Chapter 3 PULSE MODULATION SYSTEMS

Introduction

The main purpose of a communication system is to deliver the message signal from an information source to destination located at some distance using modulation. Modulation techniques are used to modify some parameters of a carrier signal according to the message signal into a form suitable for transmission over the channel. Depending on the nature of the message signal modulation techniques can be broadly classified in to two categories as shown in Fig. 3.1.

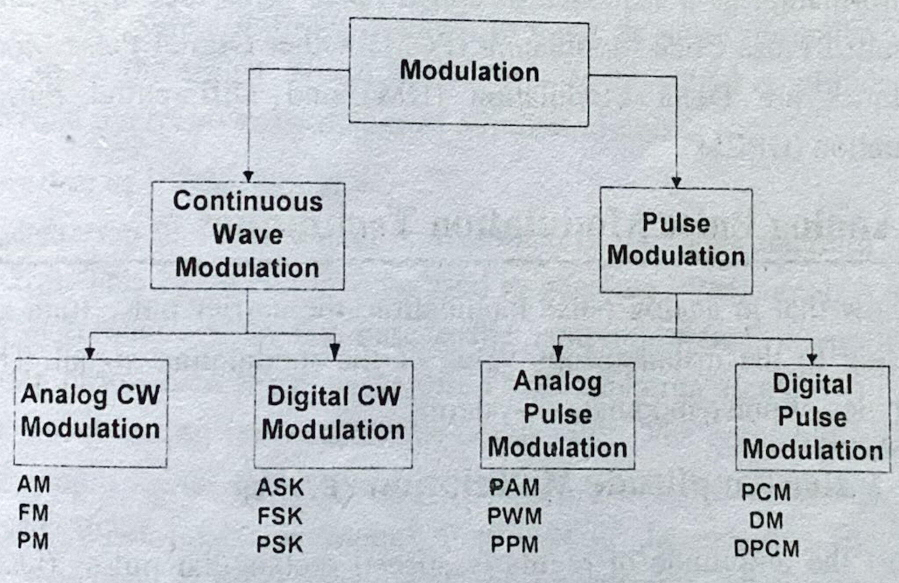


Fig. 3.1: Modulation Techniques

In continuous wave modulation, the carrier signal used for modulation is continuous in nature. The sub division of continuous wave modulation technique depending on the nature of message signal. If the message signal is analog in nature (i.e., continuous signal), than the modulation is analog continuous wave modulation. However, if the message signal is digital in nature (i.e., digital signal), than the modulation technique used is digital continuous wave modulation. In both analog and digital continuous wave modulation the carrier is analog in nature. Examples of analog continuous wave modulation are Amplitude Modulation (AM), Frequency Modulation

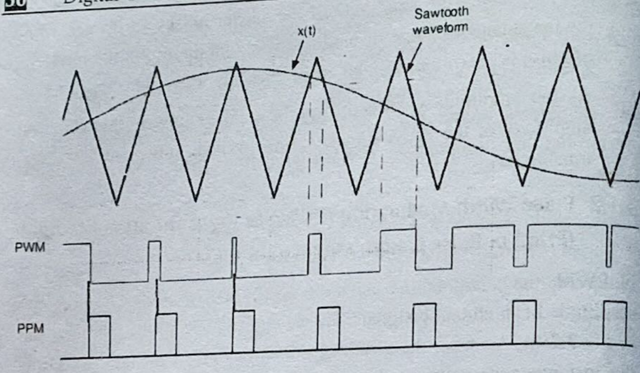


Fig. 3.2: (a) PWM and PPM generator. (b) Waveform of PWM and PPM signal.

Detection of PWM signal

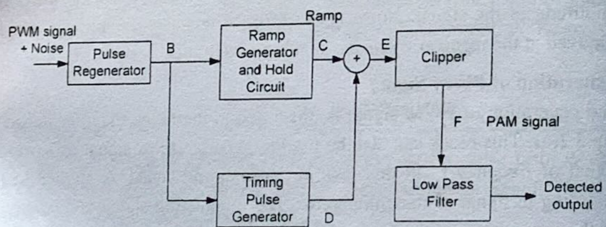


Fig. 3.3: PWM detection circuit

The detection block for the PWM signal is shown in Fig. 3.3. The PWM signal, corrupted by the additive noise is applied to the pulse regenerator circuit which regenerates the PWM signal by limiting the amplitude. Thus, some if the noise is removed and the pulses are squared up. The regenerated pulses are applied to a timing pulse generator that generates a train of constant amplitude, width pulses. The pulses are synchronized with the leading edges of the regenerated PWM pulses but delayed by a fixed interval. The regenerated PWM pulses at the output of pulse regenerator, is also applied to a ramp generator, whose output is a constant slope ramp signal for the duration of the pulse. The height of the ramp signal is thus proportional to the width of the PWM pulses which is maintained by hold circuit until the trailing edge of the reference pulse generated by timing pulse generator.

The constant amplitude pulses at the output of timing pulse generator are then added to the ramp signal. The output of the adder is then clipped off at a threshold level to generate a PAM signal at the output of the clipper. A low pass filter is used to recover the original modulating signal back from the PAM signal. The waveforms for this circuit are shown in Fig.3.4.

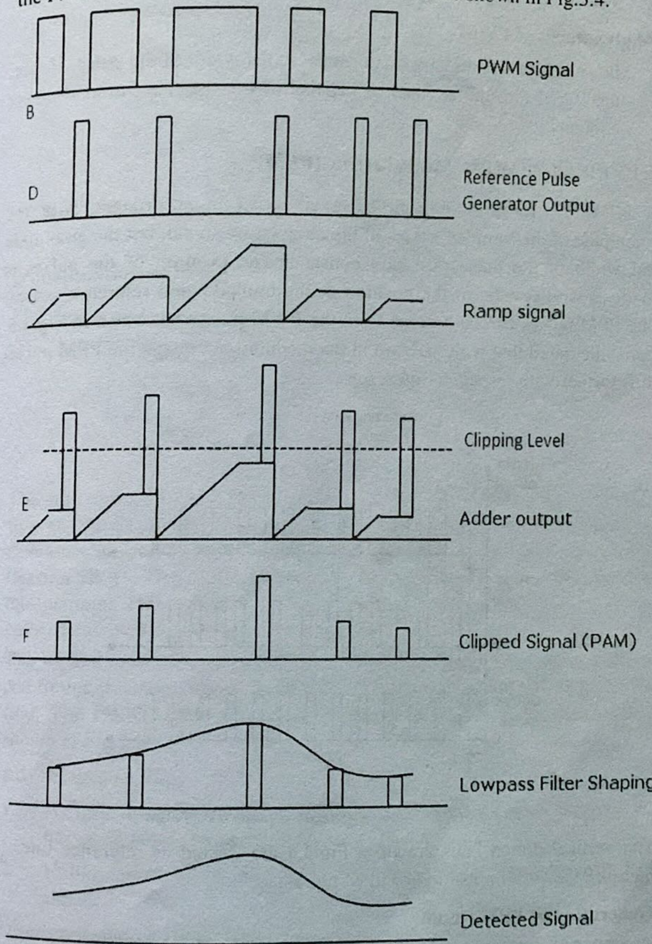


Fig.3.4: Waveforms for PWM detection circuit

Advantages of PWM

1. As width of a pulse is not affected by noise, PWM has very good noise immunity.
2. Transmitter and receiver synchronizations is not required.

Disadvantages of PWM

1. Power content is directly proportional to the width of the pulse. Hence, the PWM transmitter must be capable of transmitting variable power contents.

3.13 Pulse Position Modulation (PPM)

In PPM, the position of each pulse is varied in accordance with the magnitude of the sampled values of the modulating signal, but the amplitude and width of the pulses are kept constant. The position of the pulses is changed with respect to the position of the unmodulated reference pulses. The PPM pulses can be derived from the PWM pulses as shown in Fig.3.5. It may be noted that with increase in the modulating voltage the PPM pulses shift further with respect to reference.

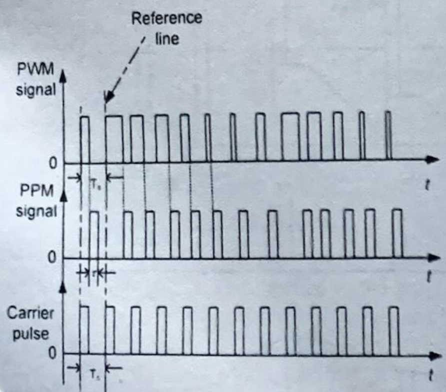


Fig.3.5: PPM pulses generated from PWM signal

The vertical dotted lines drawn in Fig.3.5 are treated as reference lines to measure the shift in position of PPM pulses.

Generation of PPM signal

The PPM signal is generated by passing the PWM signal through a negative edge triggered monostable multivibrator as shown in Fig.3.6. Hence, corresponding to each trailing edge of PWM signal, the monostable output goes high. It remains high for a fixed time decided by its own RC components. Thus, as the trailing edges of the PWM signal keeps shifting in proportion with the modulating signal x(t), the PPM pulses also keep shifting, as shown in Fig.3.5.

