Data Communications and Networking Fourth Edition

Forouzan

Chapter 4

Digital Transmission

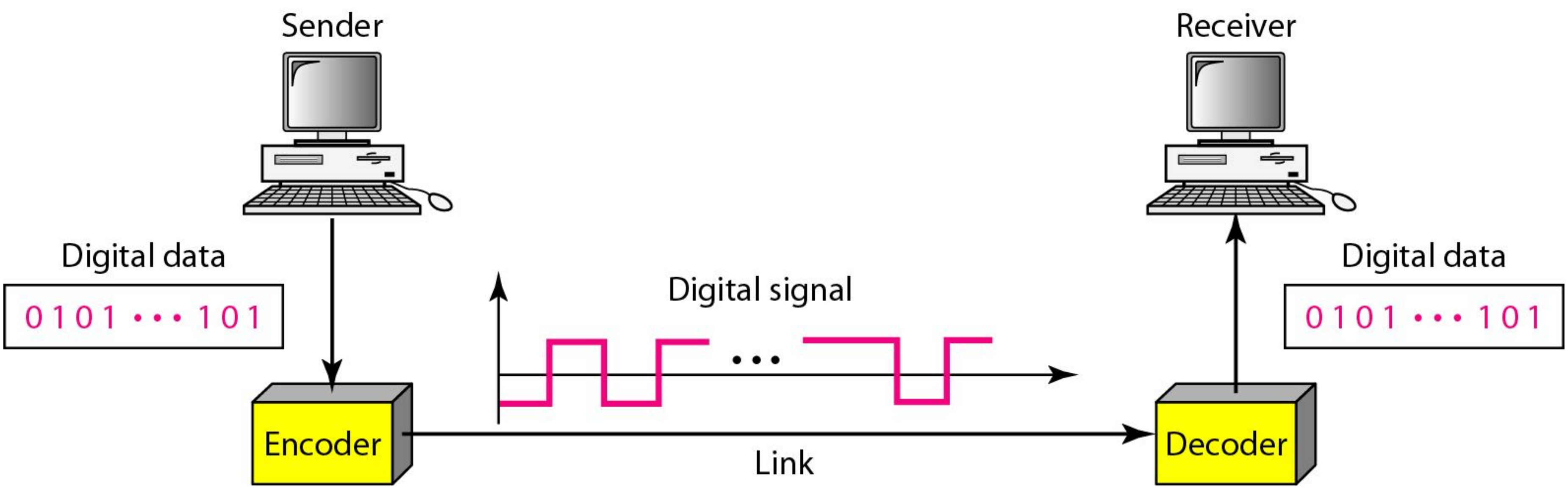
4-1 DIGITAL-TO-DIGITAL CONVERSION

In this section, we see how we can represent digital data by using digital signals. The conversion involves three techniques: line coding, block coding, and scrambling. Line coding is always needed; block coding and scrambling may or may not be needed.

Topics discussed in this section:

Line Coding
Line Coding Schemes (*Polar, Bipolar and Manchester coding*)
Block Coding
Scrambling

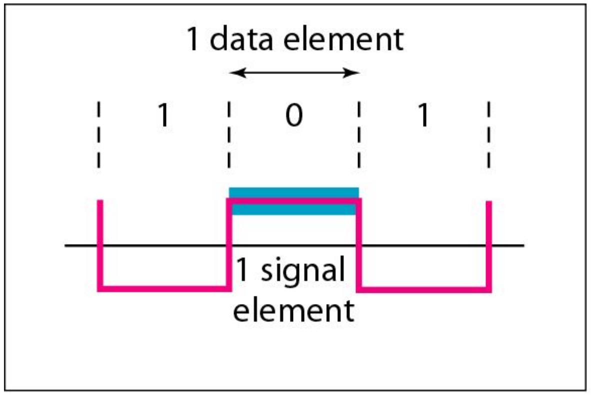
Figure 4.1 *Line coding and decoding*



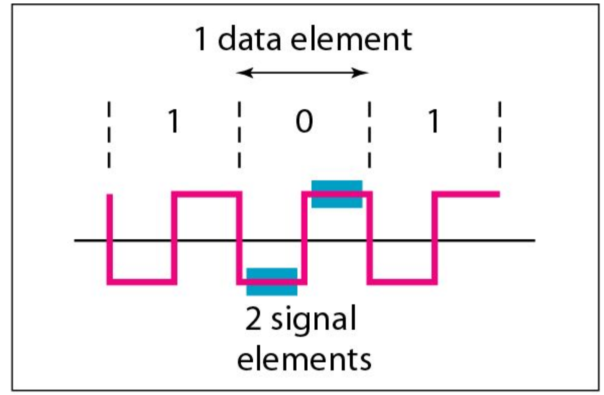
Line coding is the process of converting digital data to digital signals.

