2.1 LOW PASS SAMPLING THEOREM

SAMPLING:

A message signal may originate from a digital or analog source. If the message signal is analog in nature, then it has to be converted into digital form before it can transmit by digital means. The process by which the continuous-time signal is converted into a discrete–time signal is called Sampling. Sampling operation is performed in accordance with the sampling theorem.

SAMPLING THEOREM FOR LOW-PASS SIGNALS:-

Statement: - "If a band –limited signal g(t) contains no frequency components for $|f| > W$, then it is completely described by instantaneous values g(kTs) uniformly spaced in time with period $Ts \leq 1/2W$. If the sampling rate, fs is equal to the Nyquist rate or greater ($fs \geq 2W$), the signal g(t) can be exactly reconstructed.

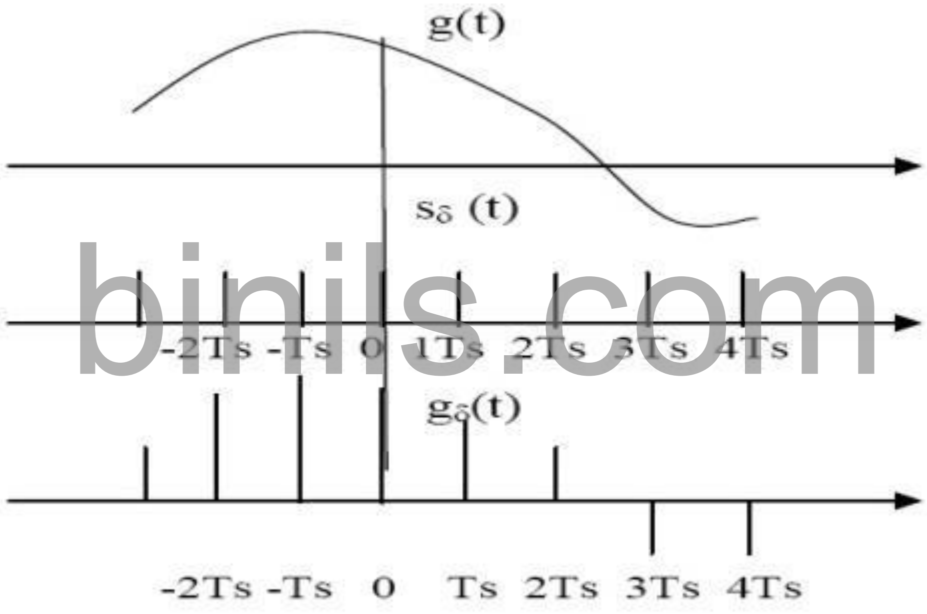


Figure 2.1.1 Sampling Process

Proof:- Part - I If a signal x(t) does not contain any frequency component beyond W Hz, thenthe signal is completely described by its instantaneous uniform samples with sampling interval (or period) of $Ts < 1/(2W)$ sec.

Part – II The signal x(t) can be accurately reconstructed (recovered) from the set of uniform instantaneous samples by passing the samples sequentially through an ideal (brick-wall) lowpass filter with bandwidth B, where $W \leq B < fs - W$ and $fs = 1/(Ts)$.

If x(t) represents a continuous-time signal, the equivalent set of instantaneous uniform samples {x(nTs)} may be represented as,

$$\{x(nTs)\} \equiv xs(t) = \Sigma\, x(t).\delta(t-nTs) \qquad \text{---------- 1.1}$$

where $x(nTs) = x(t)|_{t=nTs}$, δ(t) is a unit pulse singularity function and „n" is an integer.The continuous-time signal x(t) is multiplied by an (ideal) impulse train to obtain {x(nTs)} and can be rewritten as,

$$xs(t) = x(t).\Sigma\, \delta(t-nTs) \qquad \text{------------ 1.2}$$

Now, let X(f) denote the Fourier Transform F(T) of x(t), i.e.

$$X(f) = \int_{-\infty}^{+\infty} x(t).\exp(-j2\pi ft)dt$$

Now, from the theory of Fourier Transform, we know that the F.T of Σ δ(t- nTs), the impulse train in time domain, is an impulse train in frequency domain:

$$F\{\Sigma\, \delta(t-nTs)\} = (1/Ts).\Sigma\, \delta(f-n/Ts) = fs.\Sigma\, \delta(f-nfs) \qquad \text{-- ----- 1.3}$$

If Xs(f) denotes the Fourier transform of the energy signal xs(t), we can write using Eq. (1.2.4) and the convolution property:

Xs(f) = X(f)* F{Σ δ(t- nTs)}

X(f)*[fs.Σ δ(f- nfs)]

= fs.X(f)*Σ δ(f- nfs)

$$= f_s.\int_{-\infty}^{+\infty} X(\lambda).\Sigma\, \delta(f-nf_s-\lambda)d\lambda = f_s.\,\Sigma \int X(\lambda).\delta(f-nf_s-\lambda)d\lambda = f_s.\Sigma\, X(f-nf_s)$$

$$\text{--------------1.4}$$

Quantization:

Quantization is **the process of mapping continuous infinite values to a smaller set of discrete finite values**. In the context of simulation and embedded computing, it is about approximating real-world values with a digital representation that introduces limits on the precision and range of a value.

Types of Quantization

There are two types of Quantization - Uniform Quantization and Non-uniform Quantization.

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

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

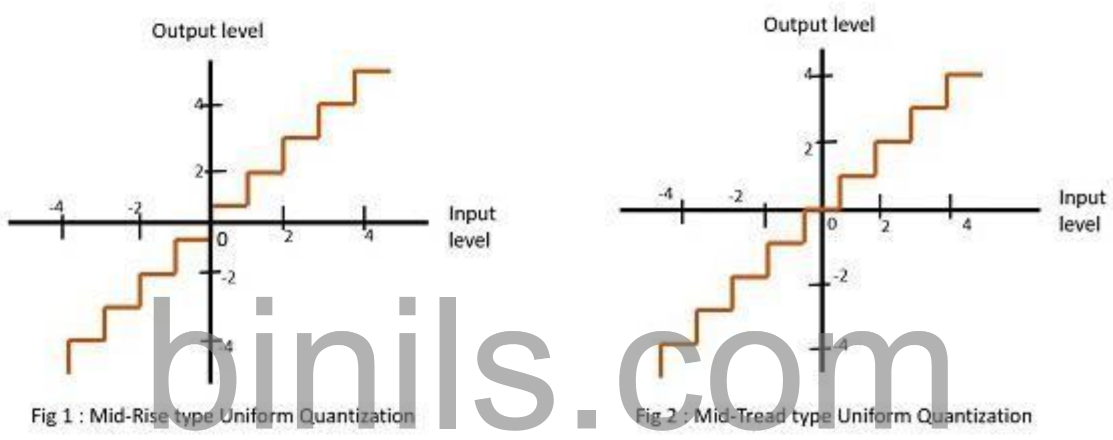


Figure1 (2.1.2) Mid-Rise type Uniform Quantization and figure2 (2.1.3) Mid-Tread type Uniform Quantization

Figure1 (2.1.2) shows the mid-rise type and figure2 (2.1.3) shows the mid-tread type of uniform quantization.

- The **Mid-Rise** type is so called because the origin lies in the middle of a raising part of the stair-case like graph. The quantization levels in this type are even in number.
- The **Mid-tread** type is so called because the origin lies in the middle of a tread of the stair-case like graph. The quantization levels in this type are odd in number.
- Both the mid-rise and mid-tread type of uniform quantizers are symmetric about the origin.

Quantization Error

For any system, during its functioning, there is always a difference in the values of its input and output. The processing of the system results in an error, which is the difference of those values.