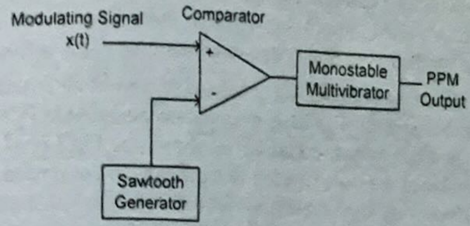


Fig.3.6: Generation PPM signal

Demodulation of PPM

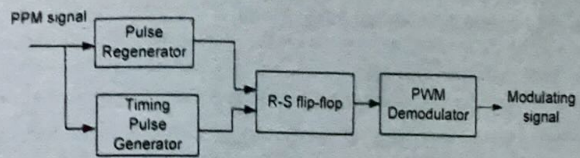


Fig.3.7: PPM demodulator circuit

The contaminated PPM signal received by the PPM demodulator circuit is first amplitude limited to reduce the effect of noise. The pulse regenerator developes a clean PPM at its output and applies these pulses to the reset pin (R) of a SR flip-flop. A fixed periodic pulse (timing pulse) is generated from the incoming PPM waveform, and the SR flip flop is set by the reference pulses generated by timing pulse generator.

The set and reset process due to the output signal from the pulse regenerator and timing pulse generator, generates a PWM signal at the output of SR flip-flop. The PWM signal can be demodulated using the PWM demodulator shown in Fig.3.3.

Advantages of PPM

1. As information carried by PPM signal is independent of amplitude, it has good noise immunity.
2. Unlike PWM, the transmitted power always remains constant in PPM signal.

Disadvantages of PPM

1. Synchronizing pulses is required to track the variations of PPM pulses positions with respect to a reference pulse.
2. Large bandwidth is required to ensure transmission of undistorted pulses.

3.2 Digital Pulse Modulation Techniques

After sampling of an analog message signal, the next step is the generation of a coded version (digital representation) of the signal. The output of sampling process is a discrete signal, which is changed into digital format, and a code is provided to each digital level. Pulse Code Modulation (PCM) provides one method of digital coded representation of an analog signal.

Pulse Code Modulation (PCM) is a technique of converting analog signals into digitally encoded signals. The basic essential operations in a PCM system are sampling, quantization and encoding as shown in Fig.3.8(a). These three processes combined are called analog to digital (A/D) conversion. The reverse process of recovering message signal from PCM is called digital to analog (D/A) conversion and the steps are shown in Fig.3.8(b).

The PCM is not a modulation in conventional sense. Modulation is the process in which some parameters of carrier are varied according to instantaneous value of the message signal. In PCM, the only section in which this happens is while sampling.

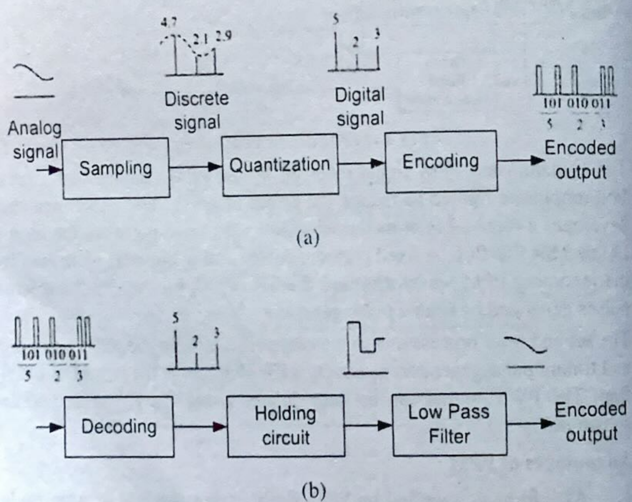


Fig.3.8: (a) Block diagram of PCM encoder. (b) Block diagram of PCM decoder.

The basic signal processing operations involved in PCM are.

3.2.1 Sampling

The first step of analog to digital conversion or PCM is the sampling. The incoming message signal (analog signal) is sampled with a train of narrow rectangular pulses (as studied under flat top sampling in chapter 2).

3.2.2 Quantization

An analog signal, such as voice, has a continuous range of amplitudes and therefore its samples covers a continuous amplitude range. In other words, within a finite amplitude range of signal we find an infinite number of amplitude levels.

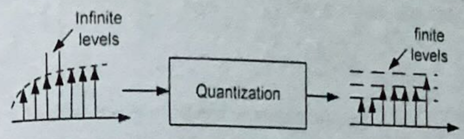


Fig.3.9: Operation of quantization

In practice, it is not necessary to transmit the exact amplitude of the samples to recover original signal. Human senses (ears or eyes) can detect only finite intensity differences and it means that the original analog signal may be approximated with high level of accuracy by a signal constructed of discrete amplitudes. The existence of a finite number of discrete amplitude levels is a basic condition of PCM. Clearly, if we assign the discrete amplitude levels with sufficiently close spacing, we may make the approximated signal practically indistinguishable from the original analog signal.

Quantization is the process of transforming the sample amplitude $x(nT_s)$ of a message signal $x(t)$ at time $t = nT_s$ into a discrete amplitude $x_q(nT_s)$ taken from a finite set of possible amplitudes as shown in Fig.3.9.

Quantization can be uniform or non-uniform. In uniform quantization, the representation levels are uniformly spaced between the minimum and maximum values of the discrete sample, whereas in non-uniform quantization the representation levels are not uniform.

1. **Uniform quantization**

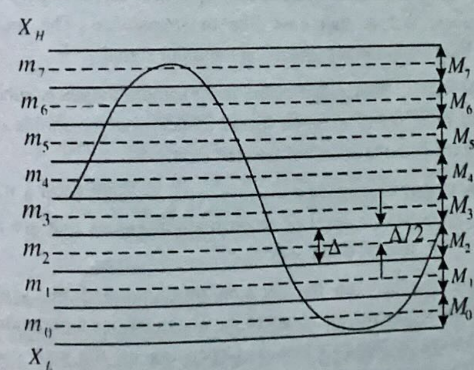


Fig.3.10: Uniform Quantization

Fig. 3.10 shows the quantization levels of a message signal $x(t)$. The operation of uniform quantization can be described in steps given below:

- Let us consider a message signal $x(t)$, with peak value X_H and X_L. The separation between X_H and X_L (called dynamic range) are divided into N equal intervals each of size Δ. Where Δ is called the step size and is given by

$$\Delta = \frac{X_H - X_L}{N}$$

  In Fig.3.11. a specific example has been given with $N = 8$ levels. At center of each of these steps, we allocate quantization level denoted by $m_o, m_1, m_2, \ldots, m_7$.

- Now the quantized signal $x_q(t)$ is generated by allocating the quantization levels to the message signal that falls within its range as follows. The message signal within the range of M_o is represented by a single level m_o, and that in the range of M_1, can by single level m_1, and so on.

- Thus we can conclude that at every instant, the message signal $x(t)$ does not change, but only jumps from one level to another level. For example, till the value of message signal is in the range M_o; the output of quantized level is m_o; and when message signal changes from M_o to M_1, the output of quantization will jump to output m_1.

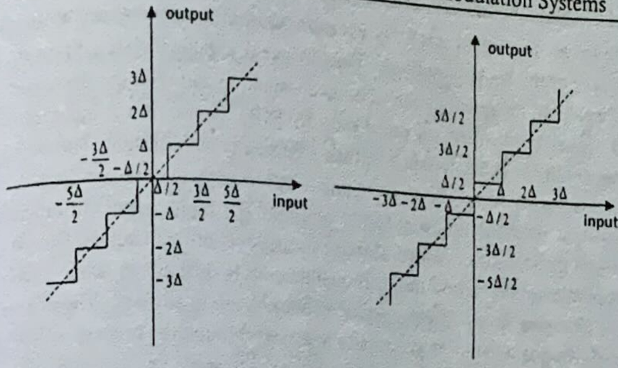
Types of uniform quantization

Graphically, the quantizing process means that a linear relation between input and output of an analog system is replaced by a transfer characteristic that is stair case like in appearance. The quantizing process has two following effects on the input signal.

i. The peak-to-peak range of the input sample values is subdivided into finite set of decision levels or decision thresholds that are aligned with the "riser" of the stair case.

ii. The output is assigned a discrete value selected from a finite set of representation level or reconstruction values that are aligned with the "trends" of the stair case.

The separation between the decision threshold and the separation between the representation levels of the quantizer has a common value and is known as the step size Δ. Depending on the stair case like transfer characteristics, the uniform quantizer is divided into two types



(a) (b)

Fig. 3.11: (a)Midtread type. (b) Midriser type

i. **Symmetric quantizer of the midtrend type**

According to the staircase-like characteristics of Fig. 3.11(a). the decision thresholds of quantizer are located at $\pm\frac{\Delta}{2}, \pm\frac{3\Delta}{2}, \pm\frac{5\Delta}{2}, \ldots$ and the representation levels are located at $0, \pm\Delta, \pm 2\Delta, \ldots$ where Δ is the step size. A uniform quantizer characterized in this value is referred to as a symmetric quantizer of midtread type quantizer because the origins lies in the middle of a tread of the staircase quantizer.

ii. **Symmetric quantizer of the midriser type**

Fig. 3.11(b) shows another staircase-like transfer characteristics, in which the decision thresholds of the quantizer are located at 0, $\pm\Delta, \pm 2\Delta, \ldots$ and the representation levels are located at $\pm\frac{\Delta}{2}, \pm\frac{3\Delta}{2}, \pm\frac{5\Delta}{2}, \ldots$ where Δ is the step size. A quantizer having this characteristics is referred to as a symmetric quantizer of the midriser type because in this case the origin lies in the middle of the riser of the staircase.

