Module 2

Data Communication Fundamentals

Lesson 4

Transmission of Digital Signal

Specific Instructional Objective

On completion, the students will be able to:

- Explain the need for digital transmission
- Explain the basic concepts of Line Coding
- Explain the important characteristics of line coding
- Distinguish among various line coding techniques
  - Unipolar
  - Polar
  - Bipolar
- Distinguish between data rate and modulation rate

2.4.1 Introduction

A computer network is used for communication of data from one station to another station in the network. We have seen that analog or digital data traverses through a communication media in the form of a signal from the source to the destination. The channel bridging the transmitter and the receiver may be a guided transmission medium such as a wire or a wave-guide or it can be an unguided atmospheric or space channel. But, irrespective of the medium, the signal traversing the channel becomes attenuated and distorted with increasing distance. Hence a process is adopted to match the properties of the transmitted signal to the channel characteristics so as to efficiently communicate over the transmission media. There are two alternatives; the data can be either converted to digital or analog signal. Both the approaches have pros and cons. What to be used depends on the situation and the available bandwidth.

Now, either form of data can be encoded into either form of signal. For digital signalling, the data source can be either analog or digital, which is encoded into digital signal, using different encoding techniques.

The basis of analog signalling is a constant frequency signal known as a *carrier signal*, which is chosen to be compatible with the transmission media being used, so that it can traverse a long distance with minimum of attenuation and distortion. Data can be transmitted using these carrier signals by a process called *modulation*, where one or more fundamental parameters of the carrier wave, i.e. amplitude, frequency and phase are being modulated by the source data. The resulting signal, called *modulated signal* traverses the media, which is *demodulated* at the receiving end and the original signal is extracted. All the four possibilities are shown in Fig. 2.4.1.

Data	Signal	Approach
Digital	*Digital*	*Encoding*
Analog	*Digital*	*Encoding*
Analog	*Analog*	*Modulation*
Digital	*Analog*	*Modulation*

Figure 2.4.1 *Various approaches for conversion of data to signal*

This lesson will be concerned with various techniques for conversion digital and analog data to digital signal, commonly referred to as **encoding** techniques.

2.4.2 Line coding characteristics

The first approach converts digital data to digital signal, known as line coding, as shown in Fig. 2.4.2. Important parameters those characteristics line coding techniques are mentioned below.

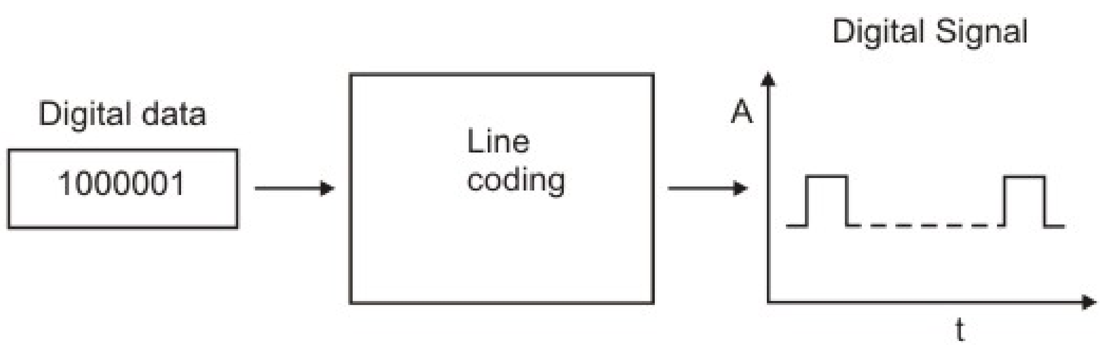


Figure 2.4.2 *Line coding to convert digital data to digital signal*

No of signal levels: This refers to the number values allowed in a signal, known as s**ignal levels**, to represent data. Figure 2.4.3(a) shows two signal levels, whereas Fig. 2.4.3(b) shows three signal levels to represent binary data.

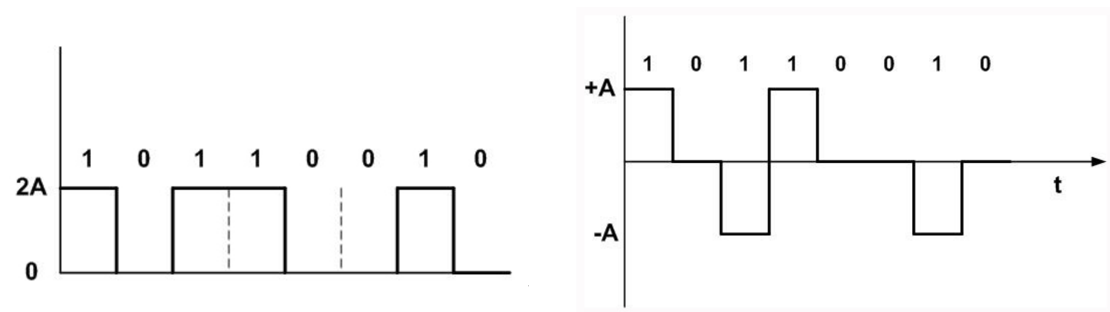
Bit rate versus Baud rate: The **bit rate** represents the number of bits sent per second, whereas the **baud rate** defines the number of signal elements per second in the signal. Depending on the encoding technique used, baud rate may be more than or less than the data rate.

DC components: After line coding, the signal may have zero frequency component in the spectrum of the signal, which is known as the direct-current **(DC) component**. DC component in a signal is not desirable because the DC component does not pass through some components of a communication system such as a transformer. This leads to distortion of the signal and may create error at the output. The DC component also results in unwanted energy loss on the line.

Signal Spectrum: Different encoding of data leads to different spectrum of the signal. It is necessary to use suitable encoding technique to match with the medium so that the signal suffers minimum attenuation and distortion as it is transmitted through a medium.

