WAVEFORM CODING AND REPRESENTATION

Linear Predictive Coding

Linear predictive coding (**LPC**) is a tool used mostly in audio signal processing and speech processing for representing the spectral envelopment of a digital signal of speech in compressed form, using the information of a linear prediction model.

Linear prediction is a mathematical operation where future values of a discrete time signal are estimated as a linear function of previous samples. In digital signal processing, linear prediction is often called linear predictive coding (LPC) and can thus be viewed as a subset of filter theory.

Filter design is the process of designing a signal processing filter that satisfies a set of requirements, some of which are contradictory. The purpose is to find a realization of the filter that meets each of the requirements to a sufficient degree to make it useful.

The filter design process can be described as an optimization problem where each requirement contributes to an error function which should be minimized. Certain parts of the design process can be automated, but normally an experienced electrical engineer is needed to get a good result.

In system analysis linear prediction can be viewed as a part of mathematical modelling or optimization.

Optimization is the selection of a best element (with regard to some criteria) from some set of available alternatives.

In the simplest case, an optimization problem consists of maximizing or minimizing a real function by systematically choosing input values from within an allowed set and computing the value of the function. The generalization of optimization theory and techniques to other formulations comprises a large area of applied mathematics More generally, optimization includes finding "best available" values of some objective function given a defined domain or a set of constraints), including a variety of different types of objective functions and different types of domains.

LPC starts with the assumption that a speech signal is produced by a buzzer at the end of a tube (voiced sounds), with occasional added hissing and popping sounds. Although apparently crude, this model is actually a close approximation of the reality of speech production. The glottis the space between the vocal folds) produces the buzz, which is characterized by its intensity (loudness) and frequency (pitch). The vocal tract (the throat and mouth) forms the tube, which is characterized by its resonances, which give rise to formats, or enhanced frequency bands in the sound produced. Hisses and pops are generated by the action of the tongue, lips and throat during sibilants and plosives.

LPC analyses the speech signal by estimating the formants, removing their effects from the speech signal, and estimating the intensity and frequency of the remaining buzz. The process of removing the formants is called inverse filtering, and the remaining signal after the subtraction of the filtered modelled signal is called the residue.

The numbers which describe the intensity and frequency of the buzz, the formants, and the residue signal, can be stored or transmitted somewhere else. LPC synthesizes the speech signal by reversing the process: use the buzz parameters and the residue to create a source signal, use the formants to create a filter (which represents the tube), and run the source through the filter, resulting in speech.

Because speech signals vary with time, this process is done on short chunks of the speech signal, which are called frames; generally 30 to 50 frames per second give intelligible speech with good compression.

It is one of the most powerful speech analysis techniques, and one of the most useful methods for encoding good quality speech at a low bit rate and provides extremely accurate estimates of speech parameters.

Differential Pulse Code Modulation (DPCM)

For the signals which does not change rapidly from one sample to next sample, the PCM scheme is not preferred. When such highly correlated samples are encoded the resulting encoded signal contains redundant information. By removing this redundancy before encoding an efficient coded signal can be obtained. One of such scheme is the DPCM technique. By knowing the past behaviour of a signal up to a certain point in time, it is possible to make some inference about the future values. Transmitter: Let x(t) be the signal to be sampled and x(nTs) be its samples. In this scheme the input to the quantizer is a signal. where x^(nTs) is the prediction for un quantized sample x(nTs). This predicted value is produced by using a predictor whose input, consists of a quantized versions of the input signal x(nTs). The signal e(nTs) is called the prediction error.

By encoding the quantizer output, in this method, we obtain a modified version of the PCM called differential pulse code modulation (DPCM).

Quantizer output,

$$v(nTs) = Q[e(nTs)] = e(nTs) + q(nTs)$$

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

$u(nTs) = x\hat{}(nTs) + v(nTs)$

$u(nTs) = x\hat{}(nTs) + e(nTs) + q(nTs)$

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

The receiver consists of a decoder to reconstruct the quantized error signal. The quantized version of the original input is reconstructed from the decoder output using the same predictor as used in the transmitter. In the absence of noise the encoded signal at the receiver input is identical to the encoded signal at the transmitter output. Correspondingly the receive output is equal to u(nTs), which differs from the input x(nts) only by the quantizing error q(nTs).