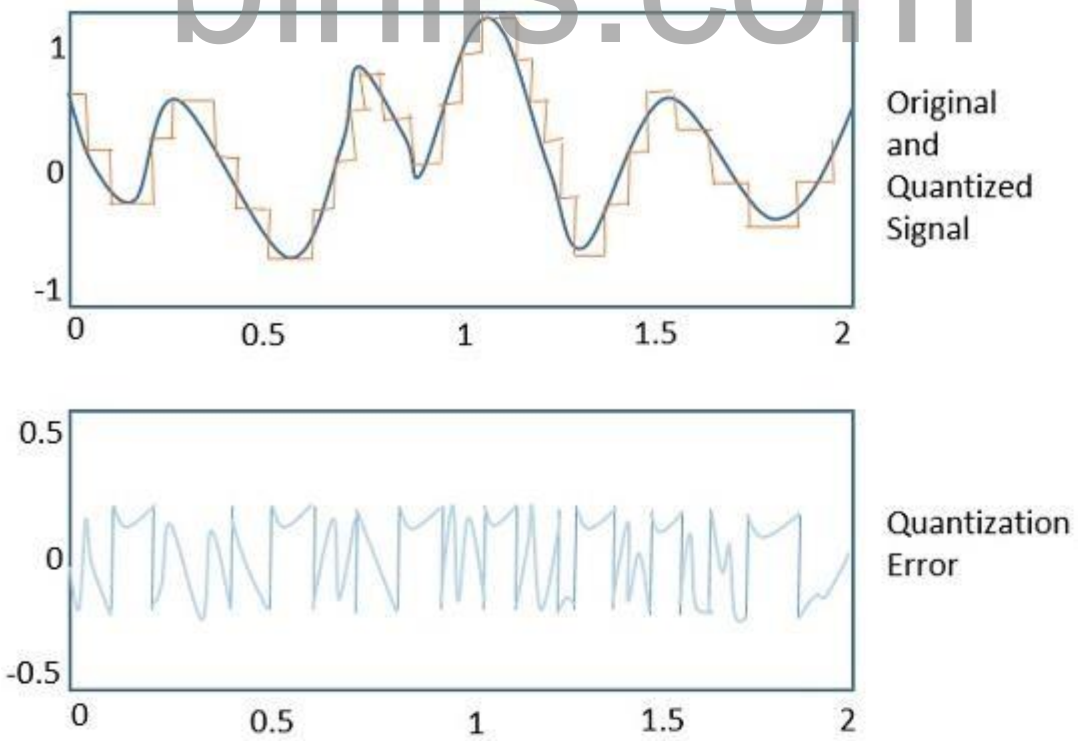
The difference between an input value and its quantized value is called a **Quantization Error**. A **Quantizer** is a logarithmic function that performs Quantization rounding off the value rounding off the value. An analog-to-digital converter (**ADC**) works as a quantizer.

The following figure illustrates an example for a quantization error, indicating the difference between the original signal and the quantized signal.

Figure 2.1.4 Quantization Error

Quantization Noise

It is a type of quantization error, which usually occurs in analog audio signal, while quantizing it to digital. For example, in music, the signals keep changing continuously, where a regularity is not found in errors. Such errors create a wideband noise called as **Quantization Noise**.

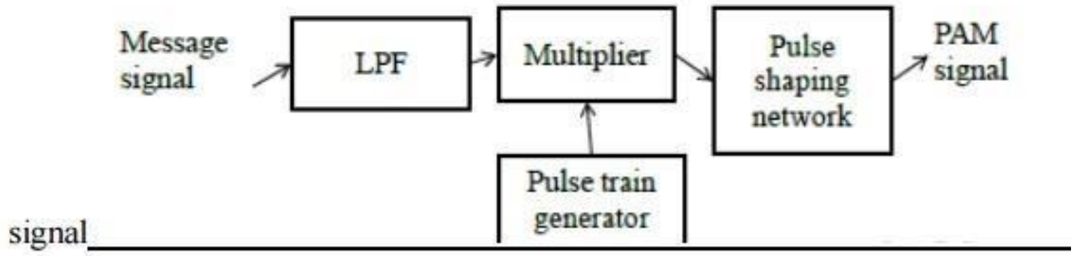


2.2 PULSE AMPLITUDE MODULATION

PAM Modulator

The amplitude of a carrier pulse is altered in accordance to that of amplitude of message signal to make it accommodate the information signal.

• Message signal is transmitted to LPF
• LPF performs band limiting.
• Band limited signal is then sampled at the multiplier.
• Multiplier samples with the help of pulse train generator
• Pulse train generator produces the pulse train
• The multiplication of message signal and pulse train produces PAM



PAM DEMODULATOR:

PAM SIGNAL --→ RECONSTRUCTION FILTER ------→ RECONSTRUCTED PAM SIGNAL

Figure 2.2.1 PAM Modulator

LINE CODING:

Line coding (also called **digital baseband modulation** or digital baseband transmission) is a process carried out by a transmitter that converts data, in the form of binary digits, into a baseband digital signal that will represent the data on a transmission line.

Types of Line Coding

There are 3 types of Line Coding

- Unipolar
- Polar
- Bi-polar

Unipolar Signaling

Unipolar signaling is also called as **On-Off Keying** or simply **OOK**.

The presence of pulse represents a **1** and the absence of pulse represents a **0**.

There are two variations in Unipolar signaling –

- Non Return to Zero NRZNRZ
- Return to Zero RZRZ

Unipolar Non-Return to Zero NRZNRZ

In this type of unipolar signaling, a High in data is represented by a positive pulse called as **Mark**, which has a duration $\mathbf{T_0}$ equal to the symbol bit duration. A Low in data input has no pulse.

The following figure clearly depicts this.

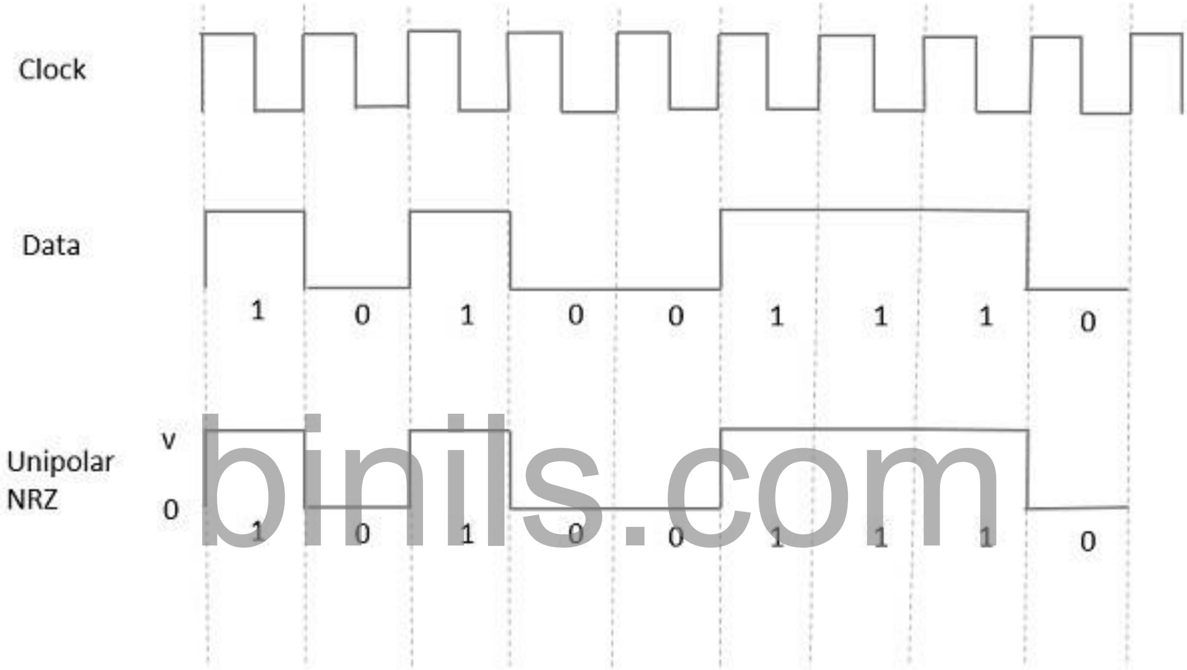


Figure 2.2.2 Unipolar Non-Return to Zero NRZNRZ

Advantages

The advantages of Unipolar NRZ are –

- It is simple.
- A lesser bandwidth is required.