Synchronization: To interpret the received signal correctly, the bit interval of the receiver should be exactly same or within certain limit of that of the transmitter. Any mismatch between the two may lead wrong interpretation of the received signal. Usually, clock is generated and synchronized from the received signal with the help of a special hardware known as Phase Lock Loop (PLL). However, this can be achieved if the received signal is self-synchronizing having frequent transitions (preferably, a minimum of one transition per bit interval) in the signal.

Cost of Implementation: It is desirable to keep the encoding technique simple enough such that it does not incur high cost of implementation.



(a) (b)

Figure 2.4.3 *(a) Signal with two voltage levels, (b) Signal with three voltage levels*

2.4.3 Line Coding Techniques

Line coding techniques can be broadly divided into three broad categories: Unipolar, Polar and Bipolar, as shown in Fig. 2.4.4.

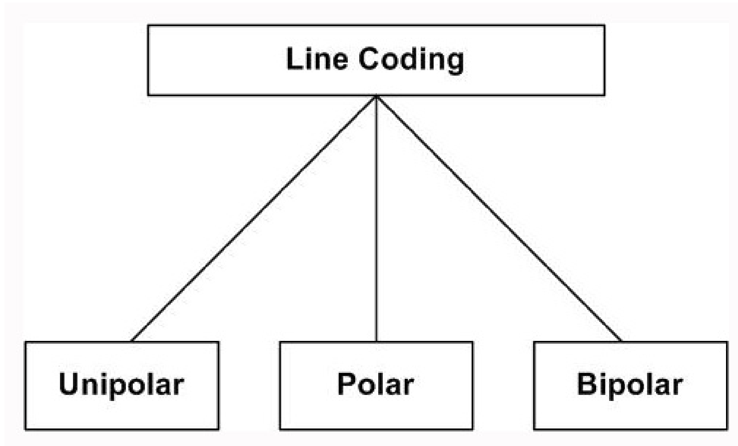


Figure 2.4.4 *Three basic categories of line coding techniques*

Unipolar: In unipolar encoding technique, only two voltage levels are used. It uses only one polarity of voltage level as shown in Fig. 2.4.5. In this encoding approach, the bit rate same as data rate. Unfortunately, DC component present in the encoded signal and there is loss of synchronization for long sequences of 0's and 1's. It is simple but obsolete.

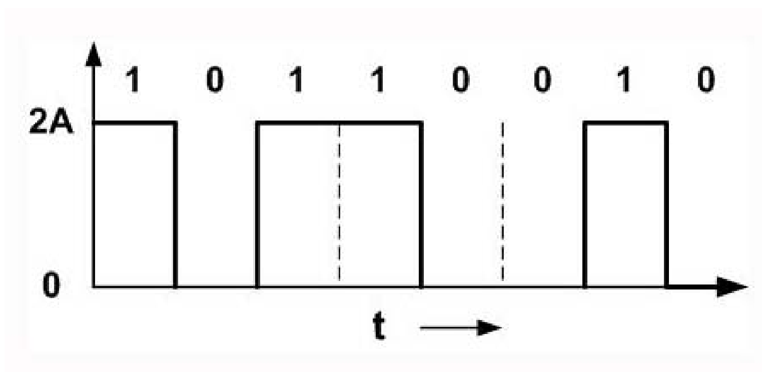


Figure 2.4.5 *Unipolar encoding with two voltage levels*

Polar: Polar encoding technique uses two voltage levels – one positive and the other one negative. Four different encoding schemes shown in Fig. 2.4.6 under this category discussed below:

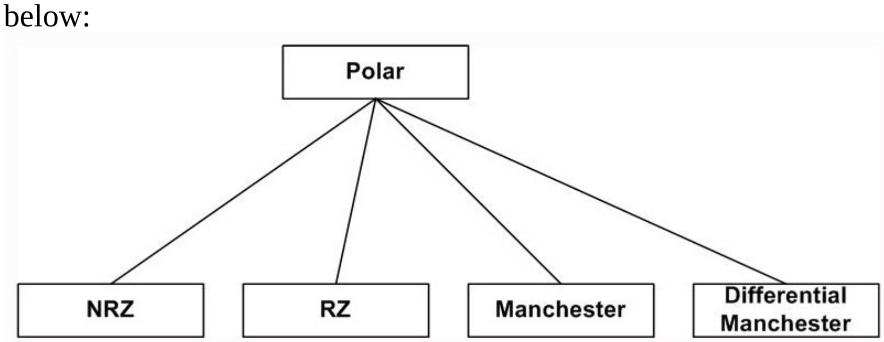
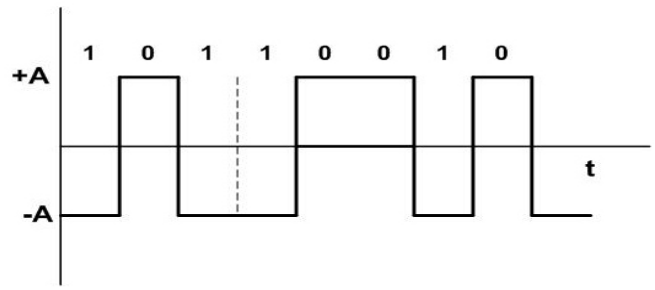
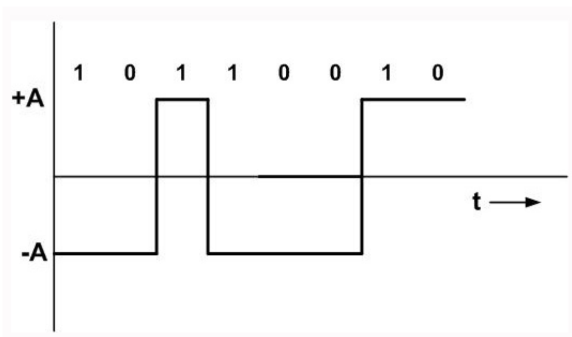


Figure 2.4.6 *Encoding Schemes under polar category*

Non Return to zero (NRZ): The most common and easiest way to transmit digital signals is to use two different voltage levels for the two binary digits. Usually a negative voltage is used to represent one binary value and a positive voltage to represent the other. The data is encoded as the presence or absence of a signal transition at the beginning of the bit time. As shown in the figure below, in NRZ encoding, the signal level remains same throughout the bit-period. There are two encoding schemes in NRZ: NRZ-L and NRZ-I, as shown in Fig. 2.4.7.