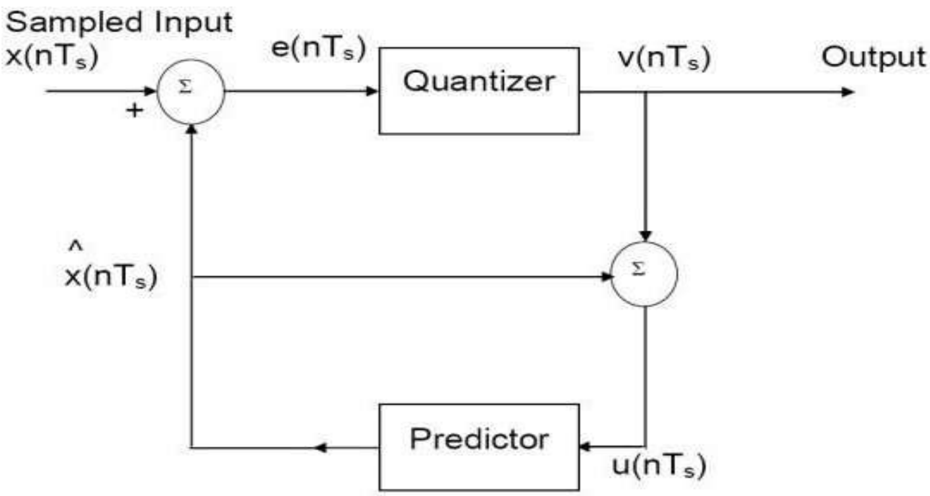


Fig 2.2.1: Block Diagram of DPCM Transmitter
(Source: Electronics Post)

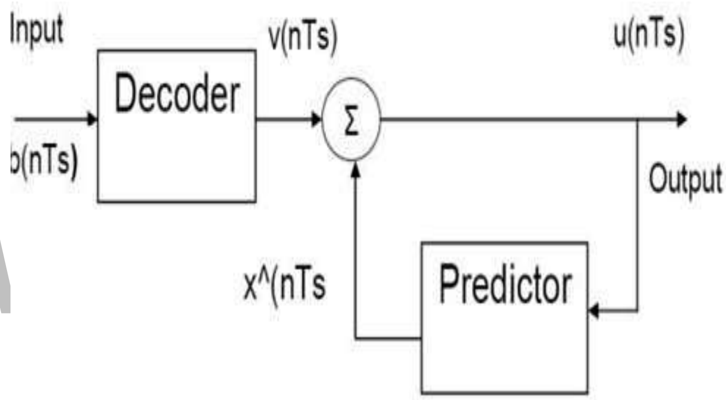


Fig 2.2.2: Block diagram of DPCM Receiver

(Source: Electronics Post)

Delta Modulation (DM)

Delta Modulation is a special case of DPCM. In DPCM scheme if the base band signal is sampled at a rate much higher than the Nyquist rate purposely to increase the correlation between adjacent samples of the signal, so as to permit the use of a simple quantizing strategy for constructing the encoded signal, Delta modulation (DM) is precisely such as scheme. Delta Modulation is the one-bit (or two-level) versions of DPCM.

DM provides a staircase approximation to the over sampled version of an input base band signal. The difference between the input and the approximation is quantized into only two levels, namely, ±δ corresponding to positive and negative differences, respectively, Thus, if the approximation falls below the signal at any sampling epoch, it is increased by δ. Provided that the signal does not change too rapidly from sample to sample, we find that the stair case approximation remains within ±δ of the input signal. The symbol δ denotes the absolute value of the two representation levels of the one-bit quantizer used in the DM.

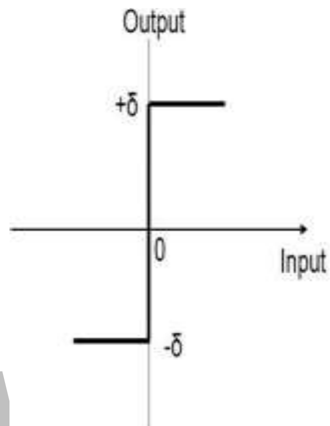


Fig 2.2.3: Input and output characteristics of Delta Modulator

(Source: Tutorials Point)

Let the input signal be x(t) and the staircase approximation to it is u(t).