Disadvantages

The disadvantages of Unipolar NRZ are –

- No error correction done.
- Presence of low frequency components may cause the signal droop.
- No clock is present.

- Loss of synchronization is likely to occur (especially for long strings of **1s** and **0s**).

Unipolar Return to Zero RZRZ

In this type of unipolar signaling, a High in data, though represented by a **Mark pulse**, its duration T_0 is less than the symbol bit duration. Half of the bit duration remains high but it immediately returns to zero and shows the absence of pulse during the remaining half of the bit duration.

It is clearly understood with the help of the following figure.

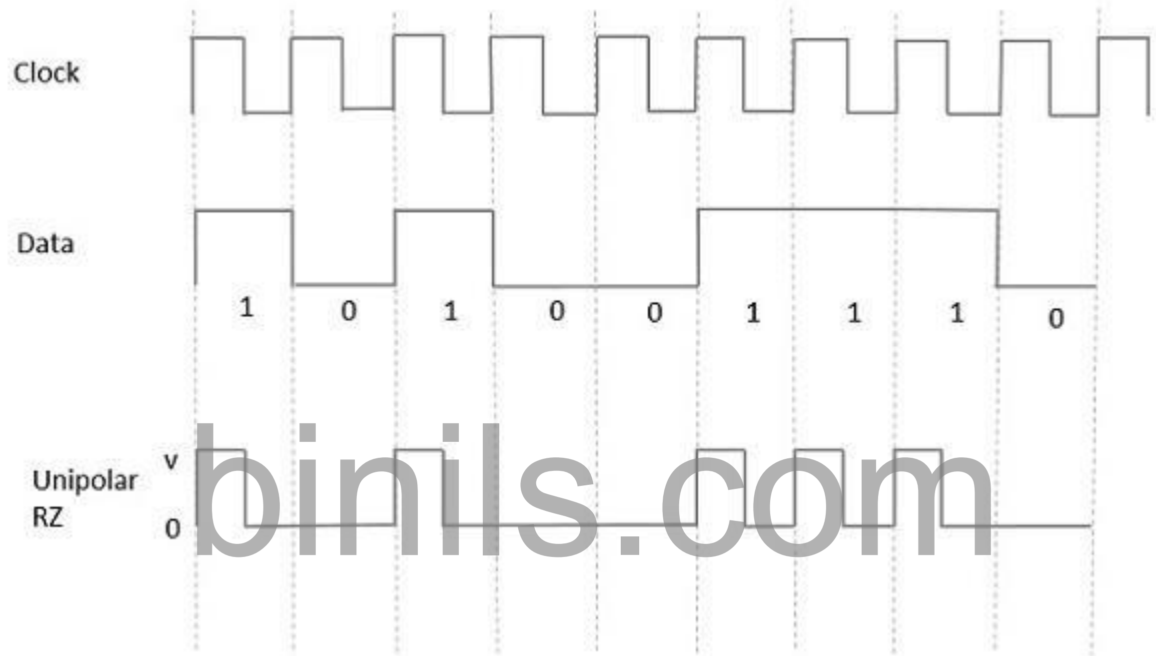


Figure 2.2.3. Unipolar Return to Zero RZRZ

Advantages

The advantages of Unipolar RZ are −

- It is simple.
- The spectral line present at the symbol rate can be used as a clock.

Disadvantages

The disadvantages of Unipolar RZ are −

- No error correction.
- Occupies twice the bandwidth as unipolar NRZ.
- The signal droop is caused at the places where signal is non-zero at 0 Hz.

Polar Signaling

There are two methods of Polar Signaling. They are –

- Polar NRZ
- Polar RZ

Polar NRZ

In this type of Polar signaling, a High in data is represented by a positive pulse, while a Low in data is represented by a negative pulse. The following figure depicts this well.

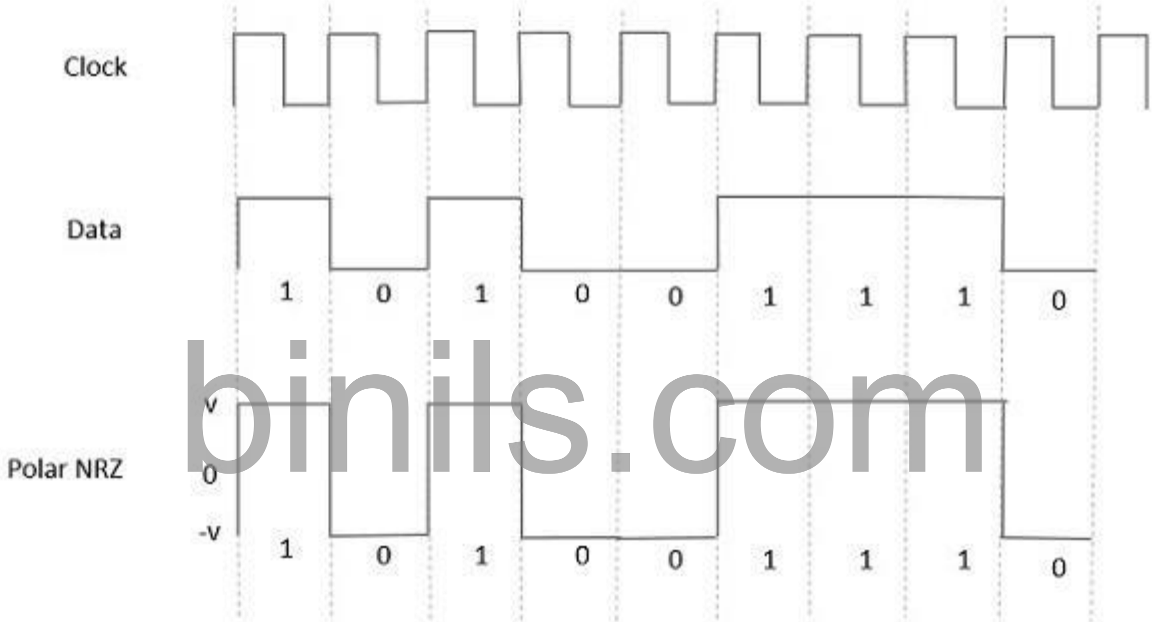


Figure 2.2.4 Polar NRZ

Advantages

The advantages of Polar NRZ are –

- It is simple.
- No low-frequency components are present.

Disadvantages

The disadvantages of Polar NRZ are –

- No error correction.
- No clock is present.

- The signal droop is caused at the places where the signal is non-zero at **0 Hz**.

Polar RZ

In this type of Polar signaling, a High in data, though represented by a **Mark pulse**, its duration T_0 is less than the symbol bit duration. Half of the bit duration remains high but it immediately returns to zero and shows the absence of pulse during the remaining half of the bit duration.

However, for a Low input, a negative pulse represents the data, and the zero level remains same for the other half of the bit duration. The following figure depicts this clearly.