These midtread or midriser type quantizer output is determined only by the value of a corresponding input sample. This value of output is totally independent of earlier (or later) analog samples applied to the input.

2. **Non-uniform quantization**

In uniform quantization, once the step size is fixed, the quantization noise power remains constant and depends only on the step size.

However, the signal power is not constant and is proportional to the square of the signal amplitude. Thus for weak signals the signal power is weak but the quantization noise is constant, resulting in the decrease of signal to quantization noise ratio (SQNR). Non-uniform quantizer has nonlinear characteristics and the step size is not constant. The step size is varied according to the signal level to keep the average SQNR constant over the entire dynamic range of the input signal. In other words the step size is made adaptive to the level of input signal. Direct implementation of non-uniform quantization is difficult to achieve as the changing levels of input signal is not known in advance. The non-uniform quantization is practically achieved through the process called companding. Companding is derived from two words i.e., compression and expansion shown in Fig.3.12.

Thus, non-uniform quantinization is basically a compressor followed by a uniform quantization. At the receiver, the signal is expanded.

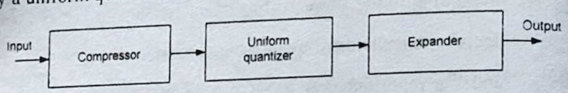


Fig.3.12: Companding Process

The input-output characteristics of non-uniform quantizer is shown in Fig. 3.13. It shows the companding characteristics which is the combination of the compressor and expander characteristics.

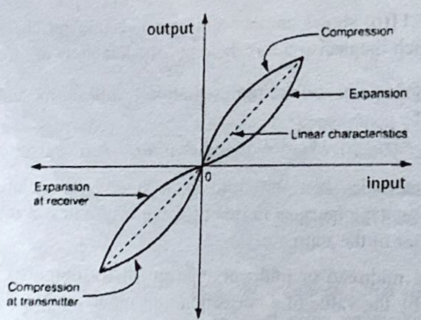


Fig.3.13: Compander characteristics

The compressor curve provides a higher gain to the weak signal and smaller gain to the strong input signals. However, the expander characteristics is exactly the inverse of the compressor characteristics resulting in the recovery of the original amplitudes at the receiver.

Types of companding characteristics

Two compression laws widely used in communication are

i. **μ law of companding**

The expression for compressed output "y" for given "x" is

$$|y_n| = \frac{|F(x)|}{x_{max}} = \frac{\log\left(1+\mu\frac{|x|}{x_{max}}\right)}{\log(1+\mu)} \quad \text{where } 0 \le \frac{|x|}{x_{max}} \le 1 \tag{3.1}$$

$$|y_n| = \frac{|F(x)|}{x_{max}} = \frac{\log(1+\mu|x_n|)}{\log(1+\mu)} \quad \text{where } 0 \le \frac{|x|}{x_{max}} \le 1 \tag{3.1}$$

In Eq.(3.2), x_n and y_n are the normalized input and output, and is only for positive amplitude. For representation of both positive and negative amplitude of the input and output, signam signal ($sgn(x)$) is used resulting in Eq.(3.3)

$$|y_n| = \frac{|F(x)|}{x_{max}} = Sgn(|x_n|)\,\frac{\log(1+\mu|x_n|)}{\log(1+\mu)} \quad \text{where } -1 \le |x_n| \le 1 \tag{3.3}$$

The compression characteristics $F(x)$ is continuous, with approximate linear characteristics for lower input level of x and logarithmic characteristics for higher input levels. When $\mu=0$, it is a special case of μ law and is a simple uniform quantization.

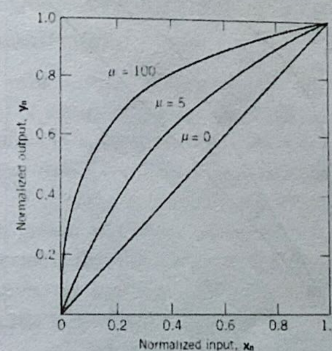


***Fig.3.14:** μ law compression characteristics*

The normalized form of μ law compression characteristics, y_n, is shown in Fig.3.14 for $0 \le |x_n| \le 1$, along with curves for three different values of μ. A practical value of μ is 255. The μ law is used for PCM telephone systems in United States, Canada, and Japan.

ii. **A law of companding**

In A law companding for a given input "x" the expression for compressed output "y" is given by

$$|y_n| = \frac{F(|x|)}{x_{max}} = \begin{cases} \dfrac{A\dfrac{|x|}{x_{max}}}{1+\log A}, & 0 \le \dfrac{|x|}{x_{max}} \le \dfrac{1}{A} \\[2ex] \dfrac{1+\log A\dfrac{|x|}{x_{max}}}{1+\log A}, & \dfrac{1}{A} \le \dfrac{|x|}{x_{max}} \le 1 \end{cases} \tag{3.4}$$

$$|y_n| = F(|x_n|) = \begin{cases} \dfrac{A|x_n|}{1+\log A}, & 0 \le |x_n| \le \dfrac{1}{A} \\[2ex] \dfrac{1+\log A|x_n|}{1+\log A}, & \dfrac{1}{A} \le |x_n| \le 1 \end{cases} \tag{3.5}$$

In Eq.(3.5) x_n and y_n are the normalized input and output, and is only for positive amplitude. The compression characteristics of $F(x)$ is piecewise, with linear segment for lower input level of x and logarithmic characteristics for higher input levels.

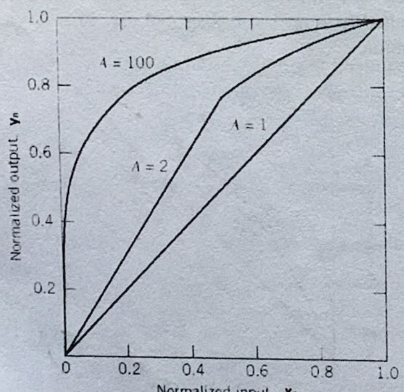


Fig.3.15: A law compression characteristics

The special case of A law is A = 1 and is a simple uniform quantization. The normalized form of A law compression characteristics, y_n, is shown in Fig.3.15 for $0 \le |x_n| \le 1$, along with curves for three different values of A . A practical value of A is 87.56. The A law companding is used for PCM telephone systems in Europe.

Both the μ law and A law curve have odd symmetry about the vertical axis.

3.2.3 Encoding

After sampling and quantization, the analog message signal becomes limited to a discrete set of amplitude values, but it is still not suited to transmission over a telephone line or radio link. The quantized samples are further converted into the combination of known set of signals so as to make them more robust to noise, interference, and other channel impairments.

One of the mostly commonly used encoding is a binary. In binary code each discrete sample value is assigned a unique set (codeword) of combination of 1' and 0's, each 1 and 0 being called a bit (binary digit) called codeword. If a fixed length (i.e., the number of 1 or 0 in any codeword of the code is fixed and equal) codeword consists of n bits, then, a total of 2^n distinct numbers or levels can be represented by the code. For example, if the sampled values are quantized into one of available 8 levels, than each level can be represented by 2 bit codeword (i.e.. 2^3=8).

The last signal-processing operation in the transmitter is line coding, the purpose of which is to represent each binary codeword by a sequence of pulses; for example, symbol 1 is represented by the presence of a pulse of fixed amplitude and duration and symbol 0 is represented by absence of the pulse.

3.2.4 Signal to Quantization Noise ratio for a PCM system (Linear quantization)

The quantization error (Q-error) is the difference between the input signal level and the level of the quantized version. These errors will appear at the output of the receiver as noise superimposed with the desired signal. As a result the quality of the received signal will degrade. This degradation is measured interms of signal to quantization noise ratio (SQNR). Higher the quantization error, greater will be the quantization noise and lower will be the SQNR. It is evident that the maximum Q-error could be only Δ/2. In uniform quantization, the step size Δ is constant for the entire dynamic range of input discrete signal level.

Q-error and hence the Q-noise is produced during the process of quantization because of rounding-off of the sampled value of a continuous message signal to the nearest representation level.

Mean square quantization error in PCM

In PCM the maximum Q-error (q_e) is ± Δ/2, where Δ is the step size. (i.e., q_e can assume any value between ±Δ/2 with equal probability). This gives a uniform PDF given by (Fig.3.16)

$$f_Q(q_e) = \frac{1}{b-a} = \frac{1}{(\Delta/2)-(-\Delta/2)} = \frac{1}{\Delta} \tag{3.6}$$

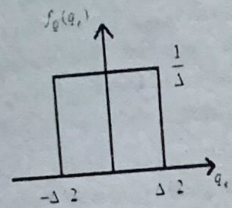


Fig.3.16: uniform PDF of quantization noise

The average quantization noise power P_q is then can be expressed as

$$P_q = \overline{q_e^2} = \int_{-\infty}^{\infty} q_e^2 f_Q(q_e)\, dq_e = \frac{1}{\Delta}\int_{\Delta/2}^{-\Delta/2} q_e^2\, dq_e = \frac{1}{\Delta}\left[\frac{q_e^3}{3}\right]_{-\Delta/2}^{\Delta/2} = \frac{\Delta^2}{12} \tag{3.7}$$

It is seen from the above Eq.(3.7) that Q-error in uniform quantization is dependent only upon the step size.