NRZ – L
1 = low level
0 = high level



NRZ – I

- ***For each 1 in the bit sequence, the signal level is inverted.***
- ***A transition from one voltage level to the other represents a 1.***

Figure 2.4.7 *NRZ encoding scheme*

The **advantages** of NRZ coding are:

- Detecting a transition in presence of noise is more reliable than to compare a value to a threshold.
- NRZ codes are easy to engineer and it makes efficient use of bandwidth.

The spectrum of the NRZ-L and NRZ-I signals are shown in Fig. 2.4.8. It may be noted that most of the energy is concentrated between 0 and half the bit rate. The main limitations are the presence of a dc component and the lack of synchronization capability. When there is long sequence of 0’s or 1’s, the receiving side will fail to regenerate the clock and synchronization between the transmitter and receiver clocks will fail.

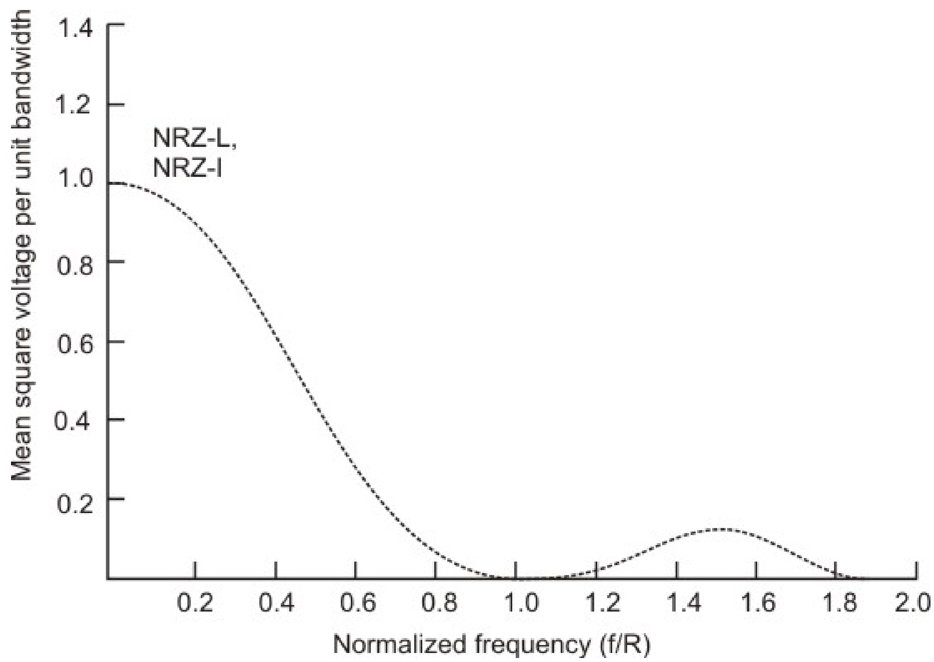


Figure 2.4.8 Signal spectrum of NRZ signals

Return to Zero RZ: To ensure synchronization, there must be a signal transition in each bit as shown in Fig. 2.4.9. Key characteristics of the RZ coding are:

- Three levels
- Bit rate is double than that of data rate
- No dc component
- Good synchronization
- Main limitation is the increase in bandwidth

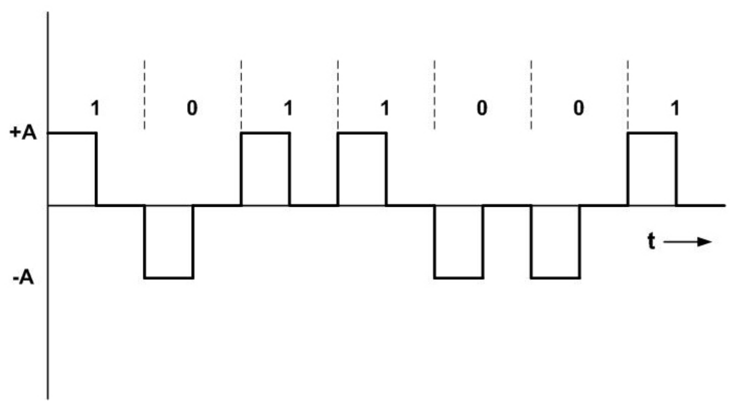


Figure 2.4.9 *RZ encoding technique*

Biphase: To overcome the limitations of NRZ encoding, biphase encoding techniques can be adopted. *Manchester* and *differential Manchester Coding* are the two common Biphase techniques in use, as shown in Fig. 2.4.10. In Manchester coding the mid-bit transition serves as a clocking mechanism and also as data.

In the standard Manchester coding there is a transition at the middle of each bit period. A binary 1 corresponds to a *low-to-high transition* and a binary 0 to a *high-to-low transition* in the middle.

In Differential Manchester, inversion in the middle of each bit is used for synchronization. The encoding of a 0 is represented by the presence of a transition both at the beginning and at the middle and 1 is represented by a transition only in the middle of the bit period.

Key characteristics are:

- Two levels
- No DC component
- Good synchronization
- Higher bandwidth due to doubling of bit rate with respect to data rate

The bandwidth required for biphase techniques are greater than that of NRZ techniques, but due to the predictable transition during each bit time, the receiver can synchronize properly on that transition. Biphase encoded signals have no DC components as shown in Fig. 2.4.11. A Manchester code is now very popular and has been specified for the IEEE 802.3 standard for base band coaxial cables and twisted pair CSMA/CD bus LANs.