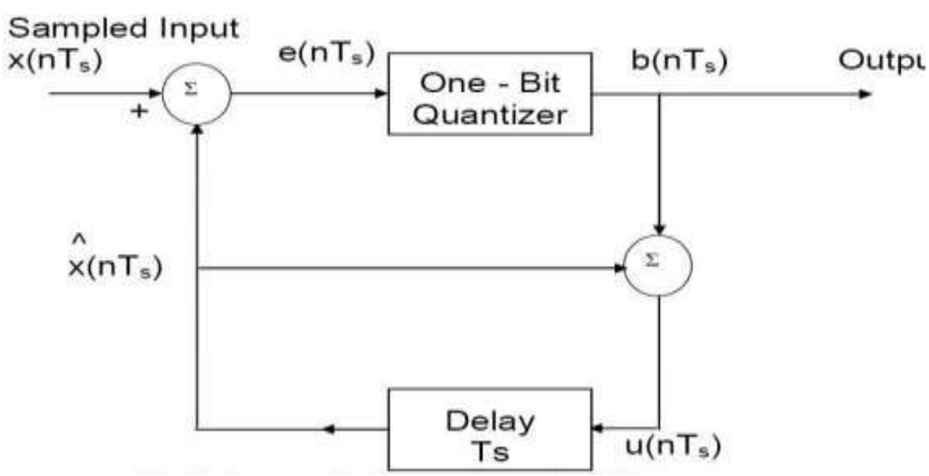


Fig 2.2.4: Block Diagram for transmitter of a DM system

(Source: Tutorials Point)

In the receiver the stair case approximation u(t) is reconstructed by passing the incoming sequence of positive and negative pulses through an accumulator in a manner similar to that used in the transmitter. The out-of –band quantization noise in the high frequency staircase waveform u(t) is rejected by passing it through a low- pass filter with a band-width equal to the original signal bandwidth. Delta modulation offers two unique features:

1. No need for Word Framing because of one-bit code word.
2. Simple design for both Transmitter and Receiver

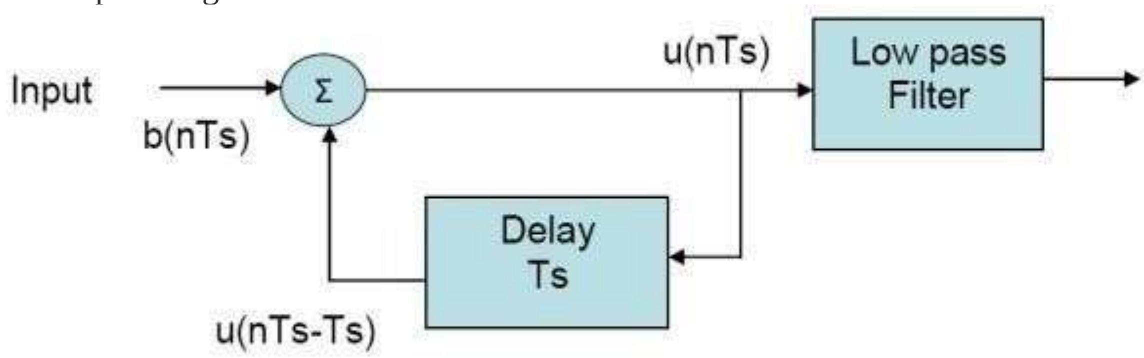


Fig 2.2.5: Block Diagram for a Receiver of a DM
(Source: Tutorials Point)

Disadvantage of DM:

Delta modulation systems are subject to two types of quantization error:

1) slope –overload distortion, and

(2) granular noise.

Adaptive Delta Modulation:

The performance of a delta modulator can be improved significantly by making the step size of the modulator assume a time-varying form. In particular, during a steep segment of the input signal the step size is increased. Conversely, when the input signal is varying slowly, the step size is reduced. In this way, the size is adapted to the level of the input signal. The resulting method is called adaptive delta modulation (ADM). There are several types of ADM, depending on the type of scheme used for adjusting the step size. In this ADM, a discrete set of values is provided for the step size.