At the sender, digital data are encoded into a digital signal; at the receiver, the digital data are recreated by decoding the digital signal.

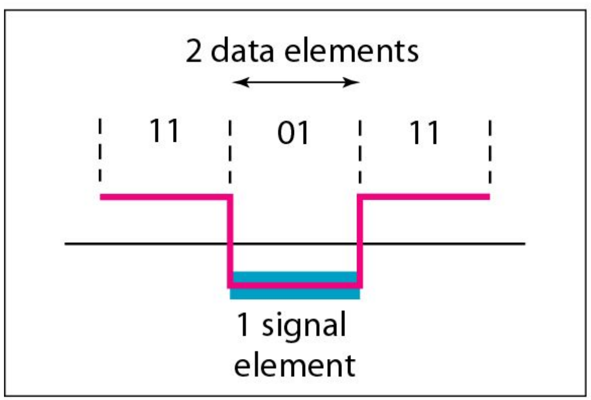
Figure 4.2 *Signal element versus data element*



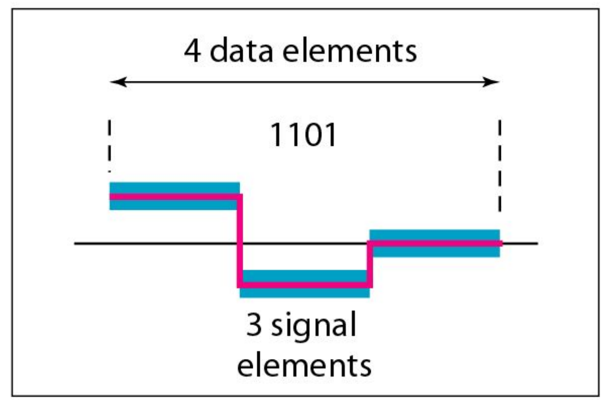
a. One data element per one signal element (r = 1)



b. One data element per two signal elements $\left(r = \frac{1}{2}\right)$



c. Two data elements per one signal element (r = 2)



d. Four data elements per three signal elements $\left(r = \frac{4}{3}\right)$

Cases of Live Example

- Suppose each data element is a person who needs to be carried from one place to another.
- We can think of a signal element as a vehicle that can carry people.
- When $r = 1$, it means each person is driving a vehicle.
- When $r > 1$, it means more than one person is travelling in a vehicle (a carpool, for example).
- We can also have the case where one person is driving a car and a trailer ($r = 1/2$).

Data Rate Versus Signal Rate

- The **data rate** defines the number of data elements (bits) sent in 1s. The unit is bits per second (bps).
- The **signal rate** is the number of signal elements sent in 1s. The unit isthe baud.
- There are several common terminologies used in the literature.
- The data rate is sometimes called the **bit rate;** the signal rate is sometimes called the **pulse rate,** the **modulation rate,** or the **baud rate.**
- One goal: To increase the data rate while decreasing the signal rate.
- Increasing the data rate increases the speed of transmission; decreasing the signal rate decreases the bandwidth requirement.

Relationship between data rate (N) and signal rate (S)

$$S = N/r \qquad\qquad S_{average} = c \times N \times (1/r) \text{ baud}$$

- Where, a ratio r which is the number of data elements carried by each signal element.
- where N is the data rate (bps);
- c is the case factor, which varies for each case;
- S is the number of signal elements per second

Example 4.1

A signal is carrying data in which one data element is encoded as one signal element (r = 1). If the bit rate is 100 kbps, what is the average value of the baud rate if c is between 0 and 1?