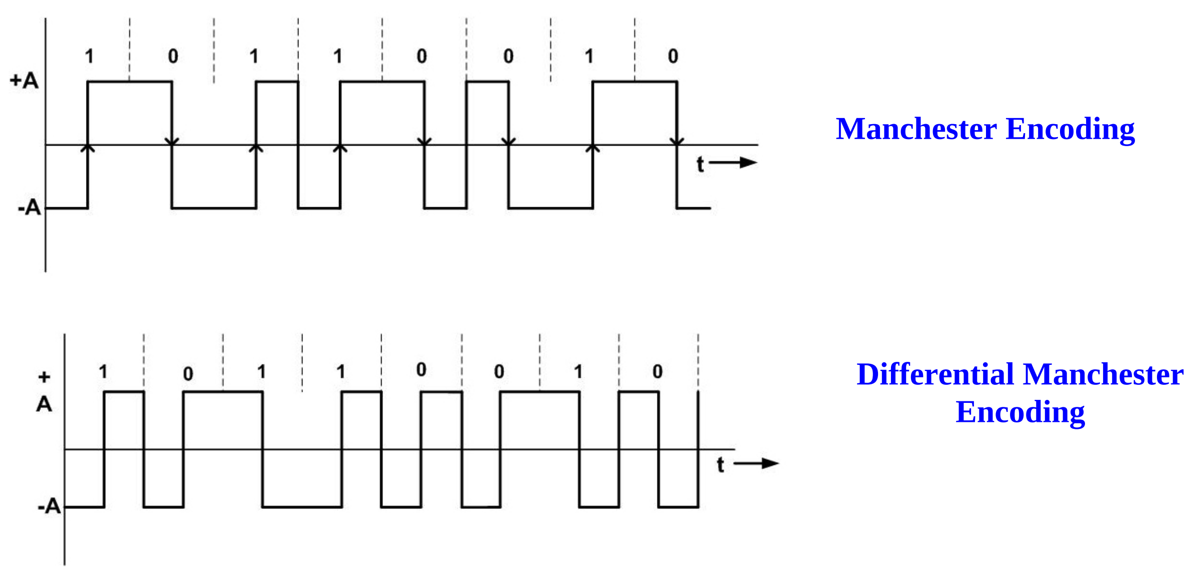


Figure 2.4.10 *Manchester encoding schemes*

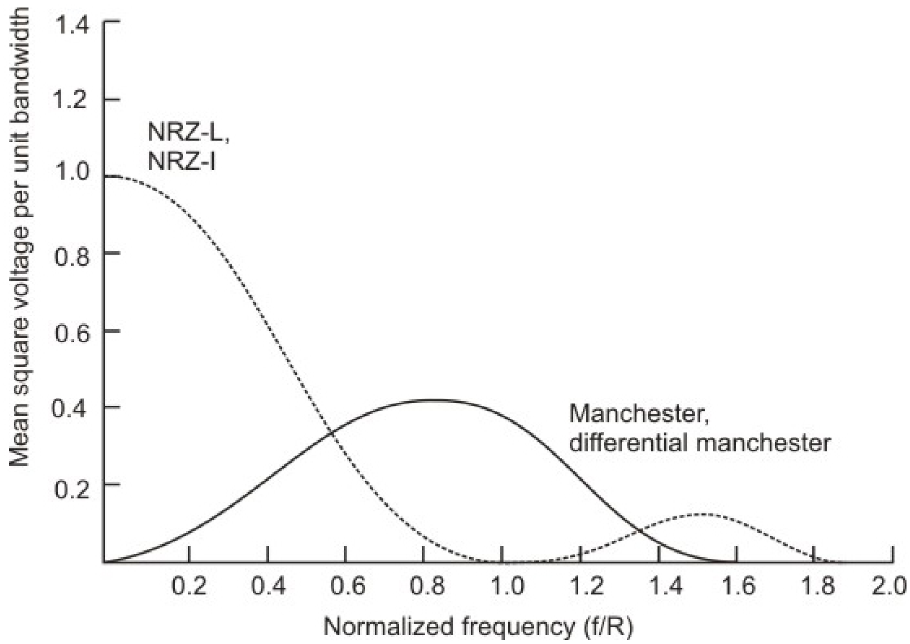


Figure 2.4.11 *Frequency spectrum of the Manchester encoding techniques*

Bipolar Encoding: Bipolar AMI uses three voltage levels. Unlike RZ, the zero level is used to represent a 0 and a binary 1's are represented by alternating positive and negative voltages, as shown in Fig 2.4.12.

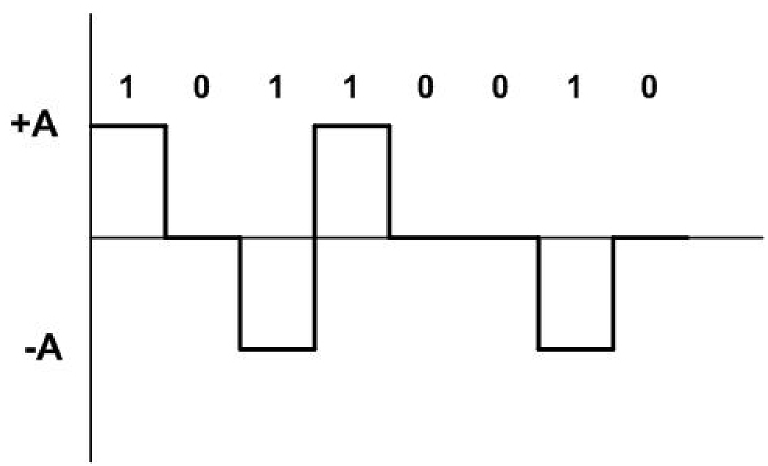


Figure 2. 4.12 *Bipolar AMI signal*

Pseudoternary: *This encoding scheme is same as AMI, but alternating positive and negative pulses occur for binary 0 instead of binary 1. Key characteristics are:*

- Three levels
- No DC component
- Loss of synchronization for long sequences of 0's
- Lesser bandwidth

Modulation Rate: Data rate is expressed in bits per second. On the other hand, modulation rate is expressed in bauds. General relationship between the two are given below:

$$D = R / b = R / \log_2 L$$

Where, D is the modulation rate in bauds, R is the data rate in bps, L is the number of different signal elements and b is the number of bits per signal element. Modulation rate for different encoding techniques is shown in Fig. 2.4.13.