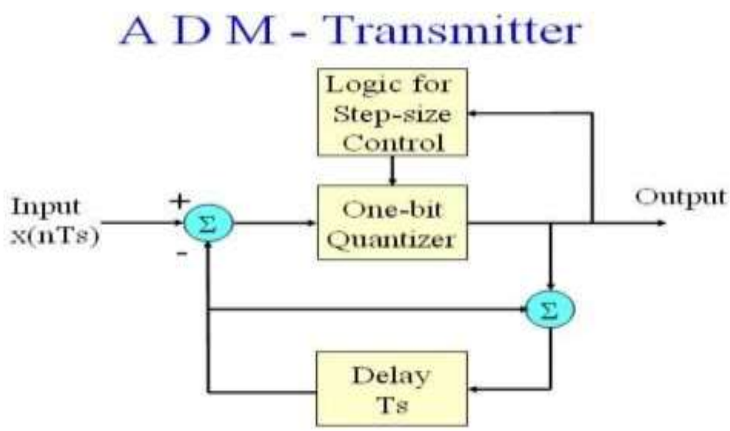


Fig 2.2.6 Block diagram of ADM Transmitter

(Source: EIProCus)

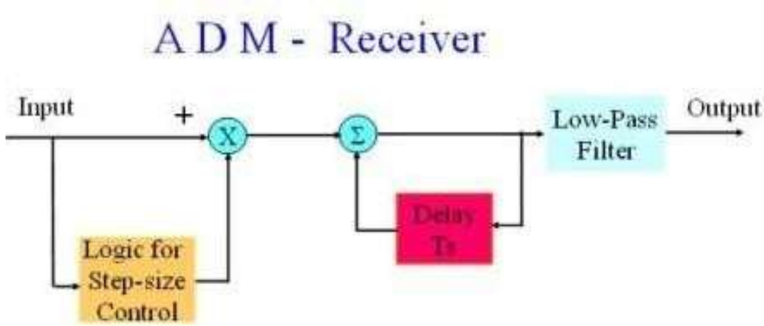


Fig 2.2.7: Block Diagram of ADM Receiver
(Source: EIProCus)

Properties of Line Codes

DC Component:

Eliminating the dc energy from the single power spectrum enables the transmitter to be ac coupled. Magnetic recording system or system using transformer coupling are less sensitive to low frequency signal components. Low frequency component may lost, if the presence of dc or near dc spectral component is significant in the code itself.

Self synchronization

Any digital communication system requires bit synchronization. Coherent detector requires carrier synchronization.

For example Manchester code has a transition at the middle of every bit interval irrespective of whether a 1 or 0 is being sent This guaranteed transmitter provide a clocking signal at the bit level.

Error detection

Some codes such as duo binary provide the means of detecting data error without introducing additional error detection bits into the data sequence.

Band width compression:

Some codes such as multilevel codes increase the efficiency of the bandwidth utilization by allowing a reduction in required bandwidth for a given data rate, thus more information transmitted per unit band width.

DIFFERENTIAL ENCODING

This technique is useful because it allow the polarity of differentially encoded waveform to be inverted without affecting the data detection. In communication system where waveform to be inverted having great advantage.

NOISE IMMUNITY

For same transmitted energy some codes produces lesser bit detection error than other in the presence of noise. For ex. The NRZ waveforms have better noise performance than the RZ type.

SPECTRAL COMPATABILITY WITH CHANNEL:

On aspect of spectrum matching is dc coupling. Also transmission bandwidth of the code musts is sufficient small compared to channel bandwidth so that ISI is not problem.

TRANSPARENCY

A line doe should be so designed that the receiver does not go out of synchronization for any line sequence of data symbol. A code is not transparent if for some sequence of symbol, the clock is lost.