Solution

We assume that the average value of c is 1/2 . The baud rate is then

$$S = c \times N \times \frac{1}{r} = \frac{1}{2} \times 100{,}000 \times \frac{1}{1} = 50{,}000 = 50 \text{ kbaud}$$

Note

Although the actual bandwidth of a digital signal is infinite, the effective bandwidth is finite.

Example 4.2

The maximum data rate of a channel is $N_{max} = 2 \times B \times \log_2 L$ (defined by the Nyquist formula). Does this agree with the previous formula for N_{max}?

Solution

A signal with L levels actually can carry $\log_2 L$ bits per level. If each level corresponds to one signal element and we assume the average case ($c = 1/2$), then we have

$$N_{\max} = \frac{1}{c} \times B \times r = 2 \times B \times \log_2 L$$

Definitions

- In decoding a digital signal, the receiver calculates a running average of the received signal power. This average is called the ***baseline*.**
- A long string of 0s or 1s can cause a drift in the baseline (**baseline wandering**) and make it difficult for the receiver to decode correctly.
- A good line coding scheme needs to prevent baseline wandering.

Definitions

- When the voltage level in a digital signal is constant for a while, the spectrum creates very low frequencies.
- These frequencies are around zero, called DC (direct-current) *components,* present problems for a system that cannot pass low frequencies or a system that uses electrical coupling (via a transformer).
- DC component means 0/1 parity that can cause base-line wondering.
- For example, a telephone line cannot pass frequencies below 200 Hz. Also a long-distance link may use one or more transformers to isolate different parts of the line electrically. For these systems, we need a scheme with no **DC component.**

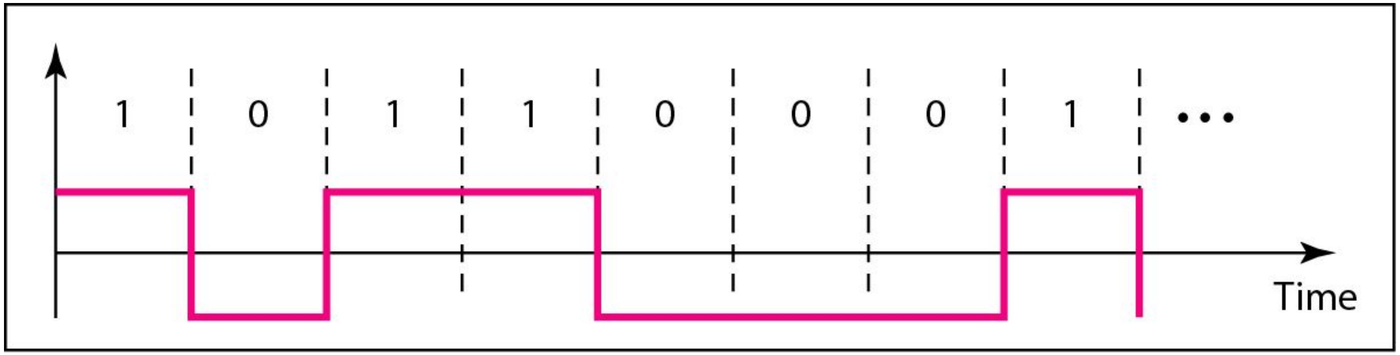
Synchronization

- To correctly interpret the signals received from the sender, the receiver's bit **intervals** must correspond exactly to the sender's bit intervals.
- If the receiver clock is **faster or slower**, the bit intervals are not matched and the receiver might misinterpret the signals.
- Figure 4.3 (next slide) shows a situation in which the receiver has a shorter bit duration.
- The sender sends 10110001, while the receiver receives 110111000011.
- A **self-synchronizing** digital signal includes timing information in the data being transmitted.
- This can be achieved if there are transitions in the signal that alert the receiver to the beginning, middle, or end of the pulse.
- If the receiver's clock is out of synchronization, these points can reset the clock.