Encoding Technique	Minimum	101010 . . .	Maximum
NRZ-L	0	1.0	1.0
NRZ-I	0	0.5	1.0
BIPOLAR-AMI	0	1.0	1.0
Manchester	1.O	1.0	2.0
Differential Manchester	1.0	1.5	2.0

Figure 2.4.13 *Modulation rate for different encoding techniques*

Frequency spectrum of different encoding schemes have been compared in Fig. 2.4.14.

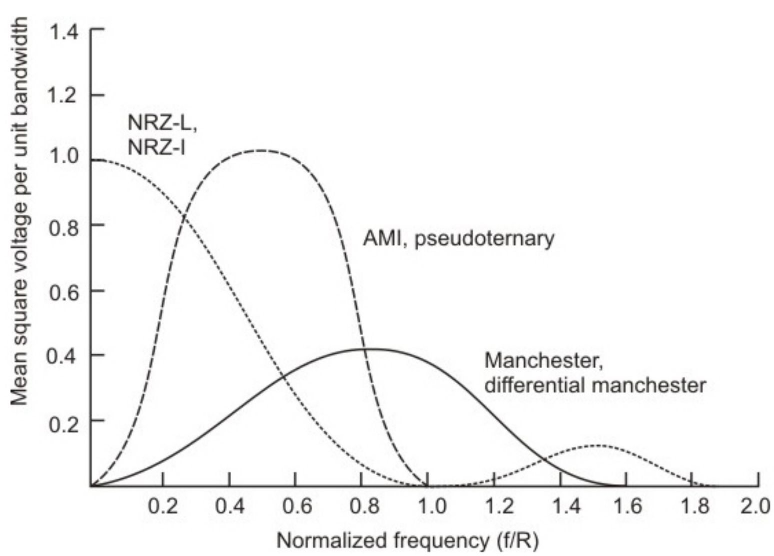


Figure 2.4.14 *Frequency spectrum of different encoding schemes*

Scrambling Schemes: *Extension of Bipolar AMI. Used in case of long distance applications. Goals:*

- No dc component
- No long sequences of 0-level line signal
- No increase in bandwidth
- Error detection capability
- Examples: B8ZS, HBD3

Bipolar with 8-zero substitution (B8ZS): *The limitation of bipolar AMI is overcome in B8ZS, which is used in North America. A sequence of eight zero's is replaced by the following encoding*

A sequence of eight 0's is replaced by 000+ - 0 + -, if the previous pulse was positive**.**
A sequence of eight 0's is replaced by 000 - + 0 + -, if the previous pulse was negative

High Density Bipolar-3 Zeros: *Another alternative, which is used in Europe and Japan is HDB3. It replaces a sequence of 4 zeros by a code as per the rule given in the following table. The encoded signals are shown in Fig. 2.4.15.*

HDB3 substitution rule		
Polarity of the Preceding pulse	**Number of bipolar pulses (ones) since last substitution**	
	odd	**even**
−	**000 −**	**+ 00 +**
+	**000 +**	**− 00 −**

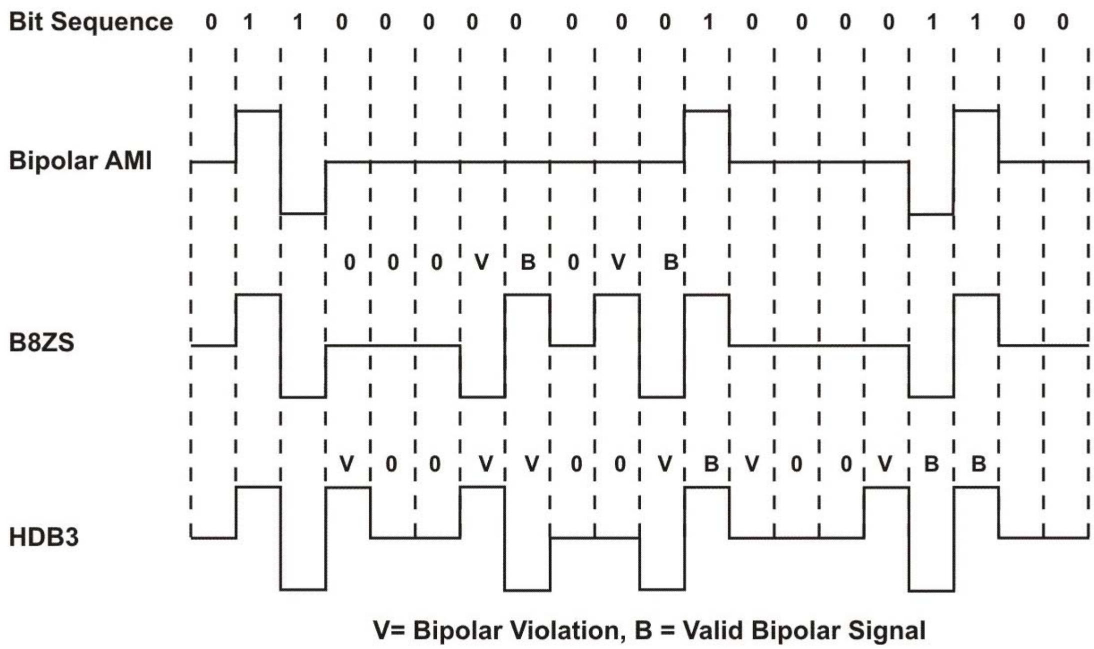


Figure 2.4.15 *B8ZS and HDB3 encoding techniques*

2.4.4 Analog Data, Digital Signals

Analog data such as voice, video and music can be converted into digital signal communication through transmission media. This allows the use of modern digital transmission and switching equipment's. The device used for conversion of analog data to digital signal and vice versa is called a *coder* (coder-decoder). There are two basic approaches:

