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.

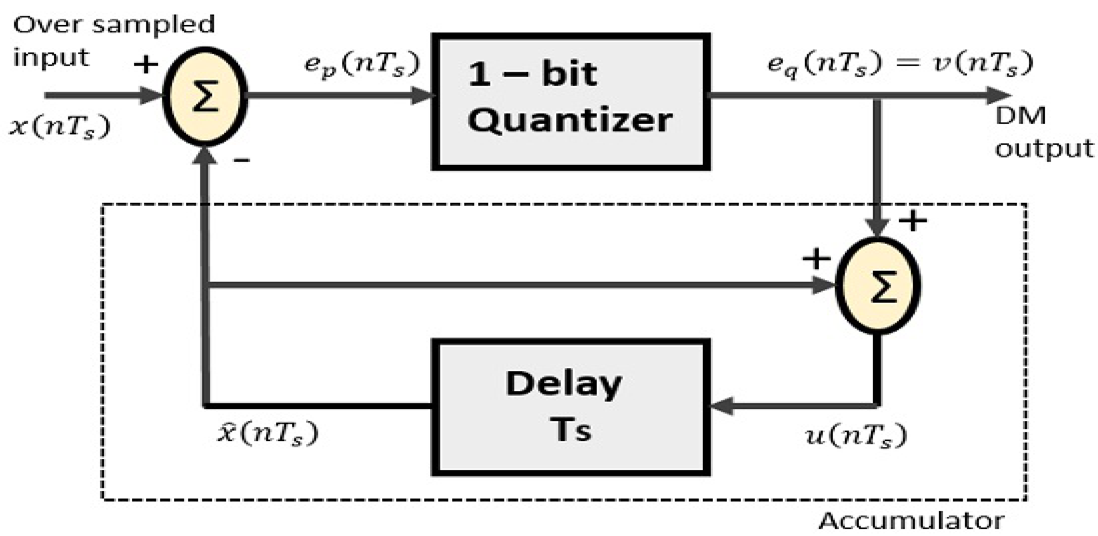

Figure 2.4.1 DM Transmitter

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

From the above diagram, we have the notations as −

- $x(nT_s)x(nT_s)$ = over sampled input
- $e_p(nT_s)e_p(nT_s)$ = summer output and quantizer input
- $e_q(nT_s)e_q(nT_s)$ = quantizer output = $v(nT_s)v(nT_s)$
- $\hat{x}(nT_s)x\hat{}(nT_s)$ = output of delay circuit
- $u(nT_s)u(nT_s)$ = input of delay circuit

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

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

---------equation 1

$=x(nT_s)-u([n-1]T_s)=x(nT_s)-u([n-1]T_s)$
$=x(nT_s)-[\hat{x}[[n-1]T_s]+v[[n-1]T_s]]=x(nT_s)-[x\hat{}[[n-1]T_s]+v[[n-1]T_s]]$

---------equation 2

Further,

$v(nT_s)=e_q(nT_s)=S.sig.[e_p(nT_s)]v(nT_s)=e_q(nT_s)=S.sig.[e_p(nT_s)]$

---------equation 3

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

Where,

- $\hat{x}(nT_s)x\hat{}(nT_s)$ = the previous value of the delay circuit
- $e_q(nT_s)e_q(nT_s)$ = quantizer output = $v(nT_s)v(nT_s)$

Hence,

$u(nT_s)=u([n-1]T_s)+v(nT_s)u(nT_s)=u([n-1]T_s)+v(nT_s)$

---------equation 4

Which means,

The present input of the delay unit

= ThepreviousoutputofthedelayunitThepreviousoutputofthedelayunit **+** thepresentquantizeroutputthepresentquantizeroutput

Assuming zero condition of Accumulation,

$u(nT_s)=S\sum_{j=1}^{n}sig[e_p(jT_s)]u(nT_s)=S\sum_{j=1}^{n}sig[e_p(jT_s)]$
Accumulated version of DM output = $\sum_{j=1}^{n}v(jT_s)\sum_{j=1}^{n}v(jT_s)$

---------equation 5

Now, note that

xˆ(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.