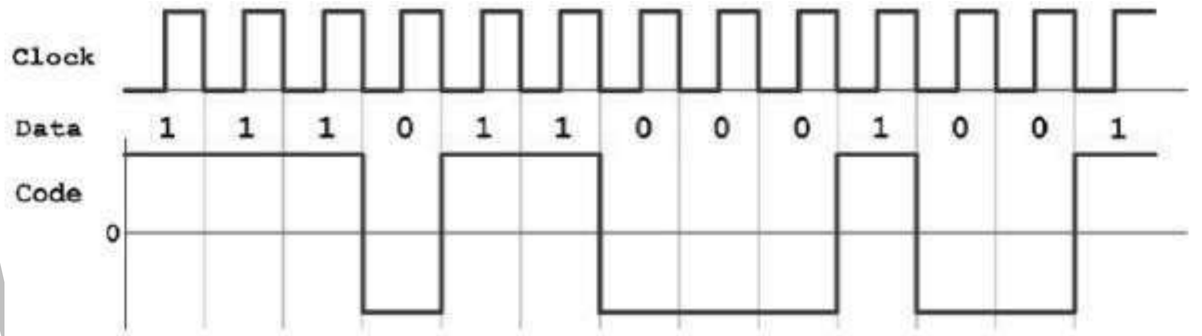
Power spectral density of unipolar NRZ line code

Line coding:

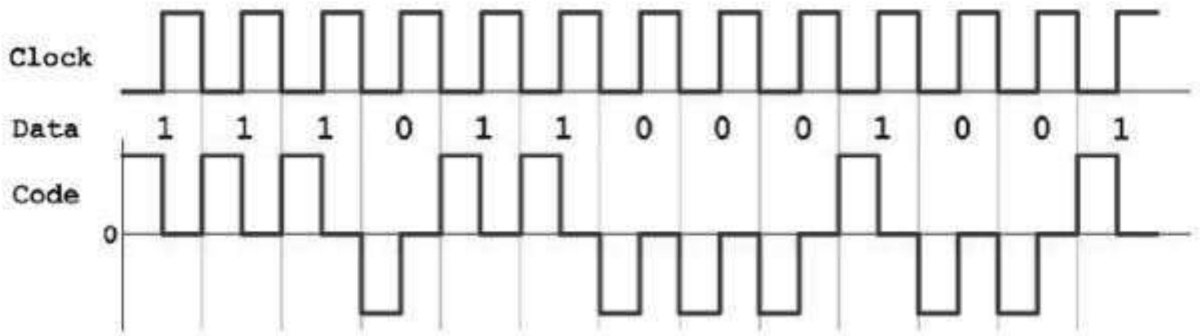
Line coding refers to the process of representing the bit stream (**1**s and **0**s) in the form of voltage or current variations optimally tuned for the specific properties of the physical channel being used. The selection of a proper line code can help in so many ways: One possibility is to aid in clock recovery at the receiver. A clock signal is recovered by observing transitions in the received bit sequence, and if enough transitions exist, a good recovery of the clock is guaranteed, and the signal is said to be **self-clocking**.

Another advantage is to get rid of DC shifts. The DC component in a line code is called the *bias* or the *DC coefficient*. Unfortunately, most long-distance communication channels cannot transport a DC component. This is why most line codes try to eliminate the DC component before being transmitted on the channel. Such codes are called *DC balanced*, *zero-DC*, *zero-bias*, or *DC equalized*. Some common types of line encoding in common-use nowadays are unipolar, polar, bipolar, Manchester, MLT-3 and Duo binary encoding. These codes are explained here: **1. Unipolar**(Unipolar **NRZ** and Unipolar **RZ**):

Unipolar is the simplest line coding scheme possible. It has the advantage of being compatible with TTL logic. Unipolar coding uses a positive rectangular pulse *p*(*t*) to represent binary **1**, and the absence of a pulse (i.e., zero voltage) to represent a binary **0**. Two possibilities for the pulse*p*(*t*) exist3: Non-Return-to-Zero (NRZ) rectangular pulse and Return-to-Zero (RZ) rectangular pulse. The difference between Unipolar NRZ and Unipolar RZ codes is that the rectangular pulse in NRZ stays at a positive value (e.g., +5V) for the full duration of the logic **1** bit, while the pule in RZ drops from +5V to 0V in the middle of the bit time. A drawback of unipolar (RZ and NRZ) is that its average value is not zero, which means it creates a significant DC-component at the receiver (see the impulse at zero frequency in the corresponding power spectral density (PSD) of this line code.