Suppose the values of the continuous signal swings between $+x_m$ and $-x_m$. Suppose the number of levels used is N. So the quantization step size can be obtained

$$\Delta = \frac{x_m - (-x_m)}{N}$$

$$\Delta = \frac{2X_m}{N} \tag{3.8}$$

$$\Delta = \frac{2x_m}{2^n} = \frac{x_m}{2_{n-1}} \tag{3.9}$$

Where $N(=2^n)$ is the total number of representation levels. Substituting the value of Δ in Eq.(3.7). we get,

The average noise power is

$$P_q = \frac{\Delta^2}{12} = \left(\frac{x_m}{2^{n-1}}\right)^2 \frac{1}{12}$$

$$P_q = \frac{x_m^2}{3 \times 4^n} \tag{3.10}$$

Assuming that the average signal power is $\overline{x^2}$, the signal to quantization noise ratio will be

$$SQNR = \frac{\overline{x^2}}{P_q} = \frac{\overline{x^2}}{\frac{x_m^2}{3\times 4^n}} = \frac{\overline{x^2}}{x_m^2} \tag{3.11}$$

The ratio $\frac{\overline{x^2}}{x_m^2}$ can be replaced by normalized signal power.

$$\nabla = \frac{\overline{x^2}}{x_m^2} \tag{3.12}$$

Then the SQNR will be

$$SQNR = \nabla \times 3 \times 4^n \tag{3.13}$$

In terms of dB, the SQNR will be

$$SQNR_{(dB)} = P_{(dB)} + 10\log 3 + 10n\log 4 \tag{3.14}$$

$$SQNR_{(dB)} = P_{(dB)} + 4.8 + 6n \tag{3.15}$$

It means that for each extra bit (n) used for representing each quantization level, the SQNR increases by 6dB (i.e., the noise power is reduced 4 times).

Since x_m is the maximum level of input signal, the normalized signal power is always less or equal to unity, therefore the upper limit of SQNR will be equal to

$$SQNR = 3 \times 4^n \tag{3.16}$$

In terms of quantization level N, this ratio of Eq.(3.14) can be expressed as

$$SQNR_{(dB)} = P_{(dB)} + 10\log 3 + 10\log 4^n \tag{3.17}$$

$$SQNR_{(dB)} = P_{(dB)} + 10\log 3 + 10\log 2^{2n} \tag{3.18}$$

$$SQNR_{(dB)} = P_{(dB)} + 4.8 + 10\log N^2 \tag{3.19}$$

$$SQNR_{(dB)} = P_{(dB)} + 4.8 + 20\log N \tag{3.20}$$

Example.3.1: ***Derive an expression for SQNR for a PCM system using uniform quantization technique. Given that input to the PCM system is a sinusoidal signal.***

Solution:

Assuming a sinusoidal signal

$$x(t) = x_m \sin wt$$

Thus we know that the average power of this signal is

$$\overline{x^2} = \frac{x_m^2}{2} \tag{3.21}$$

Substituting this value in Eq.(3.11) we get

$$SQNR = \frac{\overline{x^2}}{P_q} = \frac{\frac{x_m^2}{2}}{\frac{x_m^2}{3\times 4^n}} = \frac{3\times 4^n}{2} = 1.5 \times 4^n \tag{3.22}$$

In terms of dB, the SQNR will be

$$SQNR_{(dB)} = 10\log 1.5 + 10n\log 4 \tag{3.23}$$

$$SQNR_{(dB)} = 1.8 + 6n \text{ (for sinusoidal signal)} \tag{3.24}$$

Signaling Rate and Bandwidth consideration for PCM

Let us assume that a quantizer uses n number of binary digits to represent each level. Then the number of levels that may be presented by n digits, will be, $N = 2^n$, and suppose that the spectrum of the low pass continuous time message signal is bandlimited to f_m.

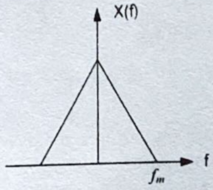


Fig.3.17: Spectrum of message signal

Thus number of bits per samples is n and the number of samples per second is $f_s (f_s \geq 2f_m)$.

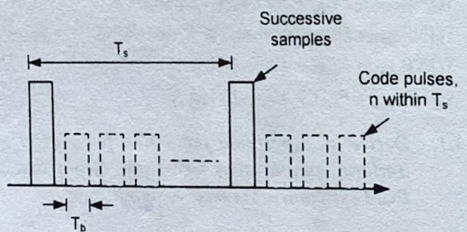


Fig.3.18: "n" number of binary digits to represent each level.

Signaling rate (r) or Baud rate is the number of symbols transmitted per second. For binary waveform signaling rate is expressed in bits per second and represented as

$$r = \frac{\text{Bits}}{\text{Sample}} \times \frac{\text{Sample}}{\text{Second}}$$

$$r = nf_s \tag{3.25}$$

Let us consider a band-limited signal with upper maximum frequency f_m as shown in Fig.3.17. We know from Nyquist creteria that $f_s \geq 2f_m$. It is seen from Nyquist's criteria that for multi-level signaling for a noiseless channel is

$$r = 2 \times B \times \log M \tag{3.26}$$

It is shown that the absolute minimum channel bandwidth required to transmit "r" bits of the signals per second is

$$\text{PCM Band width (B)} \geq \frac{\text{Signaling rate (r)}}{2}$$

$$B \geq \frac{r}{2}$$

$$B \geq \frac{1}{2} \cdot nf_s \tag{3.27}$$

$$B \geq \frac{1}{2} \cdot n. (2f_m)$$

Therefore, the minimum bandwidth required is

$$B \geq nf_m \tag{3.28}$$

In practice, the bandwidth of the channel is assumed to be higher than the absolute minimum bandwidth and therefore is equal to

$$B \geq (1 + \rho)\, nf_m \tag{3.29}$$

Where ρ is called the roll-off factor (the value lying between 0 and 1).

Now, for a telephone voice channel with highest frequency 3400 Hz, the bandwidth, considering the guard bands is standardized at 4 Hz. For 8 bit coding of the speech and with $\rho = 1$.

$$B = (1 + \rho) nf_m$$

$$B = (\text{voice}) = (1 + 1) \times 8 \times 4 = 64\ Hz$$

3.2.5 Time Division Multiplexing (TDM)

In PAM, the samples are transmitted periodically. Periodic transmission results in the time interval between successive samples, equal to the period duration. Hence, period duration between adjacent pulses of the PAM wave can be used by other independent message signals on a time-shared basis. By doing so, we obtain a time-division multiplex system (TDM), which enables the joint utilization of a common channel by a number of independent message signals without interference.

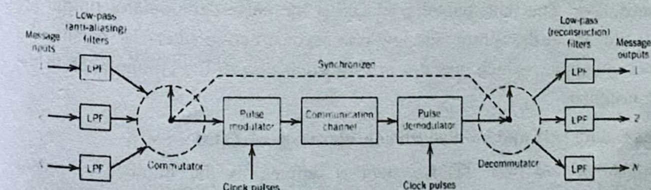


Fig.3.19: Block diagram of TDM system

The concept of TDM is illustrated by the block diagram shown in Fig. 3.19. Initially, each input message signal is band-limited by a low-pass pre-alias or anti-aliasing filter to remove the frequencies that are considered nonessential for transmission. The pre-alias filter outputs are then applied to a commutator, which is usually implemented using electronic switching circuitry. The function of the commutator is

- To take a narrow sample of each of the N input message at the rate f_s that is slightly higher than $2f_m$, where f_m is the cutoff frequency of pre-alias filter, and
- To sequentially interleave these N samples inside a sampling interval of $T_s = 1/f_s$.

Following the commutation process (the commutation itself results in the time division multiplexed PAM signal), the multiplexed signal is applied to a pulse-amplitude modulator, the purpose of which is to transform the multiplexed signal into a form suitable for transmission over the communication channel.

Suppose that the N message signals to be multiplexed together. Then the sampling rate for each message signal is determined in accordance with the sampling theorem. Let T_s denote the sampling period of each message signal and T_x denote the time spacing between adjacent samples in the time-multiplexed signal. It is obvious that

$$T_x = T_s/N \qquad (3.30)$$

Hence, the use of time-division multiplexing introduces a bandwidth expansion by factor N, because the scheme must squeeze N samples derived from N independent message signals into a time slot equal to one sampling interval T_s.

At the receiver, the received signal is applied to a pulse-amplitude demodulator, which performs the reverse operation of the pulse-amplitude modulator. The short pulses produced at the pulse demodulator output are distributed to the appropriate low-pass reconstruction filters by means of a decommutator, which operates in synchronism with the commutator in the transmitter.

Signaling rate and bandwidth consideration of TDM

Signaling rate of a TDM system is defined as the number of pulses transmitted per second and expressed as *R*. To determine the signaling rate let us consider that N samples of N independent message signals are

transmitted in T_s duration using TDM system. This combination of N samples is known as frame. Fig.3.20 shows the example of TDM for N=2.