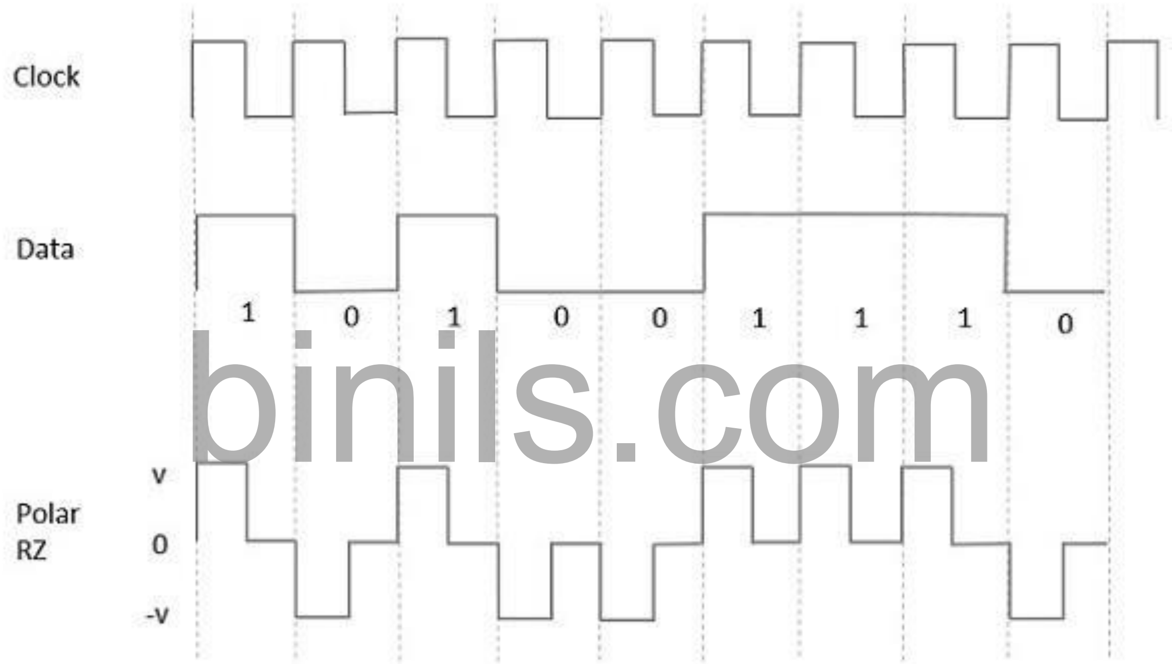


Figure 2.2.5 Polar RZ

Advantages

The advantages of Polar RZ are −

- It is simple.
- No low-frequency components are present.

Disadvantages

The disadvantages of Polar RZ are −

- No error correction.
- No clock is present.

- Occupies twice the bandwidth of Polar NRZ.
- The signal droop is caused at places where the signal is non-zero at **0 Hz**.

Bipolar Signaling

This is an encoding technique which has three voltage levels namely **+, -** and **0**. Such a signal is called as **duo-binary signal**.

An example of this type is **Alternate Mark Inversion** AMIAMI. For a **1**, the voltage level gets a transition from + to – or from – to +, having alternate **1s** to be of equal polarity. A **0** will have a zero voltage level.

Even in this method, we have two types.

- Bipolar NRZ
- Bipolar RZ

From the models so far discussed, we have learnt the difference between NRZ and RZ. It just goes in the same way here too. The following figure clearly depicts this.

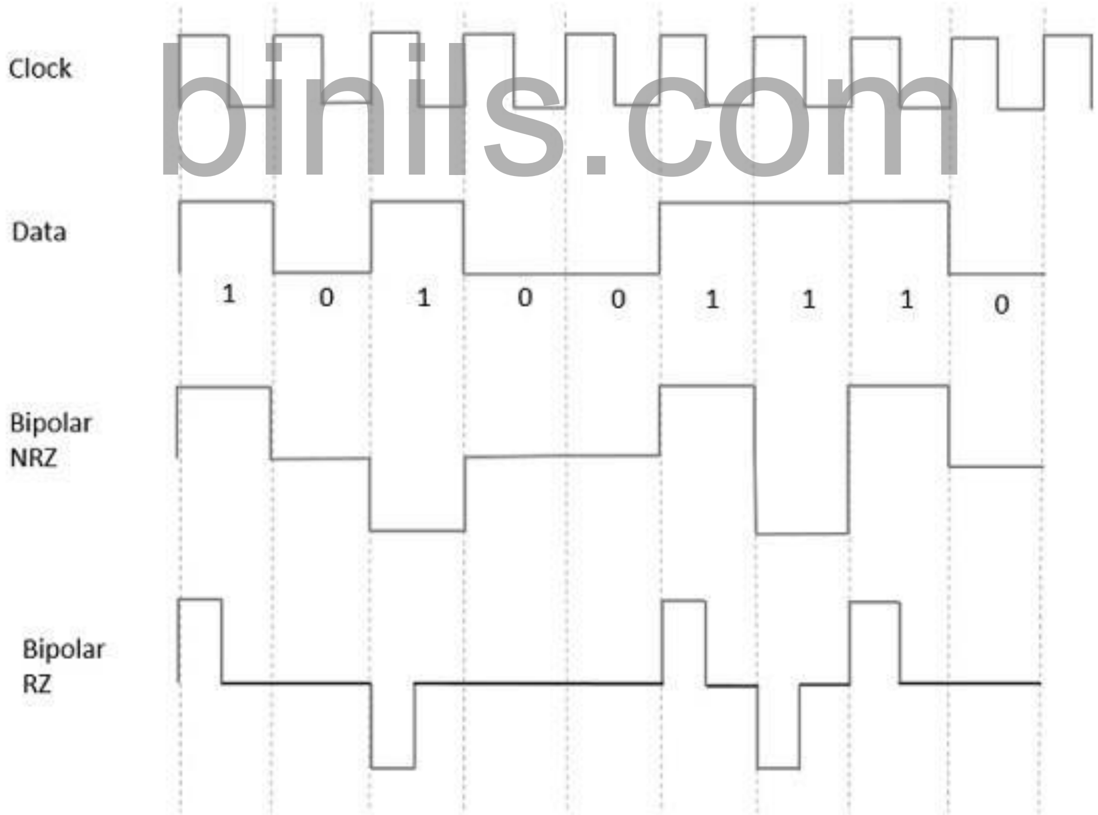


Figure 2.2.6 Bipolar NRZ

The above figure has both the Bipolar NRZ and RZ waveforms. The pulse duration and symbol bit duration are equal in NRZ type, while the pulse duration is half of the symbol bit duration in RZ type.

Advantages

Following are the advantages −

- It is simple.
- No low-frequency components are present.
- Occupies low bandwidth than unipolar and polar NRZ schemes.
- This technique is suitable for transmission over AC coupled lines, as signal drooping doesn't occur here.
- A single error detection capability is present in this.

Disadvantages

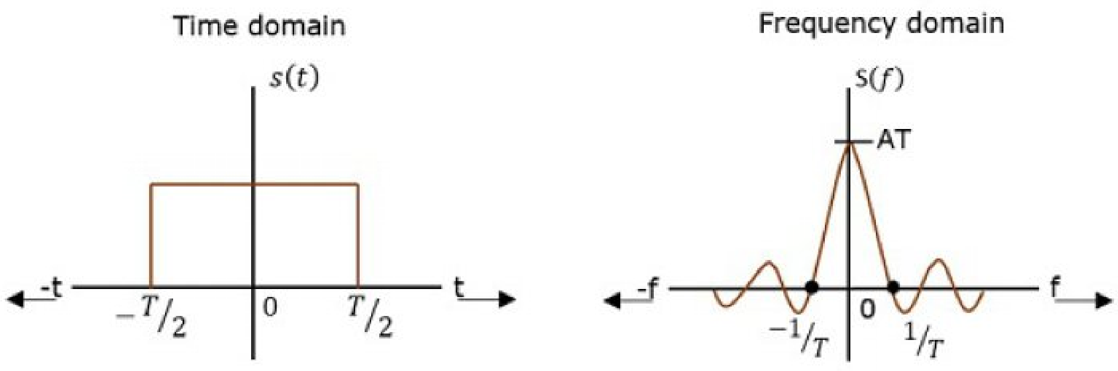
Following are the disadvantages −

- No clock is present.
- Long strings of data causes loss of synchronization.

Power Spectral Density

The function which describes how the power of a signal got distributed at various frequencies, in the frequency domain is called as **Power Spectral Density** PSDPSD.

PSD is the Fourier Transform of Auto-Correlation SimilaritybetweenobservationsSimilaritybetweenobservations. It is in the form of a rectangular pulse.



$$s(t) = A \quad |t| < T/2$$
$$= 0 \quad |t| > T/2$$

$$s(f) = AT\frac{\sin(\pi T f)}{\pi T f}$$

Figure 2.2.7 Power Spectral Density

2.3 PULSE CODE MODULATION:

Pulse code modulation refers a form of source coding. It is a form of digital modulation techniques in which the code refers a binary word that represent digital data.With PCM, the pulses are of fixed length and fixed amplitude.