Polar NRZ Code

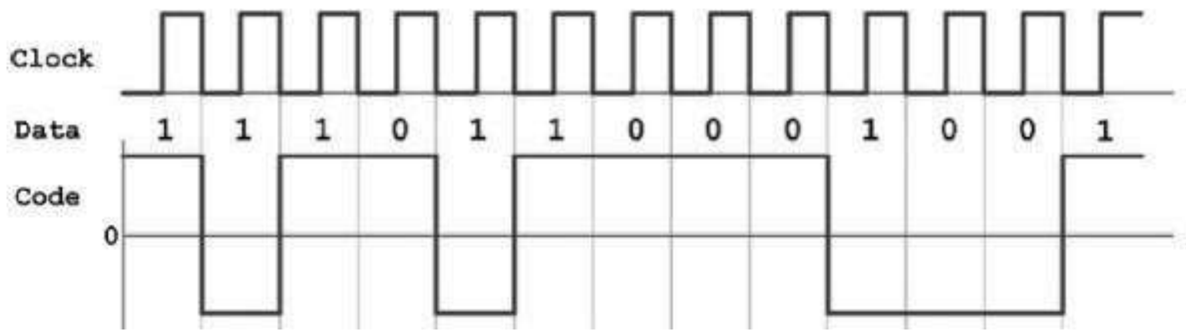


Polar RZ Code

(Source:Studytronics)

that polar signals have more power than unipolar signals, and hence have better SNR at the receiver. Actually, polar NRZ signals have more power compared to polar RZ signals. The drawback of polar NRZ, however, is that it lacks clock information especially when a long sequence of **0**‟s or **1**‟s is transmitted.

Non-Return-to-Zero, Inverted (NRZI): NRZI is a variant of Polar NRZ. In NRZI there are two possible pulses, $p(t)$ and $-p(t)$. A transition from one pulse to the other happens if the bit being transmitted is logic **1**, and no transition happens if the bit being transmitted is a logic **0**.

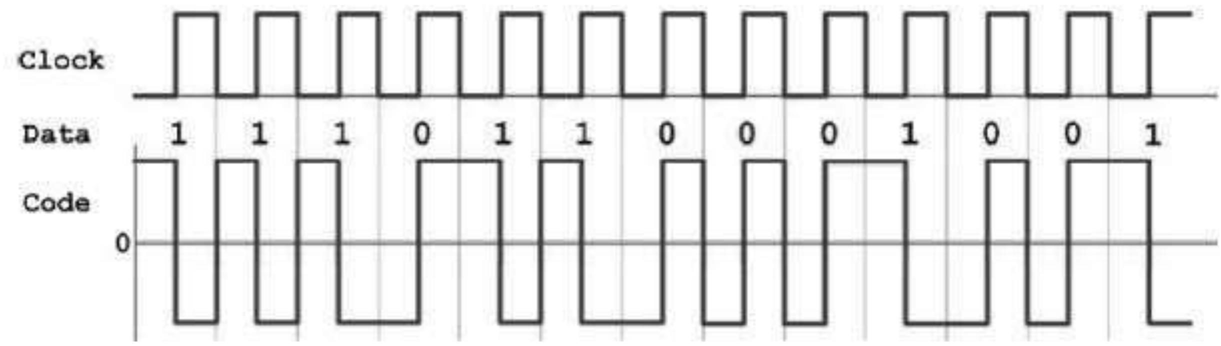


(Source:Studytronics)

This is the code used on compact discs (CD), USB ports, and on fiber-based Fast Ethernet at 100-Mbit/s.

Manchester encoding

In Manchester code each bit of data is signified by at least one transition. Manchester encoding is therefore considered to be self-clocking, which means that accurate clock recovery from a data stream is possible. In addition, the DC component of the encoded signal is zero. Although transitions allow the signal to be self-clocking, it carries significant overhead as there is a need for essentially twice the bandwidth of a simple NRZ or NRZI encoding.



(Source:Studytronics)

The power spectral density of polar signaling