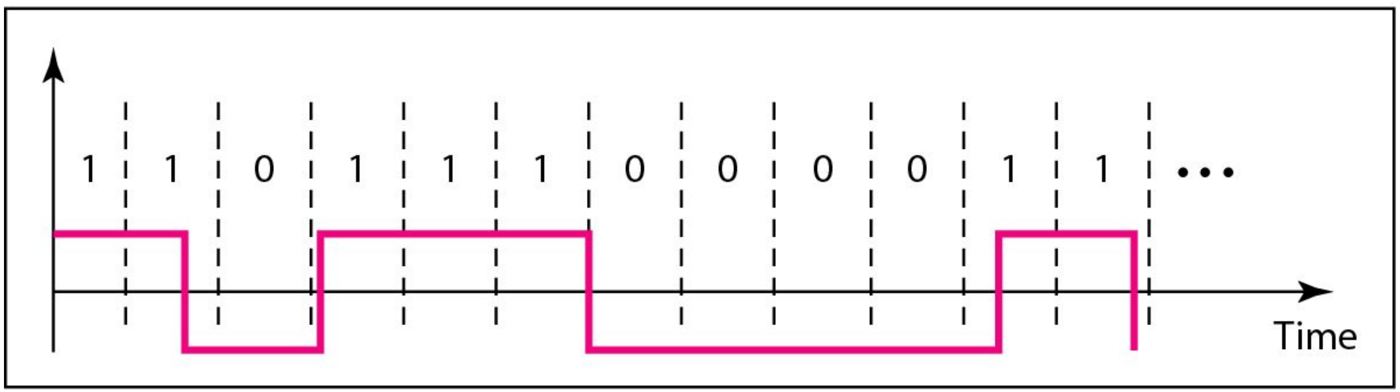
Definitions

- ***Built-in Error Detection***
  It is desirable to have a built-in error-detecting capability in the generated code to detect some or all of the errors that occurred during transmission. Some encoding schemes that we will discuss have this capability to some extent.
- ***Immunity to Noise and Interference***
  Another desirable code characteristic is a code that is immune to noise and other interferences. Some encoding schemes that we will discuss have this capability.
- ***Complexity***
  A complex scheme is more costly to implement than a simple one. For example, a scheme that uses four signal levels is more difficult to interpret than one that uses only two levels.

Figure 4.3 *Effect of lack of synchronization*



a. Sent



b. Received

Example 4.3

In a digital transmission, the receiver clock is 0.1 percent faster than the sender clock. How many extra bits per second does the receiver receive if the data rate is 1 kbps? How many if the data rate is 1 Mbps?

Solution

At 1 kbps, the receiver receives 1001 bps instead of 1000 bps.

1000 bits sent	1001 bits received	1 extra bps

At 1 Mbps, the receiver receives 1,001,000 bps instead of 1,000,000 bps.