- Pulse Code Modulation (PCM)
- Delta Modulation (DM)

2.4.4.1 Pulse Code modulation

Pulse Code Modulation involves the following three basic steps as shown in Fig. 2.4.16:

- Sampling – PAM
- Quantization
- Line coding

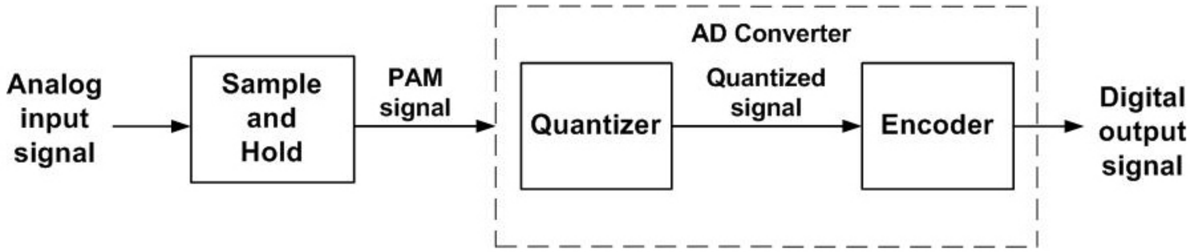


Figure 2.4.16 *Basic steps of pulse code modulation*

Sampling: This process is based on Shannon's sampling theorem. Numbers of samples of the signal are taken at regular intervals, at a rate higher than twice the highest significant signal frequency. This basic step is known as Pulse Amplitude Modulation (PAM) as shown in Fig. 2.4.17. For example, during the sampling of voice data, in the frequency range 300 to 4000 Hz, 8000 samples per second are sufficient for the coding.

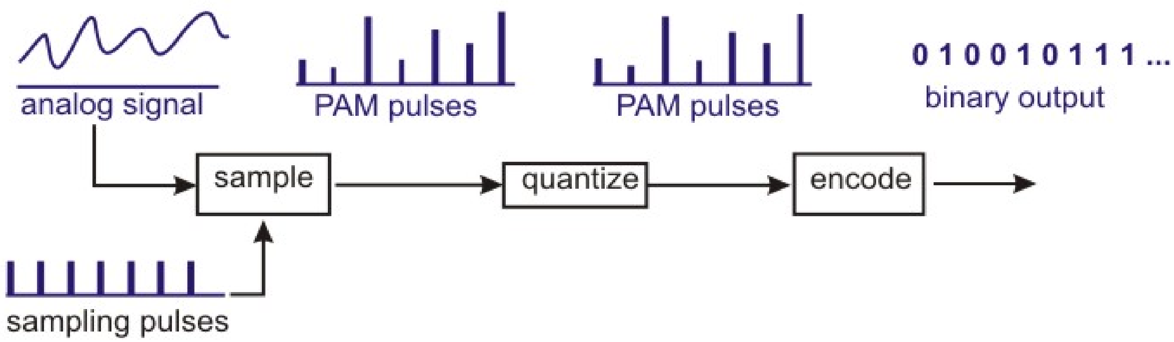


Figure 2.4.17 *Signal outputs after different steps of PCM*

Quantization: The PAM samples are quantized and approximated to n-bit integer by using analog-to-digital converter. For example, if n = 4, then there are 16 (=2^4) levels available for approximating the PAM signals. This process introduces an error are known as ***quantization* error**. Quantization error depends on step size. Use of uniform step size leads to poorer S/N ratio for small amplitude signals. With the constraint of a fixed number of levels, the situation can be improved using variable step size. The effect of quantization error can be minimized by using a technique known as **companding**. In this case, instead of using uniform stage sizes, the steps are close together at low signal amplitude and further apart at high signal amplitude as shown in Fig. 2.4.18. It uses a compressor before encoding and expander after decoding. This helps to improve the S/N ratio of the signal.

Line coding: The digital data thus obtained can be encoded into one of digital signals discussed earlier.

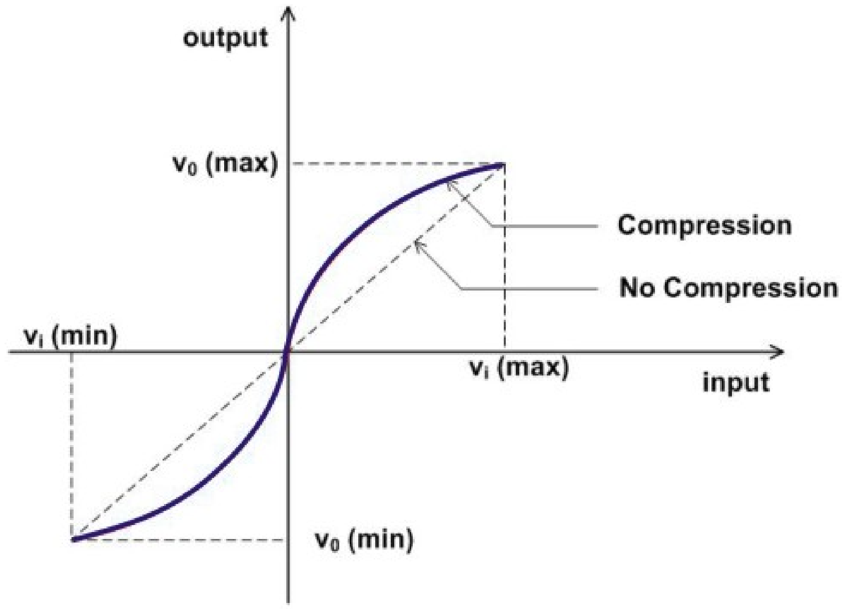


Figure 2.4.18 *The compander*

At the receiving end, an Digital-to-Analog converter followed by a low-pass filter can be used to get back the analog signal as shown in Fig. 2.4.19.