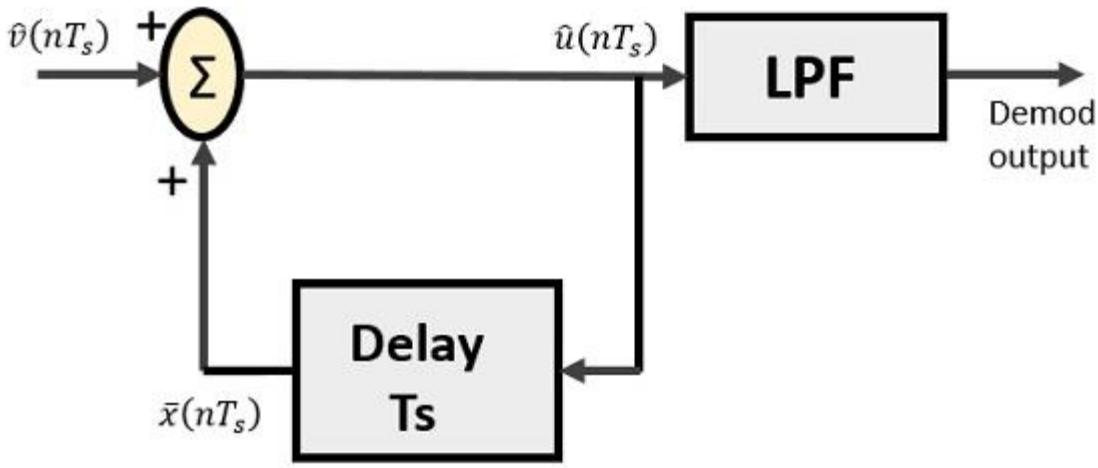

Figure 2.4.2 DM Receiver

From the above diagram, we have the notations as −

- vˆ(nTs)v^(nTs) is the input sample
- uˆ(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.

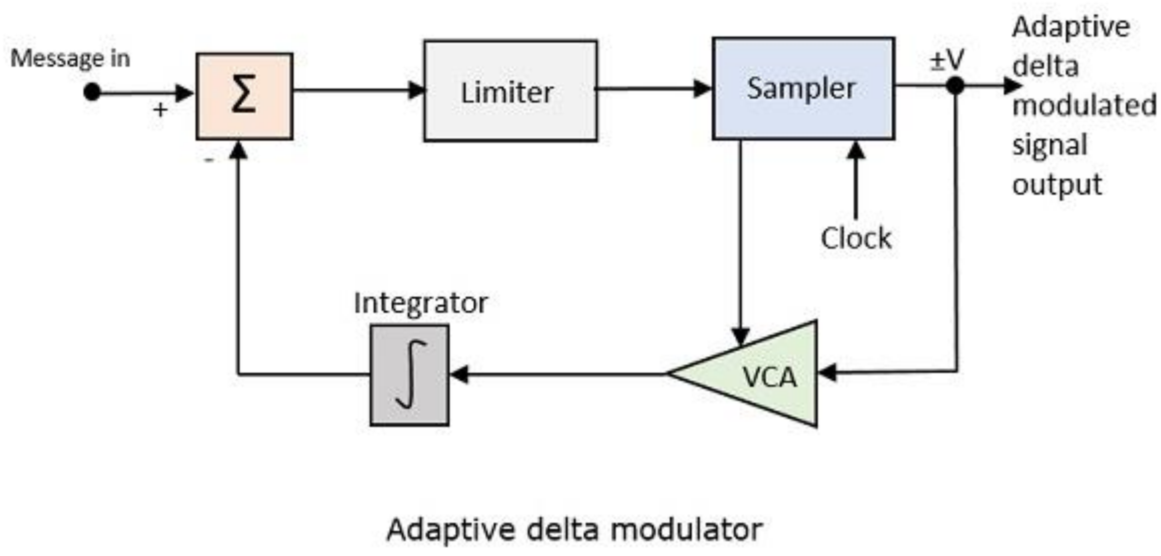

Figure 2.4.3 Adaptive delta modulator

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.