1,000,000 bits sent	1,001,000 bits received	1000 extra bps

Figure 4.4 *Line coding schemes*

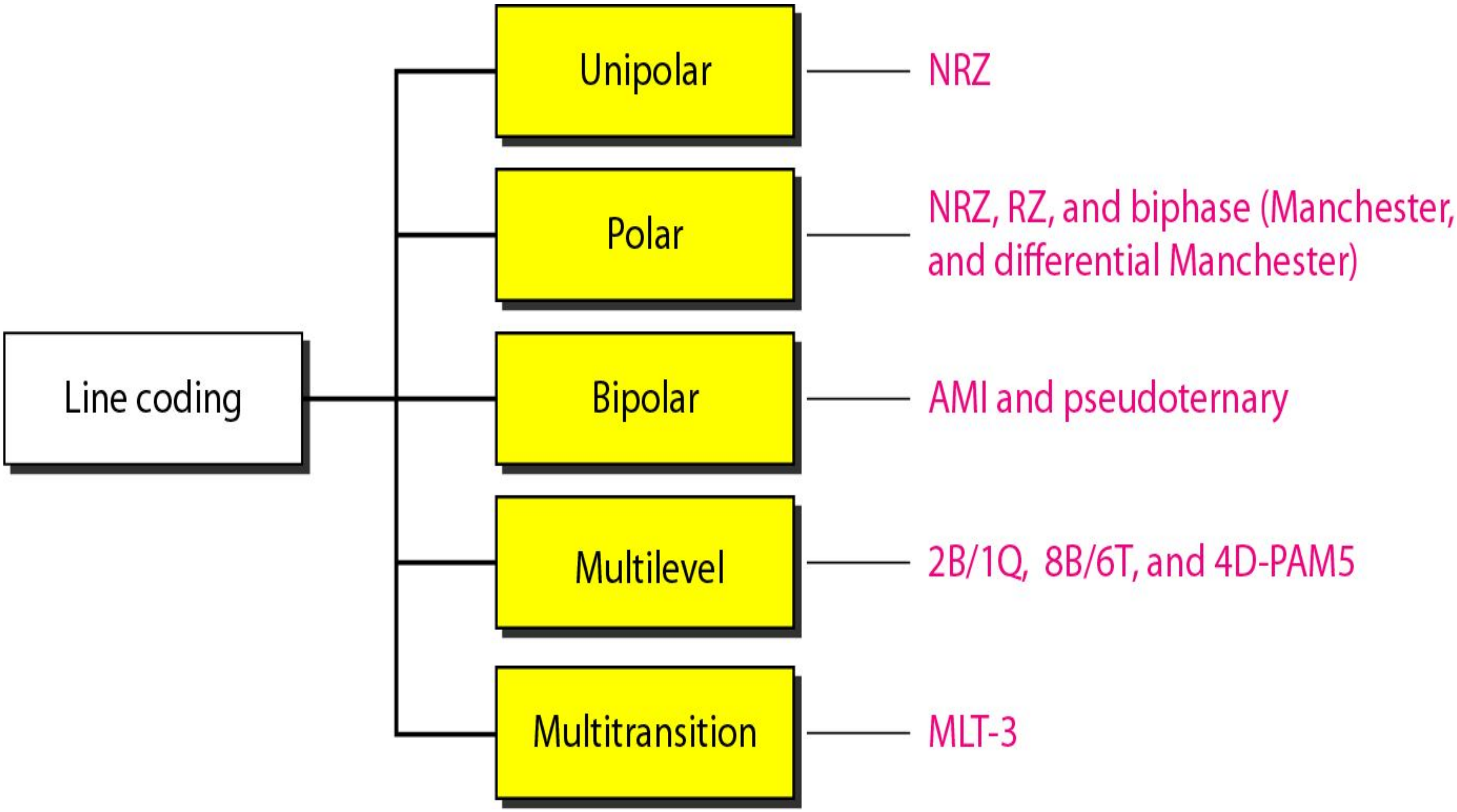
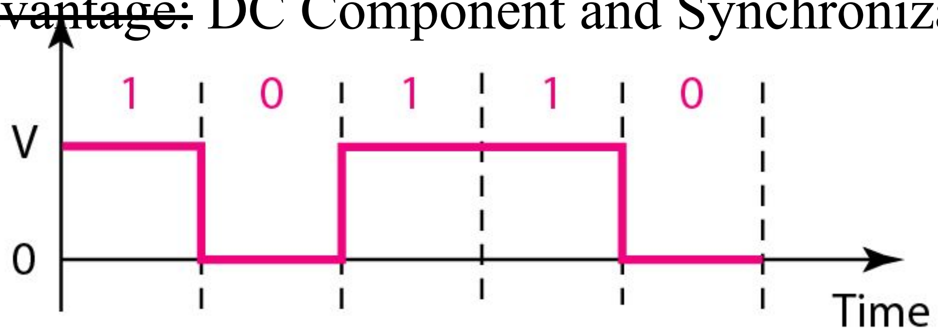


Figure 4.5 *Unipolar NRZ scheme*

In a unipolar scheme, all the signal levels are on one side of the time axis, either above or below. In ***Non-Return-to-Zero,*** the signal does not return to zero at the middle of the bit, *where* positive voltage defines bit 1 and the zero voltage defines bit 0. Costly. the normalized power (the power needed to send 1 bit per unit line resistance) is double that for polar NRZ.

~~Disadvantage:~~ DC Component and Synchronization.