Block Diagram of Transmitter

Figure 2.3.1 PCM Transmitter

Block Diagram of Receiver

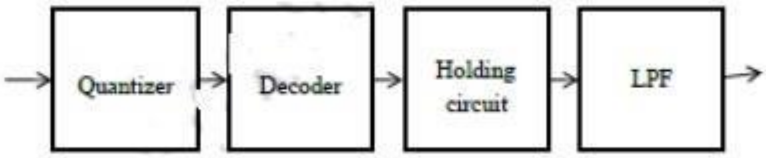


Figure 2.3.2 PCM Receiver

Pulse position modulation

The position of a carrier pulse is altered in accordance with information contained in sampled waveform.

Sampling rate

The sampling rate fs must be atleast two times the highest frequency component of the original signal to be accurately represented fs>=2fm

Baseband signal receiver.

A baseband signal receiver increases the signal to noise at the instant of sampling.

This reduces the probability of error. The baseband signal receiver is also called optimum receiver.

Matched filter.

The matched filter is a baseband signal receiver, which works in presence of white

Gaussian noise. The impulse response of the matched filter is matched to the shape of the input signal.

The impulse response of matched filter

Impulse response is given as,

h (t) = [2k/ N0]{x1 (T − t)}

Here T is the period of sampling x1 (t) and x2 (t) are the two signals used for transmission.

The value of maximum signal to noise ratio of the matched filter
Maximum signal to noise ratio of the matched filter is the ratio of energy of the signal to psd of white noise.
Correlator: It is the coherent receiver. It correlates the received noisy signal f (t) with the locally generated replica of the known signal x (t). Its output is given as,
r (t) = 0**D**T f (t) x(t) dt
Matched filter and correlator are functionally same.
The advantages of QPSK as compared to BPSK

1. For the same bit error rate, the bandwidth required by QPSK Is reduced to half as compared to BPSK.

2. Because of reduced bandwidth, the information transmission rate of QPSK is higher.

3. Variation in QPSK amplitude is not much. Hence carrier power almost remains constant.

DPCM Transmitter

The DPCM Transmitter consists of Quantizer and Predictor with two summer circuits. Following is the block diagram of DPCM transmitter.

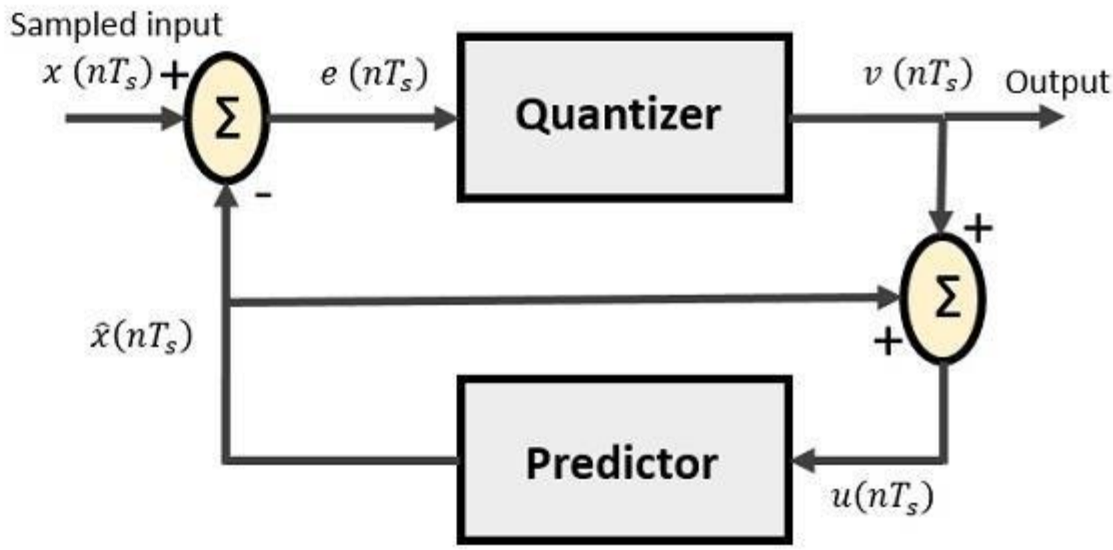


Figure 2.3.3 DPCM Transmitter

The signals at each point are named as −

- x(nTs)x(nTs) is the sampled input
- xˆ(nTs)x^(nTs) is the predicted sample
- e(nTs)e(nTs) is the difference of sampled input and predicted output, often called as prediction error
- v(nTs)v(nTs) is the quantized output
- u(nTs)u(nTs) is the predictor input which is actually the summer output of the predictor output and the quantizer output

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

Quantizer Output is represented as −

v(nTs)=Q[e(nTs)]v(nTs)=Q[e(nTs)]

=e(nTs)+q(nTs)=e(nTs)+q(nTs)

Where **q (nT_s)** is the quantization error

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

u(nTs)=xˆ(nTs)+v(nTs)u(nTs)=x^(nTs)+v(nTs)

u(nTs)=xˆ(nTs)+e(nTs)+q(nTs)u(nTs)=x^(nTs)+e(nTs)+q(nTs)

u(nTs)=x(nTs)+q(nTs)u(nTs)=x(nTs)+q(nTs)

The same predictor circuit is used in the decoder to reconstruct the original input.

DPCM Receiver

The block diagram of DPCM Receiver consists of a decoder, a predictor, and a summer circuit. Following is the diagram of DPCM Receiver.

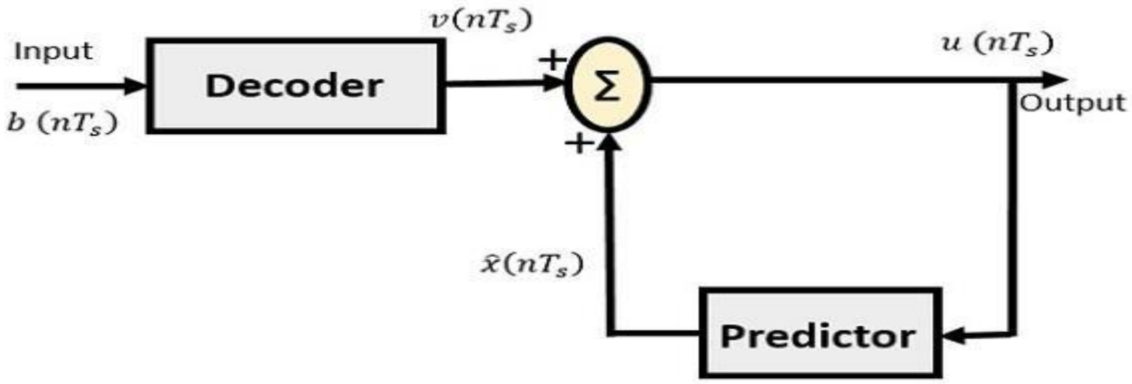


Figure 2.3.4 DPCM Receiver

2.4 DELTA MODULATION

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

Features of Delta Modulation

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** deltadelta.
- The bit rate can be decided by the user.
- This involves simpler implementation.