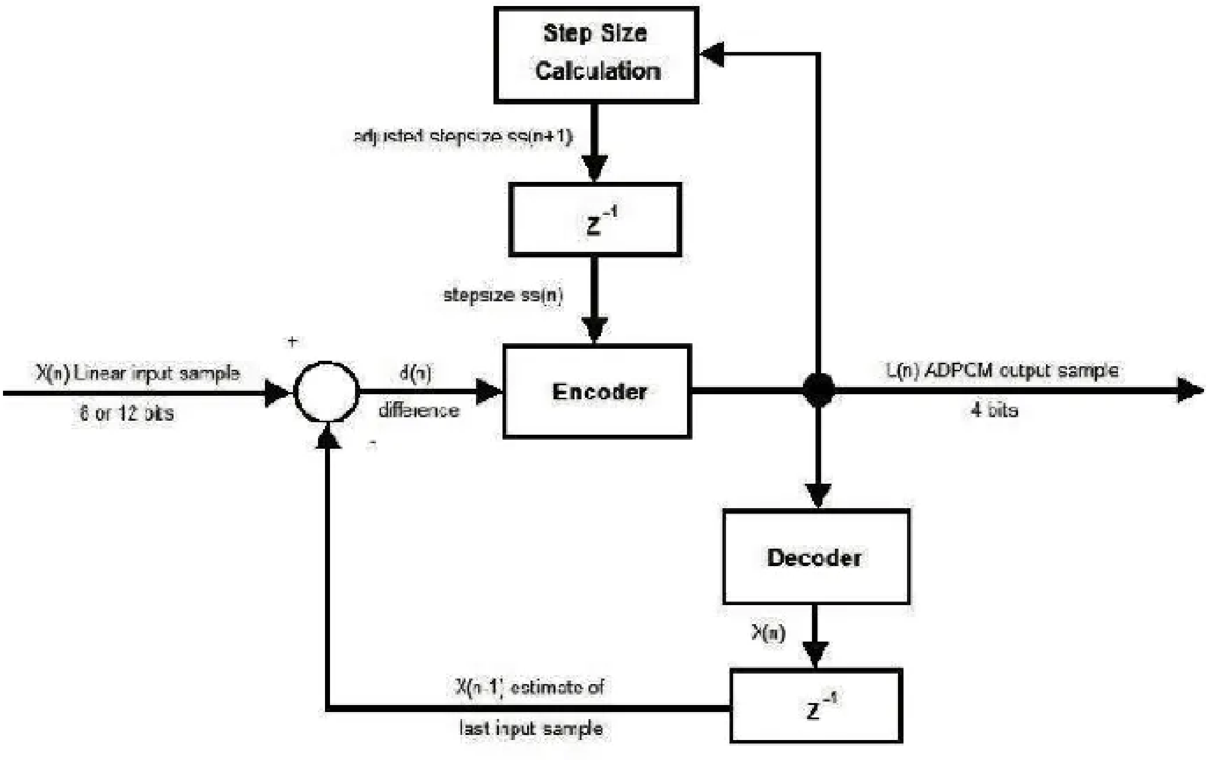

Figure 2.4.4 ADPCM Encoder and Decoder

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.

