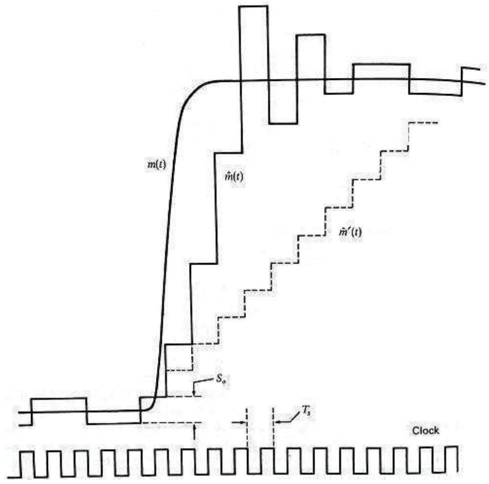


Figure b: Waveforms comparing response of DM and ADM

Figure.4.24. Block diagram of Adaptive delta modulator

The step size S is not of fixed size but it is always a multiple of basic step size S_0. The basic step size S_0 is either added or subtracted by the accumulator as required to move $m_q(t)$ more close to $m(t)$. If the direction of the step at the clock edge K is same as at edge K-1, then the processor increases the step size by an amount S_0. If the directions are opposite then the processor decreases the magnitude of the step by S_0.

In figure (a), the output $S_0(t)$ is called $e(k)$, which represents the error i.e. the discrepancy between the $m(t)$ and $m_q(t)$, and it is either V(H) or V(L).

$e(k) = +1, if\ m(t) > m_q(t)$ immediately before Kth edge,

$e(k) = -1, if\ m(t) < m_q(t)$ immediately before Kth edge,

The features of ADM are shown in figure (b) As long as the condition $m(t) > m_q(t)$ persists the jumps in $m_q(t)$ becomes larger, that's why $m_q(t)$ catches up with $m(t)$ sooner than in the case of linear DM, as shown by $m'_q(t)$.

On the other hand, when the response to the large slope in $m(t), m_q(t)$ develops large jumps and large number of clock cycles are required for these jumps to settle down. Therefore the ADM system reduces the slope overload but it increases the quantization error. Also when $m(t)$ is constant $m_q(t)$ oscillates about $m(t)$ but the oscillation frequency is half of the clock frequency.

4.18. Comparison of various systems:

S.NO	Parameter of Comparison	Pulse Code Modulation (PCM)	Delta Modulation (DM)	Adaptive Delta Modulation (ADM)	Differential Pulse Code Modulation (DPCM)
1.	Number of bits	It can use 4,8, or 16 bits per sample.	It uses only one bit for one sample	It uses only one bit for one sample	Bits can be more than one but are less than PCM.
2.	Levels and step size	The number of levels depends on number of bits. Level size is fixed.	Step size is kept fixed and cannot be varied.	According to the signal variation, step size varies.	Number of levels is fixed.
3.	Quantization error and distortion	Quantization error depends on number of levels used.	Slope overload distortion and granular noise are present.	Quantization noise is present but other errors are absent.	Slope overload distortion and quantization noise is present.
4.	Transmission bandwidth	Highest bandwidth is required since numbers of bits are high.	Lowest bandwidth is required.	Lowest bandwidth is required.	Bandwidth required is less than PCM.
5.	Feedback	There is no feedback in transmitter or receiver.	Feedback exists in transmitter.	Feedback exists.	Feedback exists.
6.	Complexity of Implementation	System is complex.	Simple	Simple	Simple