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

Delta Modulator

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

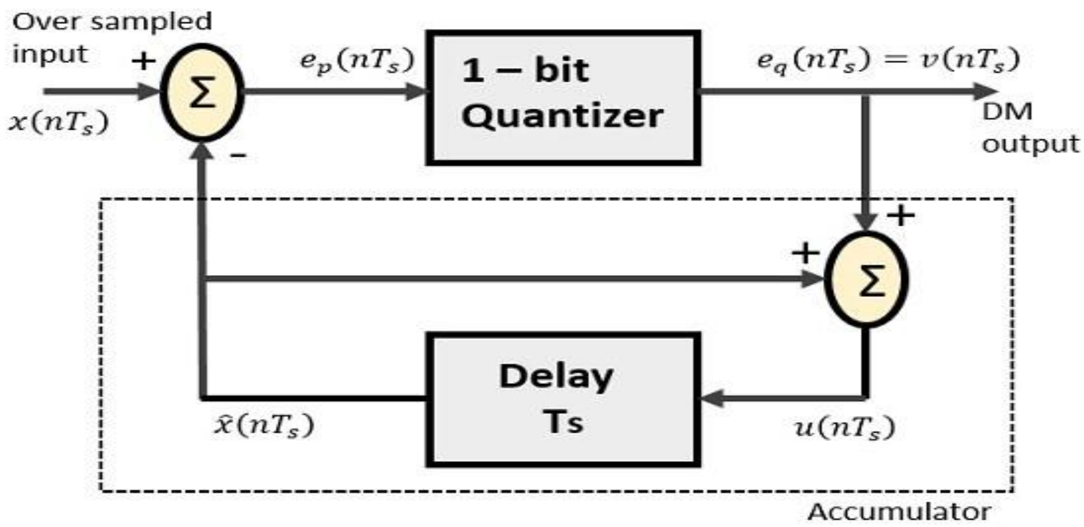


Figure 2.4.1 DM Transmitter

The predictor circuit in DPCM is replaced by a simple delay circuit in DM.

From the above diagram, we have the notations as −

- x(nTs)x(nTs) = over sampled input
- ep(nTs)ep(nTs) = summer output and quantizer input
- eq(nTs)eq(nTs) = quantizer output = v(nTs)v(nTs)
- xˆ(nTs)x^(nTs) = output of delay circuit
- u(nTs)u(nTs) = input of delay circuit

Using these notations, now we shall try to figure out the process of delta modulation.

ep(nTs)=x(nTs)−xˆ(nTs)ep(nTs)=x(nTs)−x^(nTs)

---------equation 1

=x(nTs)−u([n−1]Ts)=x(nTs)−u([n−1]Ts)
=x(nTs)−[xˆ[[n−1]Ts]+v[[n−1]Ts]]=x(nTs)−[x^[[n−1]Ts]+v[[n−1]Ts]]

---------equation 2

Further,

v(nTs)=eq(nTs)=S.sig.[ep(nTs)]v(nTs)=eq(nTs)=S.sig.[ep(nTs)]

---------equation 3

u(nTs)=xˆ(nTs)+eq(nTs)u(nTs)=x^(nTs)+eq(nTs)

Where,

- xˆ(nTs)x^(nTs) = the previous value of the delay circuit
- eq(nTs)eq(nTs) = quantizer output = v(nTs)v(nTs)

Hence,

u(nTs)=u([n−1]Ts)+v(nTs)u(nTs)=u([n−1]Ts)+v(nTs)

---------equation 4

Which means,

The present input of the delay unit

= ThepreviousoutputofthedelayunitThepreviousoutputofthedelayunit **+** thepresentquantizeroutputthepresentquantizeroutput

Assuming zero condition of Accumulation,

u(nTs)=S∑j=1nsig[ep(jTs)]u(nTs)=S∑j=1nsig[ep(jTs)]
Accumulated version of DM output = ∑j=1nv(jTs)∑j=1nv(jTs)

---------equation 5

Now, note that

x^(nTs)=u([n−1]Ts)x^(nTs)=u([n−1]Ts)
=∑j=1n−1v(jTs)=∑j=1n−1v(jTs)

---------equation 6

Delay unit output is an Accumulator output lagging by one sample.

From equations 5 & 6, we get a possible structure for the demodulator.

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

Delta Demodulator

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

Following is the diagram for delta demodulator.

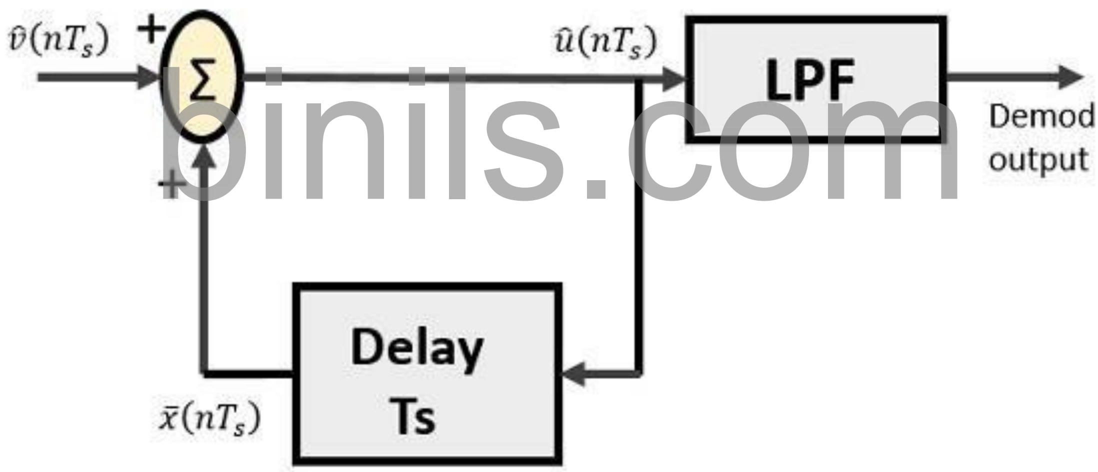


Figure 2.4.2 DM Receiver

From the above diagram, we have the notations as −

- v^(nTs)v^(nTs) is the input sample
- u^(nTs)u^(nTs) is the summer output
- x¯(nTs)x¯(nTs) is the delayed output

A binary sequence will be given as an input to the demodulator. The stair-case approximated output is given to the LPF.

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

Advantages of DM Over DPCM

- 1-bit quantizer
- Very easy design of 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)

Adaptive Delta Modulation ADMADM

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 steep slope of modulating signal and a smaller stepsize 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 block diagram of Adaptive delta modulator.

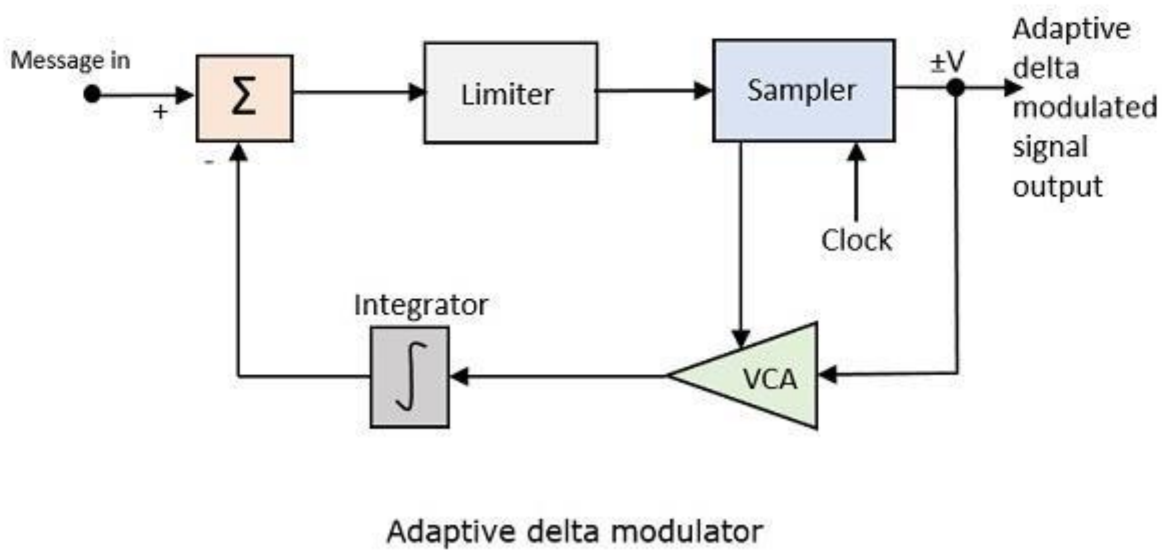


Figure 2.4.3 Adaptive delta modulator

The gain of the voltage controlled amplifier is adjusted by the output signal from the sampler. The amplifier gain determines the step-size and both are proportional.