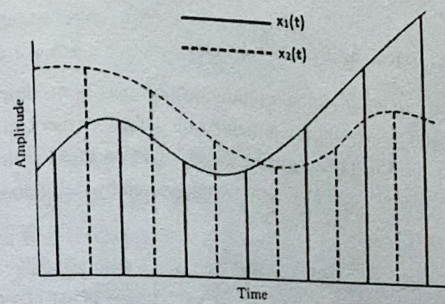


Fig.3.20: TDM of two signals

We know that in order to transmit N samples within T_s duration, the duration of sample to samples spacing must be as given by Eq.(3.30) above.i.e.,

T_x(sample to sample duration)$=T_s/N$

We know that $T_s=1/f_s$

Thus,

$$T_x = 1/Nf_s \qquad (3.31)$$

Now, number of samples transmitted per second $(f_x)=1/T_x=Nf_s$

Therefore, signaling rate of a TDM system is

$$R = Nf_s \qquad (3.32)$$

But in order to satisfy Nyquist criteria $f_s \geq 2f_m$

Thus the signaling rate of a TDM system is given by

$$R \geq 2Nf_m \qquad (3.33)$$

The minimum transmission bandwidth of a TDM channel is given by

$$\text{Bandwidth(B)} = \frac{1}{2} \times \text{Signaling Rate(R)}$$

Therefore, transmission bandwidth is given by

$$B \geq \frac{1}{2} \times 2Nf_m = Nf_m \qquad (3.34)$$

Hence, the minimum bandwidth is

$$B = Nf_m \qquad (3.35)$$

Multiplexing hierarchy for digital communication

There are two different time division multiplexing patterns used for digital communication.

1. **North American or T carrier system**

The breakthrough in digital communication was in the form of TDM-PCM telephone system implemented by AT&T (American Telephone & Telegraph) Co. The original concept of TDM-PCM multiplexing of 24 channel known as T1 system waspioneered by Bell Laboratories.

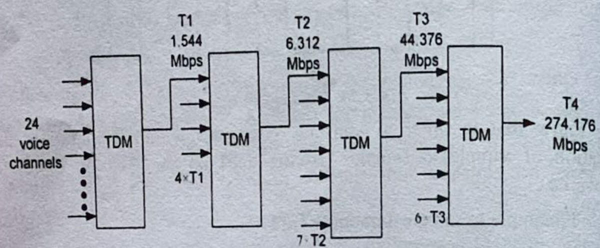


Fig.3.21: Block representation of North American or T carrier system

T1 carrier system

In T1 (the first level of T multiplexing) system 24 voice channels are multiplexed to obtain a T1 signal. Each voice channels are sampled at 8 kHz. This rate is obtained by band limiting the individual voice signal to 3400 Hz. An extra 600 Hz bandwidth is separated as a guard band resulting in maximum frequency f_m equal to 4000 Hz or 4 kHz. Now sampling at the Nyquist rate, the sampling frequency is

$$f_s = 2f_m = 2 \times 4kHz = 8kHz$$

The time duration between successive samples of the first message (frame duaration) is therefore

$$\text{Frame duration } (T_s) = \frac{1}{f_s} = \frac{1}{8kHz} = 125\mu s$$

Each sample is quantized and coded into 7 bit PCM code word. The eight bit is added for synchronization purpose. Thus the total number of bits per sample is 8.

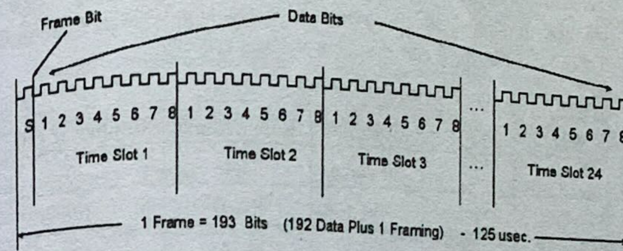


Fig.3.22: Frame arrangement for T1 carrier system

Number of bits per frame is 192(i.e., 24x8). Additional bit is added as frame synchronization bit. Therefore a total of 193 bits need to be accommodated within the period of a frame i.e. 125 μs.

Hence,

$$\text{Transmission Rate} = \frac{\text{Number of bits per Frame}}{\text{Frame duration}} = \frac{193}{125\times10^{-6}} = 1.544\,Mbps$$

In addition to the voice signals, special supervisory signals are also to be sent to the receiving end. These are needed to transmit dial pulses and telephone off-hook/on-hook signals.

The other levels of the T multiplexing hierarchy are T2, T3, T4 and T5. The corresponding voice channels and the data rates are shown in Fig.3.21 above.

Table.3.1: T carrier system hierarchy.

S.No.	Line	Multiplex	Rate (Mbps)	Number of voice
1	T1	24 voice	1.544	24
2	T2	4×T1	6.312	96
3	T3	7×T2	44.736	672
4	T4	6×T3	274.176	4032

2. **European or E carrier system**

In digital telecommunications, where a single physical wire pair can be used to carry many simultaneous voice conversations by time-division multiplexing, worldwide standards have been created and deployed. The European Conference of Postal and Telecommunications Administrations (CEPT) originally standardized the E-carrier system, which revised and improved form of the earlier American T-carrier technology, and this has now been adopted by the International Telecommunication Union, Telecommunication Standardization Sector (ITU-T). It was widely adopted in almost all countries outside the US, Canada, and Japan. Table.3.2 shows the E carrier system and their capacity.

Table.3.2. E carrier system hierarchy

S.No.	Line	Multiplex	Rate (Mbps)	Number of voice
1	E1	30 voice	2.048	30
2	E2	4×E1	8.448	120
3	E3	4×E2	34.368	480
4	E4	4×E3	139.264	1920

3.2.6 Differential Pulse Code Modulation (DPCM)

The digital transmission of the signals result in reception of very high quality signals at the receiving end, as the effect of all other types of noise can be totally eliminated at the receiver side. The only type of noise that can not be avoided is the quantization noise. The quantization or Q-noise is inherent to the digital system in which the continuous time analog signals are sampled and quantized. However the signal to noise ratio (SQNR) resulting from the Q-noise is very high and roughly equal to "6n" dB as seen from Eq.(3.24). The price paid for or the trade-off made is the very high channel bandwidth requirements to transmit digital data. For example a voice channel with the highest frequency of 4 kHz (f_m) would occupy the channel bandwidth of 64 kHz ($2nf_m$) when converted into a 8 (i.e., n) bit PCM signal. But the signal quality received would be almost 48 dB ($\approx$ 6n . dB).

For normal voice telephony or voice communication the required channel bandwidth of 64 kHz is a huge wastage, whereas the quality is more than required. Therefore there is a need for reducing the channel bandwidth without substantially compromising the quality. One way achieve this is to play with the common single parameter that defines the channel bandwidth and the SQNR i.e., "n", the number of bits to represent each quantized levels. Now if the value of "n" is reduced, the required channel bandwidth will reduce proportionally (i.e., $B = 2nf_m$) but the SQNR will also decrease by 6 dB per " n" decrease. Ultimately when n=1, the channel bandwidth required for a voice channel would be 8 kHz (at par with DSB-FC channel bandwidth) and the SQNR only 6 dB. This value of SQNR may not be acceptable for voice communications. Therefore other techniques of quantization and encoding are explored where the channel bandwidth (or the effective data rates) is reduced without substantially compromising the quality (SQNR). These techniques are DPCM and DM.

When the signal is sampled at a rate higher than Nyquist rate, the resulting sampled signal is found to exhibit a high correlation between adjacent

samples as the samples come closer to each other. This property of the bandlimited signal is used to arrive at DPCM and DM techniques.

Due to high correlation between the samples at higher Nyquist rate, the signal doesn't change rapidly from one sample to the next. Thus, if we encode these signal output directly, as in PCM, the resulting encoded signal contains redundant information (the repeated information about the signal in the successive digital output). However at higher Nyquist rate, the performance of quantizers can be significantly improved by quantizing a signal that is the difference between the present sample and its prediction derived from the previous samples instead of quantizing the original signal.

Thus, instead of quantizing the whole sample value, DPCM quantizes the difference only, thus reducing the required quantization levels to minimum and thereby reducing the data rate or channel bandwidth.

3.2.7 DPCM transmitter

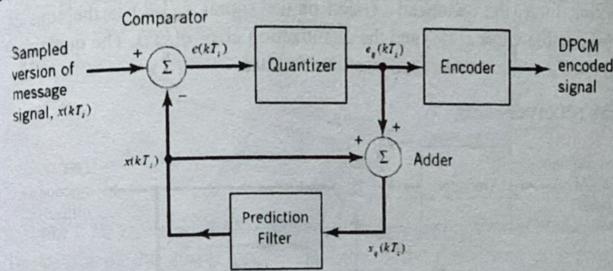


Fig.3.23: DPCM Transmitter

Fig.3.23 shows the transmitter of DPCM system. Let, $x(kT_s)$ be the sampled sequence of the message signal $x(t)$ sampled at T_s intervals. The sequence of pulses $\hat{x}(kT_s)$ is the prediction of $x(kT_s)$, derived from the previous $x\{(k-1)T_s\}$ sample. The comparator evaluates the difference between $x(kT_s)$ and $\hat{x}(kT_s)$ to determine what is known as prediction error $e(kT_s)$.

$$e(kT_s) = x(kT_s) - \hat{x}(kT_s) \tag{3.36}$$

This error signal is quantized to produce quantized version of the error signal $e_q(kT_s)$. Thus the quantized output can be represented as

$$e_q(kT_s) = e(kT_s) + q(kT_s) \tag{3.37}$$

Where, $q(kT_s)$ is the quantization error. The quantized output signal $e_q(kT_s)$ and previous predicted signal is added to generate an input to the prediction filter and thus generates the current predicted output which is more and more closer to the actual sampled signal after every cycle. We can see that the quantized error signal $e_q(kT_s)$ is very small and can be encoded by using small number of bits. Thus, the number of bits per sample and subsequently the required channel bandwidth is reduced in DPCM.