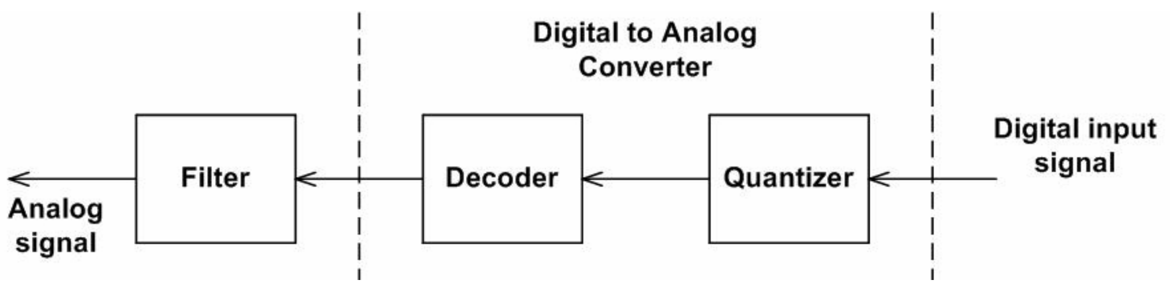


Figure 2.4.19 *Conversion of digital to analog signal*

Limitations: The PCM signal has high bandwidth. For example, let us consider voice signal as input with bandwidth of 4 kHz. Based on Nyquist theorem, the Sampling frequency should be 8 kHz. If an 8-bit ADC is used for conversion to digital data, it generates data rate of 64 Kbps. Therefore, to send voice signal a data rate of 64 Kbps is required. To overcome this problem a technique known as **Differential PCM** (DPCM) can be used. It is based on the observation that voice signal changes slowly. So, the difference between two consecutive sample values may be sent. Since the signal changes slowly, the difference between two consecutive sample values will be small and fewer number of bits can be used with consequent reduction in data rates.

Delta Modulation (DM)

Delta Modulation is a very popular alternative of PCM with much reduced complexity. Here the analog input is approximated by a staircase function, which moves up or down by one quantization level (a constant amount) at each sampling interval. Each sample delta modulation process can be represented by a single binary digit, which makes it more efficient than the PCM technique. In this modulation technique, instead of sending the entire encoding of each and every sample, we just send the change from previous sample. If the difference between analog input and the feedback signal is positive, then encoded output is 1, otherwise it is 0. So, only one bit is to be sent per sample. Figure 2.4.20 shows the Delta modulation operation.

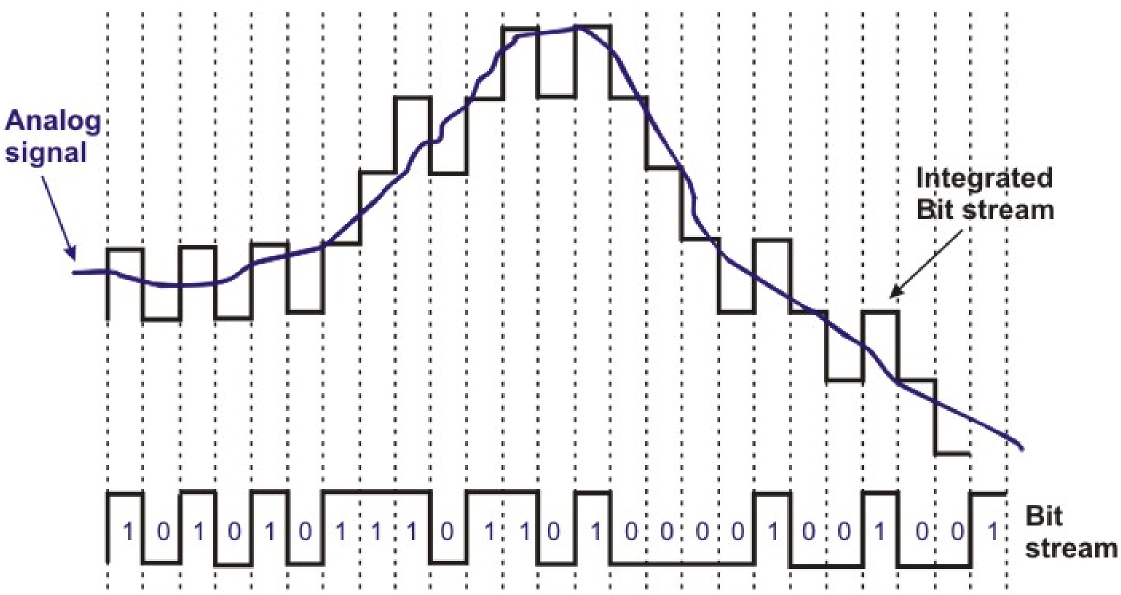


Figure 2.4.20 *Delta modulation*

Advantages: Main advantage of Delta Modulation is its simplicity of implementation as shown in Fig. 2.4.21. Each sample is represented by a single binary digit, which makes it more efficient than the PCM technique. Two important parameters:

- The size of the step
- The sampling rate

In the transmitting end, the analog input is compared to the most recent value of the approximating staircase function at each sampling time. If the value of the sampled waveform that of the staircase function, a 1 is generated; otherwise a 0 is generated as shown in Fig. 2.4.20. The output of the DM is a binary sequence that can be used to reconstruct the staircase function at the receiving end as shown in Fig. 2.4.21.

Disadvantages: Fixed step size leads to overloading. Overloading occurs not only due to higher voltage, but due to its slope as shown in Fig. 2.4.22. This problem can be overcome using adaptive delta modulation. The steps sizes are small, when the signal changes are small. The steps sizes are large, when the signal changes are large.