ADM quantizes the difference between the value of the current sample and the predicted value of the next sample. It uses a variable step height to predict the next values, for the faithful reproduction of the fast varying values.

ADAPTIVE DIFFERENTIAL PULSE-CODE MODULATION

ADPCM stands for Adaptive Differential Pulse-Code Modulation, is a technique for converting analog sound, such as speech, into binary digital information by frequently sampling the sound and expressing its modulation in binary form.

Adaptive Differential Pulse-Code Modulation (ADPCM) codecs convert analog signals into digital information by quantizing the differences between the actual analog signal and a predicted signal.

The result is that analog signals encoded into files using ADPCM have a smaller size than many other formats. ADPCM enables speech information to be compressed into small files for storage and transmission.

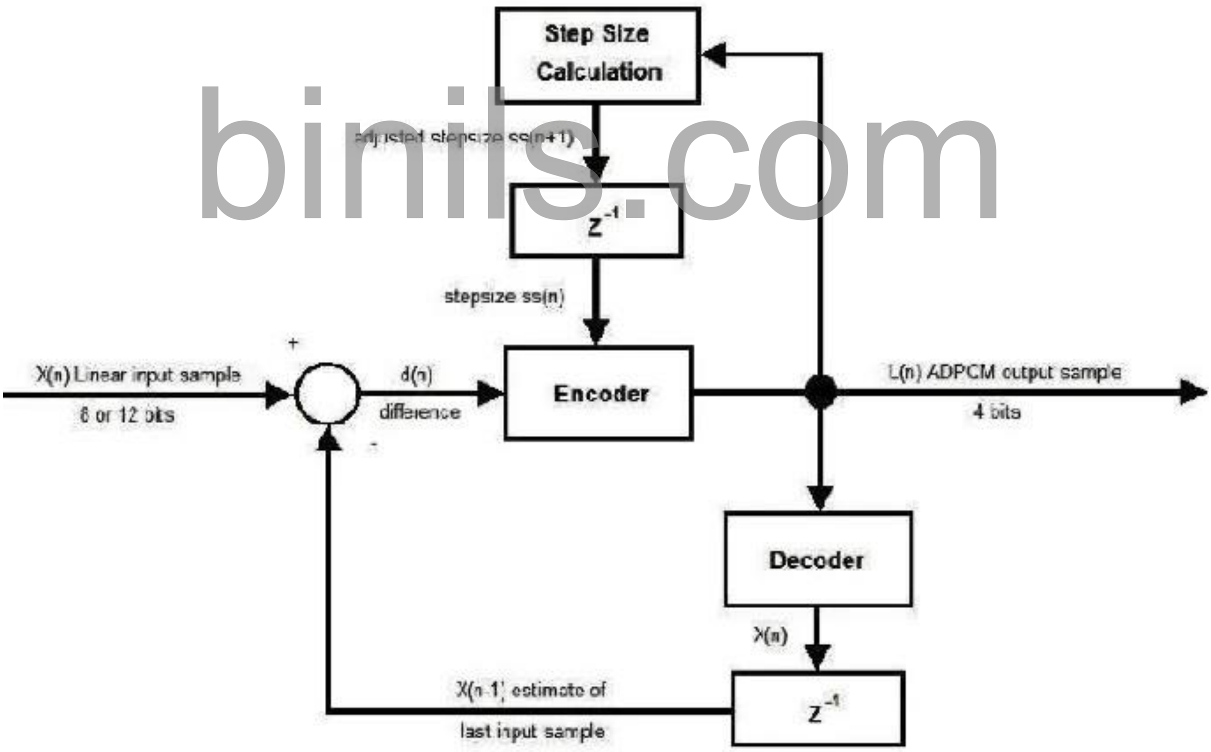


Figure 2.4.4 ADPCM Encoder and Decoder

Encoder

Subsequent to the exchange of both the laws like μ-law & A-law, the input signal of PCM to consistent PCM, a disparity signal can be acquired through subtracting an approximation of the i/p signal from the i/p signal itself. An adaptive 7-, 15-, 31- otherwise 4 level quantizer can be for assigning 5,4,3, or 2 binary digits correspondingly to the difference signal value for broadcast toward the decoder.

An inverse quantizer generates a quantized dissimilarity signal using these binary digits correspondingly. The estimation of the signal can be added toward this quantized difference signal for generating the reconstructed edition of the i/p signal. Both the signals like quantized difference as well as reconstructed are functioned ahead through an adaptive predictor that generates the estimation of the i/p signal, thus finishing the feedback loop.

Decoder

This decoder uses an identical structure toward the feedback part of the encoder, as one through a consistent PCM toward the conversion of A-law otherwise μ-law & an adjustment of synchronous coding.

The adjustment of synchronous coding stops increasing distortion happening on synchronous tandem coding like PCM, ADPCM, ADPCM, etc, under specific conditions. The adjustment of synchronous coding can be achieved through adjusting the output codes of pulse code modulation in a way that efforts to remove quantizing distortion within the subsequent ADPCM encoding phase.

CHANNEL VOCODER:

Definition: **Vocoder** is an audio processor that is used to **transmit speech or voice signal in the form of digital data**. The vocoder is used as short form for **voice coder**. Vocoders are basically used for digital coding of speech and voice simulation. The bitrate for available narrowband vocoders is from **1.2 to 64 kbps**.

Vocoder operates on the principle of **formants**. Formants are basically the meaningful components of a speech that is generated due to the human voice.

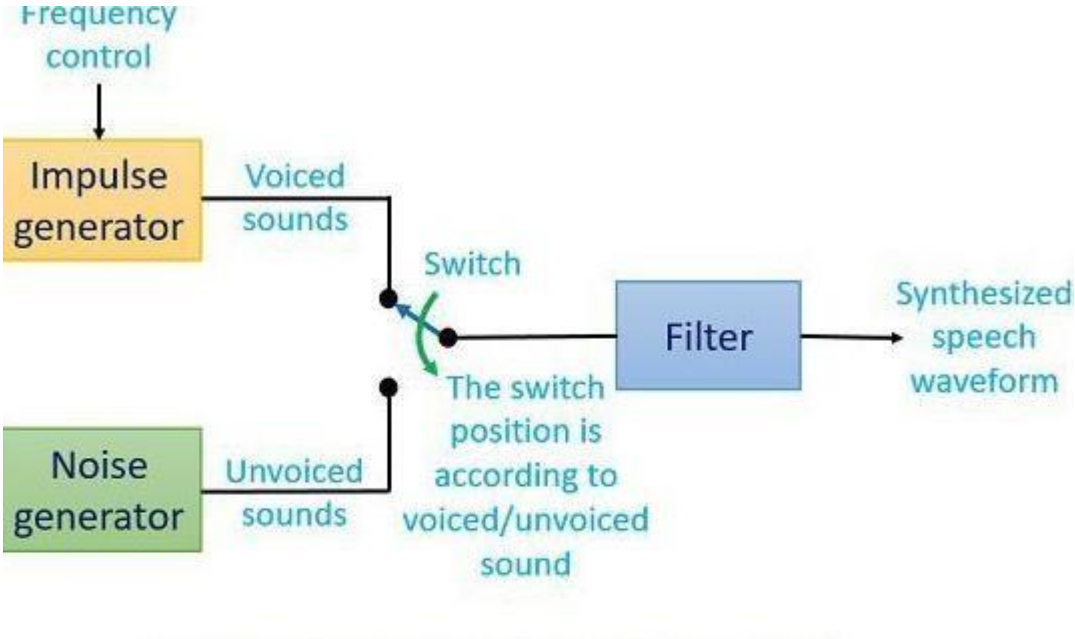


Figure 2.4.5 Speech Model of Vocoder

LPC is extensively used in case of speech and music application. LPC is an acronym for **Linear Predictive Coding**. It is basically a technique to **estimate future values**. In simple words we can say, by analysing two previous samples it predicts the outcome.