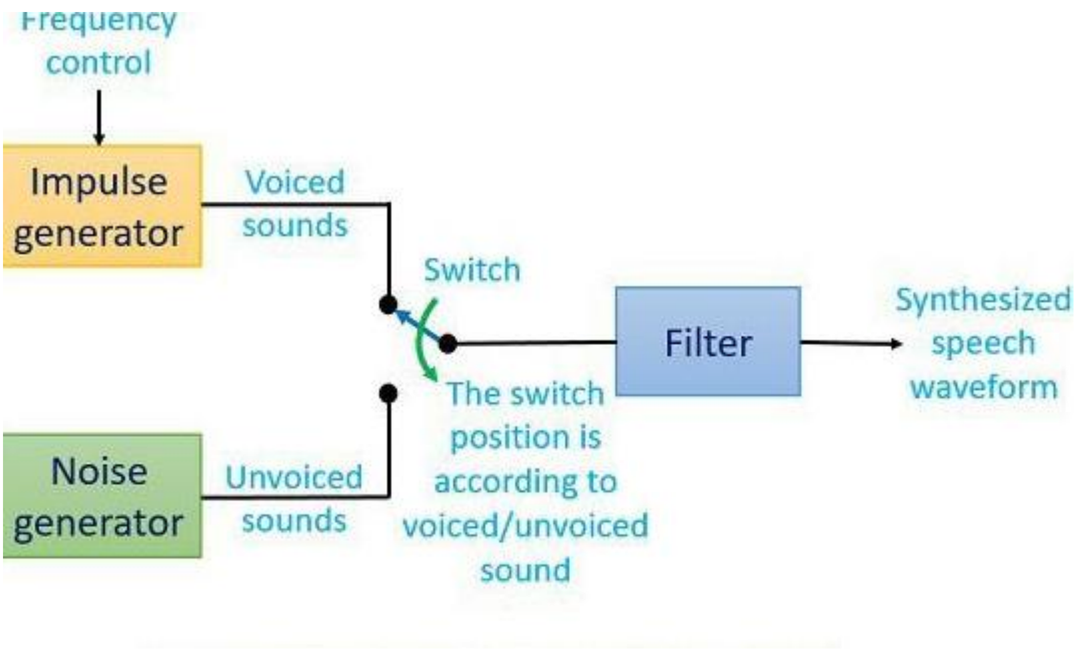

Figure 2.4.5 Speech Model of Vocoder

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-

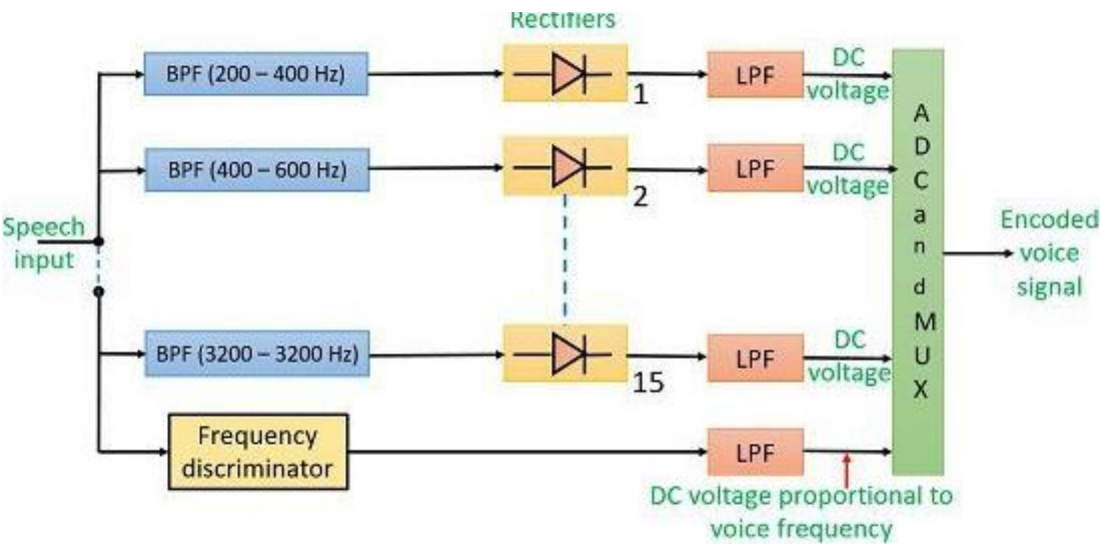

Figure 2.4.6 Voice Encoder

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-

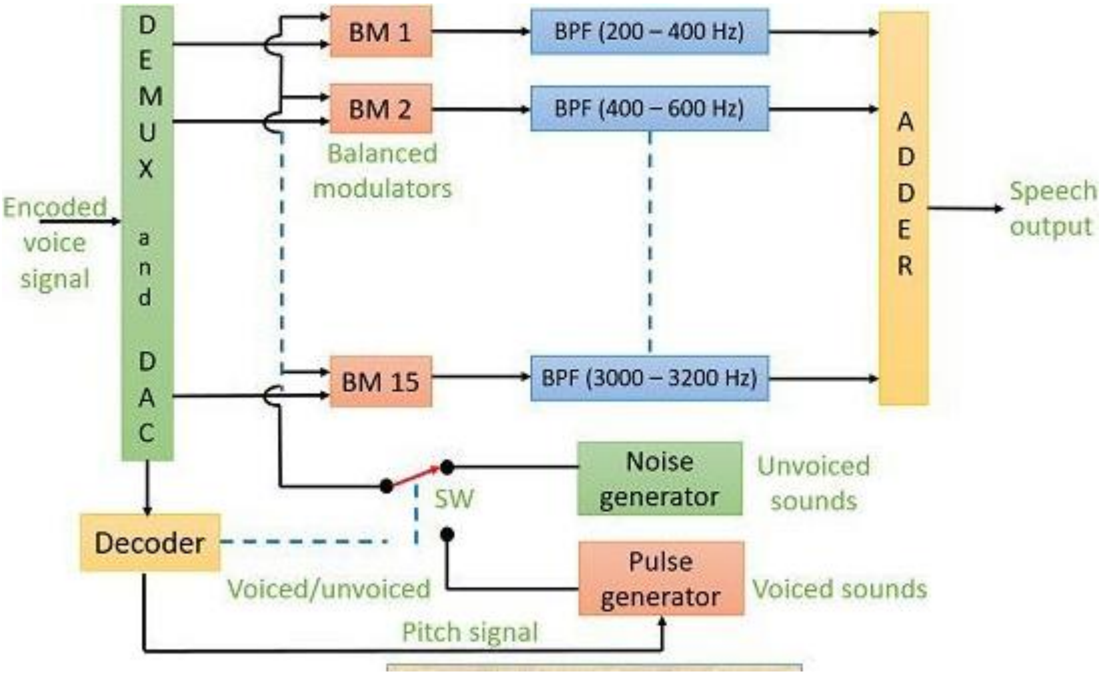

Figure 2.4.7 Voice Decoder

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.