Now the prediction filter input $x_q(kT_s)$ is obtained by

$$x_q(kT_s) = \hat{x}(kT_s) + e_q(kT_s) \tag{3.38}$$

Substituting the value of $e_q(nT_s)$ from Eq.(3.37) in Eq.(3.38) ,we get,

$$x_q(kT_s) = \hat{x}(kT_s) + e(kT_s) + q(kT_s) \tag{3.39}$$

From Eq.(3.36) we can write

$$e(kT_s) + \hat{x}(kT_s) = x(kT_s) \tag{3.40}$$

Now substituting this value in Eq.(3.39) we get

$$x_q(kT_s) = x(kT_s) + q(kT_s) \tag{3.41}$$

From Eq.(3.41) it can be concluded that irrespective of the properties of prediction filter, the quantized version of the signal $x_q(kT_s)$ is the sum of original sample value $x(kT_s)$ and the quantization error $q(kT_s)$. The quantized signal $e_q(kT_s)$ is now transmitted over the channel.

DPCM receiver

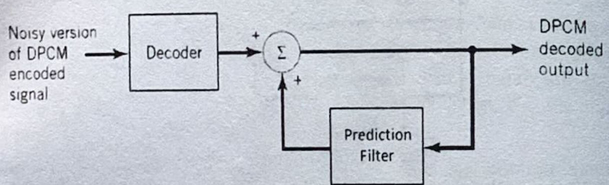


Fig.3.24: DPCM receiver

The receiver shown in Fig.3.24 is identical to the shaded portion of the transmitter. The inputs in both cases are also the same (i.e., $e_q(kT_s)$). Therefore, the predictor output must be $\hat{x}(kT_s)$ (the same as the predictor output at the transmitter). Hence, the receiver output (which is the predictor input) is also the same

$$x_q(kT_s) = x(kT_s) + q(kT_s) \tag{3.42}$$

Thus we are able to receive the desired signal $x(kT_s)$ plus the quantization noise $q(kT_s)$. The received samples $x_q(kT_s)$ are decoded and passed through a lowpass filter for digital to analog conversion.

As the correlation between successive samples are high, less number of levels are required to encode. Thus, the required bandwidth is reduced. However, the only disadvantage is in the complexity of implementation of prediction filter.

Advantage of DPCM

1. Less number of quantization levels N (or the number of bits "n") is required to quantize the small value of prediction error compared to large sample itself. The reduction in N (or n) will not severely affect the SQNR because the quantization noise power is dependent upon step size only and in both cases the step sizes are maintained at the same level.
2. The design of system is simple. At the output there is no need of digital to analog conversion as in the case of PCM.

Prediction filter

The prediction filter in general is a tapped delay line filter type, where the predicted value $\hat{x}(kT_s)$ is modeled as linear combination of past values of quantized inputs.

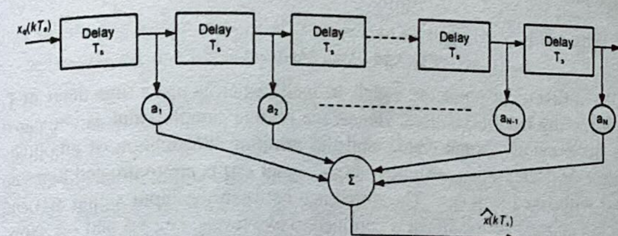


Fig.3.25: Transversal filter (tapped delay line) used as a linear predictor.

The output of the above prediction filter is the linear sum of n previous values of samples scaled by some coefficients.

$$\hat{x}(kT_s) = \sum_{m=1}^{N} a_k\, x_q\{(k-m)T_s\} \tag{3.43}$$

Where, N is the order of prediction filter.

Delta Modulation (DM)

In DPCM the number of bits to be transmitted is considerably reduced in comparison to PCM. But at the cost of increased transmitter and receiver complexity compared to the PCM. This is basically due to the need of prediction filter. Thus, maintaining the tradeoff between complexity-bandwidth, the complexity of DPCM can be relaxed by use of new modulation technique known as Delta Modulation (DM). DM is less complex than DPCM but at the cost of increased bandwidth (under equal SQNR conditions) compared to DPCM.

In DM, the incoming message signal $x(t)$ is oversampled at rate higher than the Nyquist rate. Thus, the sample-to-sample amplitude difference is considerably reduced or in other words the correlation between corresponding samples is greatly increased. So, it may be possible for two level quantization (i.e., 1 bit representation) of the difference signal. The Delta Modulation (DM) is based on this principle. Delta modulation is also viewed as a 1-bit version of DPCM scheme.

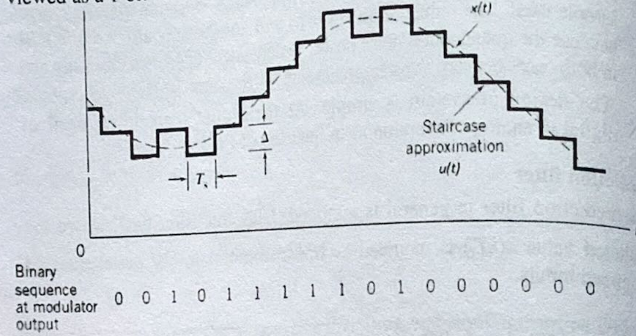


Fig.3.26: Delta Modulation

In DM, a first order predictor, which, as seen earlier, is just a time delay of T_s (the sampling interval).is used. Hence, the present sample value is compared with the previous sample value, and the resulting information of amplitude increase or decrease is transmitted. Input signal x(t) is approximated to a step signal with fixed step size. The difference between the input signal x(t) and staircase approximation signal is confined to two levels, i.e., +Δ and - Δ. Now, if the difference is positive, then approximated signal is increased by one step, i.e. Δ and if the difference is negative, then the approximated signal is reduced by Δ. The encoding is done such that, when the step size is reduced, '0' is transmitted and if the step size is increased, '1' is transmitted. Hence, for each sample, only one bit is transmitted. Fig.3.26 shows the analog signal x(t) and its staircase approximated signal by delta modulation.

DM transmitter

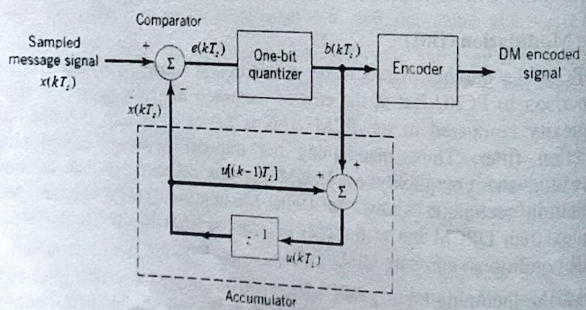


Fig. 3.27: DM transmitter

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

$$e(kT_s) = x(kT_s) - \hat{x}(kT_s) \tag{3.44}$$

where, $e(kT_s)$ is the error at present sample. $x(kT_s)$ is the present sampled signal of x(t). $\hat{x}(kT_s)$ is the last sample approximation of the staircase waveform.

If the present value of staircase output is assumed to be $u(kT_s)$, then, $u[(k-1)T_s] = \hat{x}(kT_s)$

Now let us define a term $b(kT_s)$ as an output of one bit quantizer, which produces only +Δ and –Δ. Thus,

$$b(kT_s) = \Delta\, Sgn[e(kT_s)] \tag{3.45}$$

Thus, depending on the sign of $e(kT_s)$ the sign of step size Δ is determined. In other words, we can write

$$b(kT_s) = \begin{cases} +\Delta \text{ if } x(kT_s) \geq \hat{x}(kT_s) & \text{[Transmit binary '1']} \\ -\Delta \text{ if } x(kT_s) < \hat{x}(kT_s) & \text{[transmit binary '0']} \end{cases}$$

Fig.3.27 shows the transmitter of DM. The summer in the accumulator adds quantizer output (±Δ) with the previous sample approximation. This gives present sample approximation. i.e.,

$$u(kT_s) = u(kT_s - T_s) + b(kT_s) \tag{3.46}$$

$$u(kT_s) = u[(k-1)T_s] + [\pm\Delta] \tag{3.47}$$

The previous sample approximation $u[(k-1)T_s]$ is restored by delaying one sample period T_s. The error signal $e(kT_s)$ is obtained by substration of the samples input signal $x(kT_s)$ and staircase approximated signal $\hat{x}(kT_s)$.

Thus, depending on the sign of $e(kT_s)$, one bit quantizer generates an output of +Δ and -Δ . If the step size is +Δ, then binary '1' is transmitted and if it is -Δ, then binary '0' is transmitted.