Vocoder is comprised of **voice encoder** and **decoder**. Let us now discuss the operation of each in detail-

Voice Encoder

The figure given below shows the block diagram of voice encoder-

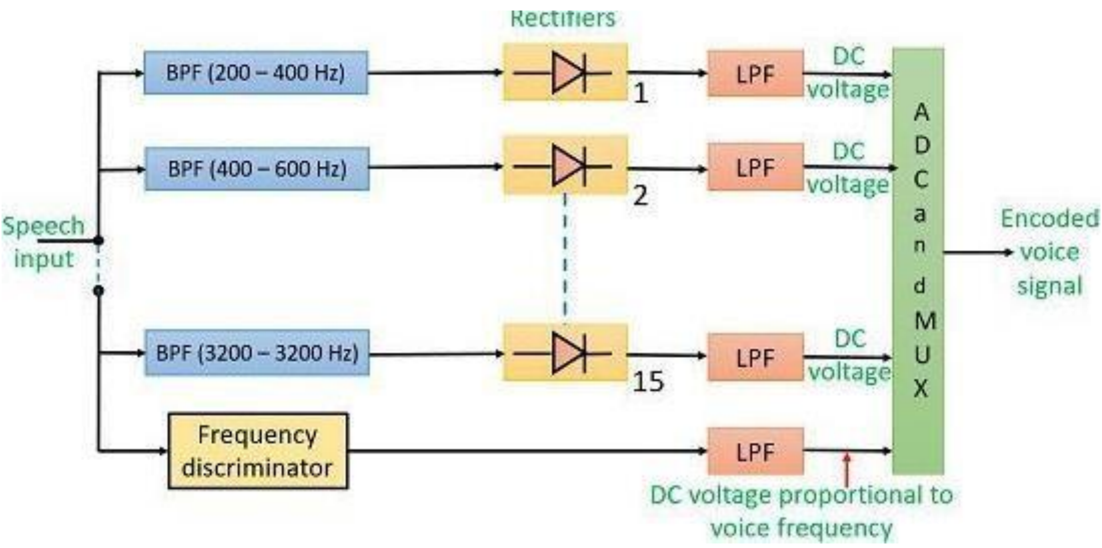


Figure 2.4.6 Voice Encoder

The frequency spectrum of the speech signal **(200Hz – 3200Hz)** is divided into 15 frequency ranges by using 15 Bandpass filter(BPF) each having bandwidth range of 200Hz. The output of BPF acts as input for the rectifier unit.

Here, the signal is rectified and filtered so as to produce a dc voltage. This generated dc voltage is proportional to the amplitude of AC signal present at the output of the filter.

Voice Decoder

The digital voice signal generated by the voice encoder is firstly decoded. Then voice decoder using a speech synthesizer produces voice signal at its output. It generally generates an **approximate voice signal**.

The block diagram of voice decoder section is shown below-

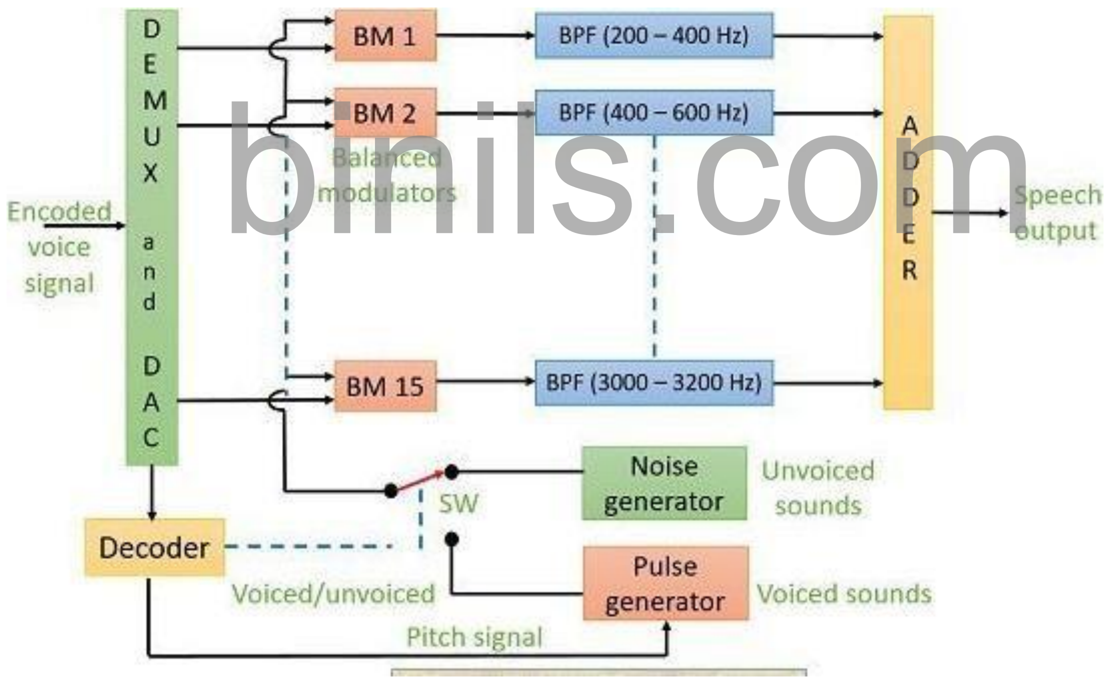


Figure 2.4.7 Voice Decoder

The demultiplexer and DAC section convert the received encoded signal back to its analog form. Here, a balanced modulator(BM)-filter combination is used in correspondence to rectifier-filter combination at the encoder. The carrier to this BM is either the output of noise generator or pulse generator. But this depends on the position of the switch.

However, the **switch position is decided by the decoder**. It is so because when the voiced signal is received, the switch connects the pulse generator output to the input of all the BM.

2.5 TIME DIVISION END FREQUENCY DIVISION MULTIPLEXING

Multiplexing is used in cases where the signals of lower bandwidth and the transmitting media is having higher bandwidth. In this case, the possibility of sending a number of signals is more. In this, the signals are combined into one and are sent over a link that has greater bandwidth of media than the communicating nodes.

1. Frequency Division Multiplexing (FDM) –
In this, a number of signals are transmitted at the same time, and each source transfers its signals in the allotted frequency range. There is a suitable frequency gap between the 2 adjacent signals to avoid over-lapping. Since the signals are transmitted in the allotted frequencies so this decreases the probability of collision. The frequency spectrum is divided into several logical channels, in which every user feels that they possess a particular bandwidth. A number of signals are sent simultaneously at the same time allocating separate frequency bands or channels to each signal. It is used in radio and TV transmission. Therefore to avoid interference between two successive channels **Guard bands** areused.

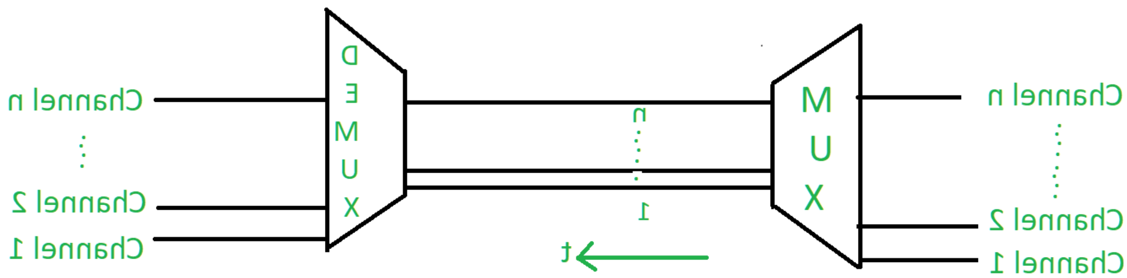


Figure 2.5.1 FDM

2. Time Division Multiplexing (TDM) –

This happens when data transmission rate of media is greater than that of the source, and each signal is allotted a definite amount of time. These slots are so small that all transmissions appear to be parallel. In frequency division

The slots are allocated dynamically depending on the speed of the source or their ready state. It dynamically allocates the time