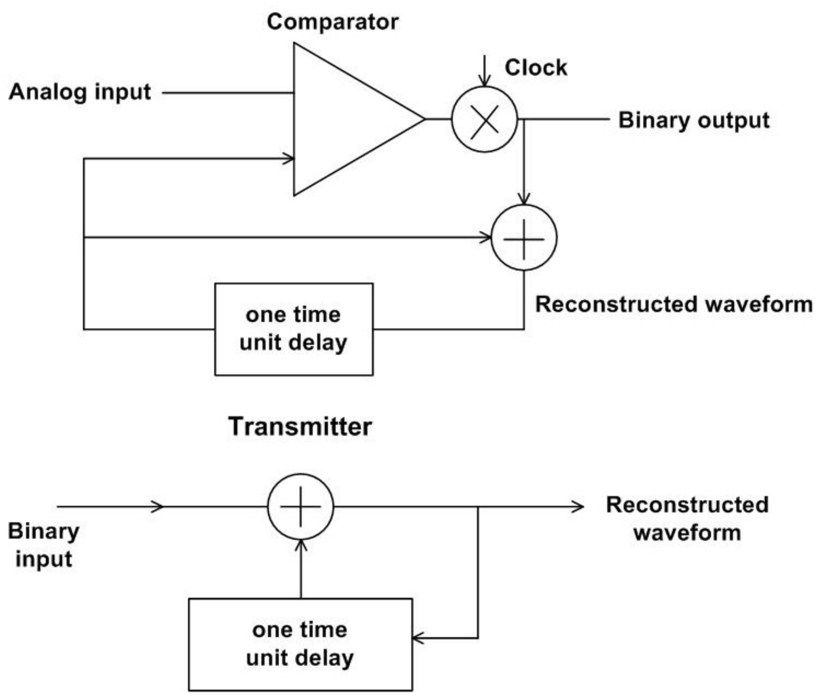


Figure 2.4.21 *Implementation of Delta modulation*

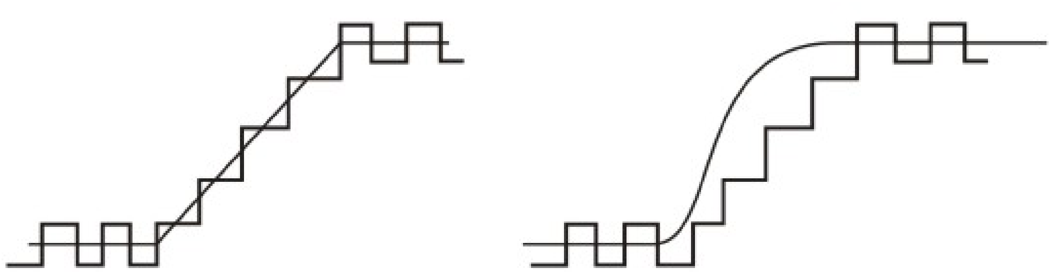


Figure 2.4.20 *Slope overloading*

Review Questions

1. Why do you need encoding of data before sending over a medium?
Ans: Suitable encoding of data is required in order to transmit signal with minimum attenuation and optimize the use of transmission media in terms of data rate and error rate.

2. What are the four possible encoding techniques? Give examples.
Ans: The four possible encoding techniques are

- Digital Data to Digital Signal; Example - Transmitter
- Analog Data to Digital Signal; Example - Codec (Coder-Decoder)
- Digital Data to Analog Signal; Example - Modem
- Analog Data to Digital Signal; Example - Telephone

3. Between RZ and NRZ encoding techniques, which requires higher bandwidth and why?
Ans: RZ encoding requires more bandwidth, as it requires two signal changes to encode one bit**.**

4. How does Manchester encoding differ from differential Manchester encoding?
Ans: In the Manchester encoding, a low-to-high transition represents a 1, and a high-to-low transition represents a 0. There is a transition at the middle of each bit period, which serves the purpose of synchronization and encoding of data.
In Differential Manchester, the encoding of a 0 is represented by the presence of a transition at the beginning of a bit period, and a 1 is represented by the absence of a transition at the beginning of a bit period. In this case, the midbit transition is only used for synchronization.

5. How Manchester encoding helps in achieving better synchronization?
Ans: In Manchester encoding, there is a transition in the middle of each bit period and the receiver can synchronize on that transition. Hence better synchronization is achieved.

6. Why B8ZS coding is preferred over Manchester encoding for long distance communication?
Ans: The B8ZS encoding is preferred over Manchester encoding, because B8ZS encoding requires lesser bandwidth than Manchester encoding.
2. Why is it necessary to limit the band of a signal before performing sampling?
Ans: It is necessary to limit the bandwidth of a signal before sampling so that the basic requirement of sampling theorem, i.e. the sampling rate should twice or more than twice the maximum frequency component of the signal, is satisfied. This is known as Nyquist rate. If it is violated, original signal cannot be recovered from the sampled signal.

7. Distinguish between PAM and PCM signals?
Ans: In order to convert Analog data to Digital signal, initially sampling is done on the analog data by using Sample & Hold (S/H) circuit. The output of the S/H circuit is known as PAM (Pulse Amplitude Modulated) signal. The PAM signal is then converted to PCM

(Pulse Code Modulated) data by using a Analog-to-Digital (A/D) converter circuit. This digital data (PCM) is passed through an encoder to generate PCM signal.

8. What is quantization error? How can it be reduced?

Ans: To convert analog signal to digital signal, the analog signal is first sampled and each of these analog samples must be assigned a binary code. In other words, each sample is approximated by being quantized into some binary codes. As the quantized values are only approximations, it is impossible to recover the original signal exactly. Error due to this quantization is known as quantization error. Quantization error can be minimized by using non-linear encoding.

9. Explain how and in what situation DPCM performs better than PCM

Ans: DPCM performs better when the input is slowly changing, as in case of a voice signal.