DM receiver

The receiver of DM consists of an accumulator followed by a low-pass filter as shown in Fig.3.28. The accumulator generates the staircase approximation signal output and is delayed by one sample period T_s. It is then add to the current input signal. If the received binary is '1', then it adds Δ to the delayed signal and if the received signal is binary '0' it subtracts Δ to the delayed signal. The out off band quantization noise in the high-frequency staircase waveform is rejected by passing it through a low-pass filter with a bandwidth equal to the original signal bandwidth.

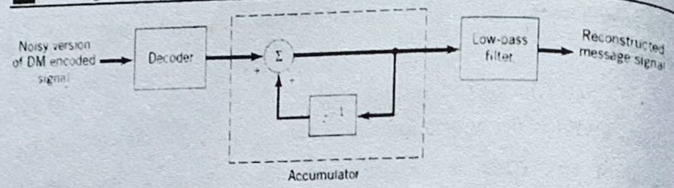


Fig. 3.28: DM receiver

Drawbacks of DM

In the delta modulation, the two parameters plays an important role as we have discussed: the step size and the sampling frequency. These parameter must be chosen very cleverly otherwise we may have two major sources of distortion in DM systems.

(i) **Slope over load distortion**

If the rate of change of quantization is slow in comparison to the slope (rate of change) of message signal (i.e., the step size may not be sufficient to follow rate of change of the signal), it is called slope overload distortion. To evaluate the condition, at which the slope overload can be eliminated, let us consider an example given below.

Example 3.2 ***Consider a sinusoidal signal of A_m amplitude and f_m frequency applied to a delta modulator with representation level $\pm\Delta$. Show that in order to avoid slope overload distortion, it is necessary that***

$$A_m < \frac{\Delta}{2\pi f_m T_s}$$

Where Ts = sampling period

Solution:

Let us consider a sine wave represented as,

$$x(t) = A_m \sin(2\pi f_m t)$$

and the maximum slope of the signal is

$$\text{Maximum slop} = \left|\frac{dx(t)}{d(t)}\right|_{\max}$$

Now, the slope of delta modulation is be given as

$$\text{Maximum slope} = \frac{\text{Step size}}{\text{Sample period}} = \frac{\Delta}{T_s}$$

For given signal, slope overload can be avoided if the slope of the signal is less than the slope of delta modulation. Thus,

$$\left|\frac{dx(t)}{dt}\right|_{\max} < \frac{\Delta}{T_s}$$

$$|2\pi f_m \cdot A_m \cdot \cos(2\pi f_m t)|_{\max} < \frac{\Delta}{T_s}$$

$$2\pi f_m \cdot A_m < \frac{\Delta}{T_s}$$

$$A_m < \frac{\Delta}{2\pi f_m T_s}$$

Thus, to avoid slope overloading the amplitude A_m of the given sine wave must be less than the term $\frac{\Delta}{2\pi f_m T_s}$.

(ii) **Granular noise**

When the slope of the signal is low (i.e., signal is almost constant with respect to time), and the step size Δ is relatively high, the approximation starts swinging from –Δ to + Δ. causing high noise level. This type of noise is known as granular noise and can be minimized by reducing the step size Δ.

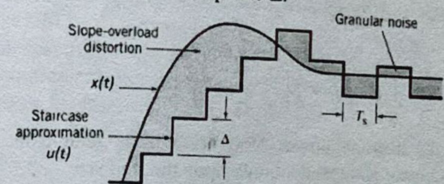


Fig.3.29: Slope overload distortion and granular noise of DM.

Therefore, a large step size is required to reduce slope overload distortion and small step size is required to reduce granular noise. Adaptive DM is the modification to overcome these errors.

3.2.8 Signal to quantization noise ratio in DM

We know that the condition to avoid the slope overload distortion is expressed as

$$A_m < \frac{\Delta}{2\pi f_m T_s} = \frac{\Delta}{2\pi}\left(\frac{f_s}{f_m}\right) \quad (3.48)$$

The maximum output power considering a sinusoidal signal with amplitude A_m is

$$P_{max} = \frac{A_m^2}{2}$$

Hence,

$$P_{max} = \frac{A_m^2}{2} = \left(\frac{\Delta f_s}{2\pi f_m}\right)^2 \times \frac{1}{2}$$

$$P_{max} = \frac{\Delta^2}{8\pi^2}\left(\frac{f_s^2}{f_m^2}\right) \quad (3.49)$$

In DM the maximum Q-error (q_e) is $\pm\Delta$, where Δ is the step size (i.e., $-\Delta \leq$ Q – error $\leq \Delta$ with equal probability). This gives a uniform PDF given by

$$f_Q(q_e) = \frac{1}{b-a} = \frac{1}{(\Delta)-(-\Delta)} = \frac{1}{2\Delta} \tag{3.50}$$

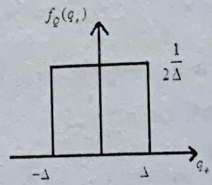


Fig.3.30: uniform PDF of error in DM

For uniform distribution, mean=0 and Mean square value of Q-noise is equal to variance

$$P_q = \overline{q_e^2} = \int_{-\infty}^{\infty} q_e^2 f_Q(q_e)\, dq_e$$

$$P_q = \frac{1}{2\Delta}\int_{-\Delta}^{\Delta} q_e^2\, dq_e = \frac{1}{\Delta}\left[\frac{q_e^3}{3}\right]_{-\Delta}^{\Delta} = \frac{\Delta^2}{3} \tag{3.51}$$

Where, P_q is the average power of Q-noise. It is seen from the above Eq.(3.51) that Q-error is dependent only upon the step size.

At the output of DM receiver the signal is passed through a reconstruction low-pass filter (LPF). We assume the bandwidth of LPF is such that

$f_{LPF} \geq f_m$ and $f_{LPF} << f_s$

Now assuming that the qantization noise power P_q is distributed uniformly over the frequency band upto f_s. The PSD of P_q is

$$G_q(f) = \frac{P_q}{2f_s} = \text{for } |f| \leq f_s \tag{3.52}$$

$$G_q(f) = \frac{\Delta^2}{3 \times 2f_s} \tag{3.53}$$

Thus, the output quantization noise power within the bandwidth f_{LPF} is given by

$$P_q' = \int_{f_{LPF}}^{-f_{LPF}} G_q(f)\, df \tag{3.54}$$

$$P_q' = \int_{f_{LPF}}^{-f_{LPF}} \frac{\Delta^2}{3 \times 2f_s}\, df \tag{3.55}$$

$$P_q' = \int_{f_{LPF}}^{-f_{LPF}} \frac{\Delta^2}{3 \times 2f_s}\, df = \frac{\Delta^2 f_{LPF}}{3f_s} \tag{3.56}$$

Thus the expression for output signal to quantization noise ratio is

$$SQNR_{DM} = \frac{P_{max}}{P_q'} = \frac{\Delta^2}{8\pi^2}\left(\frac{f_s^2}{f_m^2}\right) \times \frac{3f_s}{\Delta^2 f_{LPF}} \tag{3.57}$$

$$SQNR_{DM} = \frac{3f_s^3}{8\pi^2 f_m^2 f_{LPF}} \tag{3.58}$$

For, the ideal case when $f_{LPF} = f_m$, Eq.(3.58) becomes

$$SQNR_{DM} = \frac{3}{8\pi^2}\left(\frac{f_s}{f_m}\right)^3 \tag{3.59}$$

Delta modulation does not require analog to digital conversion

Comparison of PCM and DM

To compare the two modulation schemes, there should be some identical conditions. Let us assume that both systems use approximately the same bandwidth for transmitting a base-band analog signal.

Let f_s (normally the Nyquist rate) denote the sampling rate of an n-bit PCM and f_s' denotes the sampling rate of n-bit DM. Then the bit rate of PCM is nf_s and rate of DM is f_s' (as n=1 for DM).

If the signal spectrum extends up to f_m Hz, then $f_s = 2f_m$ and identical bandwidth requirements implies that

Bandwidth of PCM=Bandwidth of DM

Signaling rate of PCM (R_{PCM})=Signaling rate of DM (R_{DM})

$$f_s' = nf_s$$
$$f_s' = 2nf_m$$

Two modulation techniques are compared based on the SQNR.

From the above equations for a fixed bandwidth the performance of DM is always poorer than PCM.

For 8-bit PCM and DM, we have

$(SQNR)_{PCM} \approx 48$ dB and $(SQNR)_{DM} \approx 22$ dB

It shows that under similar conditions the performance of PCM is superior to DM. With ordinary DM, a comparable quality with respect to PCM can be achieved only when the signal rate is around 100kbits/s. However, the hardware requirement for DM is much more simpler than PCM. A single chip called continuous variable slope DM coder-decoder (CODECS) can perform both transmission and reception processing, whereas for PCM two separate chips, one for analog to digital conversion and another for digital to analog converters are required.

3.2.9 Adaptive Delta Modulation

As we have discussed above there are two types of problems called slope overload and granular noise.

So the performance of a delta modulation can be improved significantly by making the step size of the modulator assume a time-varying form. In particular, when input signal is varying fast (resulting in slope overload distortion), the step size must be increased. When signal is varying low (resulting in granular noise), the step size must be reduced. In this way, the step size is adapted to the level of the input signal and the resulting method is called adaptive delta modulation (ADM).

ADM Transmitter

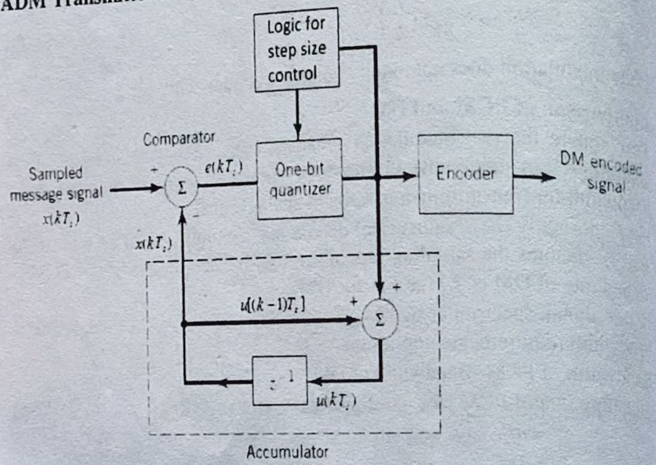


Fig.3.31: ADM transmitter

Fig.3.31 shows the transmitter part of the adaptive delta modulation (ADM). The logic for step size control is added in the diagram. The step size increases or decreases according to a specified rule depending on one bit quantizer output. For example, is one bit quantizer output is high (i.e.,1), then the step size may be doubled for next sample. If one bit quantizer output is low (i.e.,0) then step size may be reduced by 1 step.

ADM Receiver

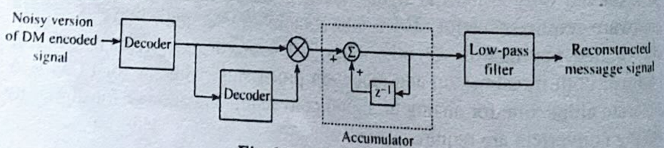


Fig.3.32: ADM receiver

The receiver of ADM is shown in Fig.3.32, contains two distinct portions. The first portion produces the step size from the incoming bit. The output is then applied to an accumulator which builds the staircase waveform. The low-pass filter then smoothens out the staircase waveform to reconstruct the original signal.

3.3 Linear Prediction Theory

Linear prediction is an important source coding technique for the digitization of speech signals. Unlike waveform coding, linear prediction uses linear predictive vocoder in order to analyze the parameters of a speech signals according to a physical model for speech production process. The term vocoder refers to the device that performs voice coding and hence derived from it.

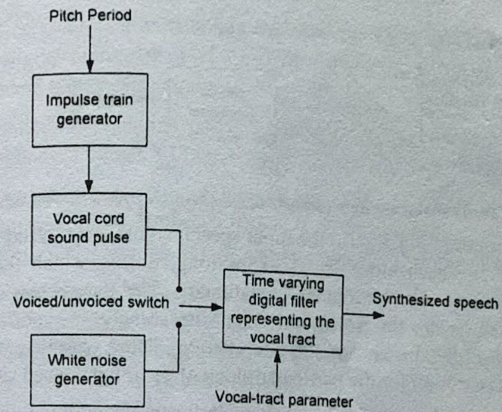


Fig.3.33: Speech production model

General model of speech production is shown in Fig.3.33. The model assumes that the sound generating mechanism (i.e. source of excitation) is linearly separable from the intelligence modulation mechanism (i.e., vocal tract filter). The excitation of vocal tract filter depends on whether the speech sound is voiced or unvoiced.

Voice sounds are produced when the air from the lungs passes through vocal cord creating a quasi periodic pulses of air that excite the vocal tract. Unvoiced sounds are produced by constriction at some point in the vocal tract (usually towards the mouth end), and forcing air through the constriction at a high enough velocity to produce turbulence.

The model of speech generation or voiced sound consists of the vocal tract filter excited with a periodic sequence of impulses (very short pulses) spaced by a fundamental period equal to the pitch period. The unvoiced sound is modeled by exciting the vocal tract filter with a white noise sequence. The vocal tract filter is time varying, so that its coefficients can provide an adequate representation for the input segment of voiced or unvoiced sound.

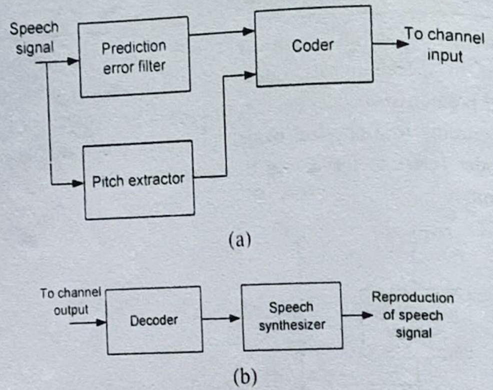


Fig.3.34: Transmitter and receiver block of linear predictive vocoder.

At the transmitter in Fig.3.34(a) the input speech signal is analyzed block by block. Typically, each block is 10-30 ms long, during which the speech generation process may be treated as stationary. The parameters extracted from the analysis are the prediction error filter (analyzer) coefficients, a voiced/unvoiced parameter, and the pitch period. These parameters provide a complete description for the particular block of the input speech signal.

The receiver, shown in Fig.3.34(b), first performs decoding, followed by synthesis of the speech signal. The standard result of this analysis/synthesis is an artificial sounding reproduction of the original speech signal.

Previous Exam Questions

1. Derive the expression for evaluating SQNR for uniform quantization in terms of number of quantization levels and number of bits per source sample in PCM.
2. Derive the expression for evaluation of the signal to quantization noise ratio(SQNR) for Delta Modulation. Assume minimum slope overload distortion..
3. Consider a sine wave frequency f_m and amplitude A_m applied to a delta modulator with level $\pm\Delta$. Show that slope overload distortion will occurs if $A_m > \Delta/2\pi f_m T_s$ where: T_s Sampling period.
4. Explain the operation of Differential PCM along with its derivations and diagram. Draw the staircase approximation diagram using delta modulation for the data 1111000010101010.
5. What do you mean by Pulse Code modulation? Explain the functional block diagram of the PCM system. Find the signaling rate of the T1 System and Draw its frame diagram.

6. Differentiate between uniform quantization and non-uniform Quantization.
7. Why non-uniform quantization is preferred over uniform quantization? Explain any method of non-uniform Quantization.
8. What do you mean by companding. Why is it necessary? Explain different types of campanding methods.
9. Compare PCM and Differential PCM.
10. Define Quantization. Explain the basic principle of TDM. Explain the E1 digital hierarchy of TDM-PCM Telephony. Estimate the signaling rate and bandwidth requirements 16 Channel TDM-PCM telephony.
11. Estimate the signaling rate and bandwidth requirements in TDM-PCM telephony. Discuss T1 hierarchy of TDM-PCM Telephony.
12. A speech signal with maximum frequency of 4KHz and maximum amplitude of ±1.1V is applied to a PCM system with its bit rate of 32Kbps.Calculate the SQNR and number of bits per sample.
13. A signal having the dynamic range of ±5V is to be uniformly quantized to 128 representation levels. Estimate the required step size, the power of quantization noise produced and maximum signal to quantization noise ratio that can be achieved.
14. A linear delta Modulator is designed to operate on speech signals to 3.4KHz .The specifications of the modulator are as follows.
15. Sampling rate=10.f_N where f_N is the Nyquist Rate of the speech signal.
    a. Step Size Δ= 100m V.
16. The modulator is tested with a 1kHz sinusoidal signal. Determine the maximum amplitude this signal required to avoid slope overload.
17. A video signal having bandwidth of 6MHz is to be transmitted using binary pulse code modulation. Assuming the number of uniform quantization levels to be equal to 256. Estimate (a) Code word length (b) Absolute minimum and practical bandwidth of the signal (c) Final data rate in kbps (d) Output SQNR.
18. What are the signaling (bit) rate and bandwidth requirement for the T1 and E1 digital carrier system. Explain briefly about Differential Pulse Code Modulation (DPCM) encoder.
19. An audio signal of frequency 4 kHz amd maximum dynamic range of ±2.4V is digitaized by PCM system with its bit rate of 64 kHz. Calculate number of bits per sample, quantization noise power and $SQNR_{dB}$. Estimate the minimum bandwidth required for TDM of 10 such audio signals (assume no extra framing and synchronization bits).

20. A delta modulator is used to encode speech signal band-limited to 3kHz with sampling frequency 10 kHz. For maximum signal. For maximum signal amplitude of $A_{max}=1$, find

    i. Minimum step size to avoid slope overloading.

    ii. Assuming the speech signal to be sinusoidal, find signal to quantization noise ratio

    iii. Determine the minimum transmission bandwidth.

21. A message signal $x(t) = 6\cos(5000\pi t)$ is quantized in 128 levels using Nyquist sampling rate:

    a. Find SQNR of the PCM signal

    b. Find the sampling frequency required when same signal uses delta modulation for same SQNR

    c. If the system uses DM using Nyquist sampling rate, find SQNR degradation in DM as compared to PCM.

22. Define PAM, PWM and PPM with corresponding waveforms. A television signal having a bandwidth of 4.8 MHz is transmitted using binary PCM system. Given that the number of quantization levels is 512. Determine:

    i. Code word length

    ii. Transmission bandwidth

    iii. Final bit rate

    iv. Output signal to quantization noise ratio

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