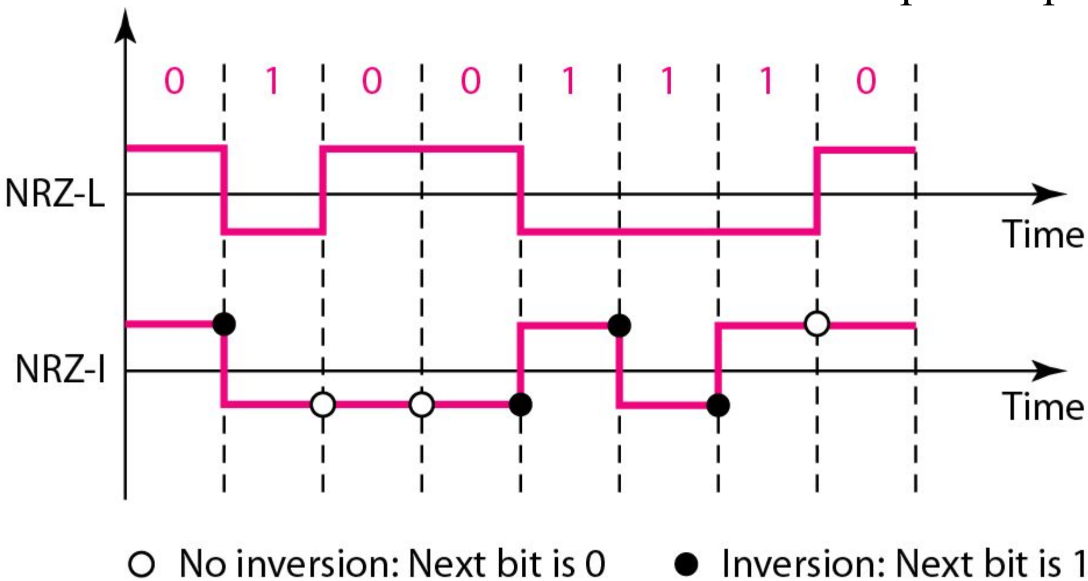


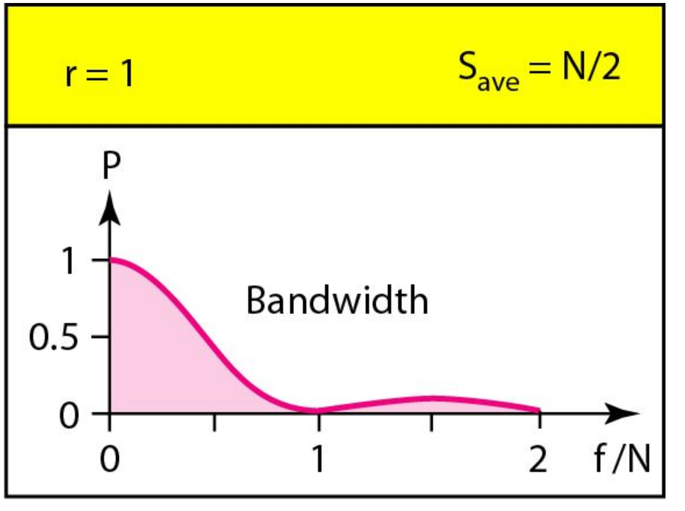
$$\frac{1}{2}V^2 + \frac{1}{2}(0)^2 = \frac{1}{2}V^2$$

Normalized power

Figure 4.6 *Polar NRZ-L and NRZ-I schemes*

Non-Return-to-Zero (NRZ) with L (Level) and I (Invert). In NRZ-L the level of the voltage determines the value of the bit. In NRZ-I the inversion or the lack of inversion determines the value of the bit. If there is a long sequence of 0s or 1s in NRZ-L, the average signal power becomes skewed. In NRZ-I this problem occurs only for a long sequence of 0s. The synchronization problem. Another problem with NRZ-L occurs when there is a sudden change of polarity in the system. NRZ-L and NRZ-I both have an average signal rate of N/2 Bd. NRZ-L and NRZ-I both have a DC component problem.





Example 4.4

A system is using NRZ-I to transfer 1Mbps data. What are the average signal rate and minimum bandwidth?

Solution

The average signal rate is $S = N/2 = 500$ kbaud. The minimum bandwidth for this average baud rate is $B_{min} = S = 500$ kHz.

Figure 4.7 *Polar RZ scheme*

- ***Return-to-Zero (RZ)*** uses three values: positive, negative, and zero.
- Signal changes not between bits but during the bit.
- Occupy greater bandwidth as needs change during the bits.
- No DC component problem.
- Another problem is the complexity due to 3 signals.
- Not in use.

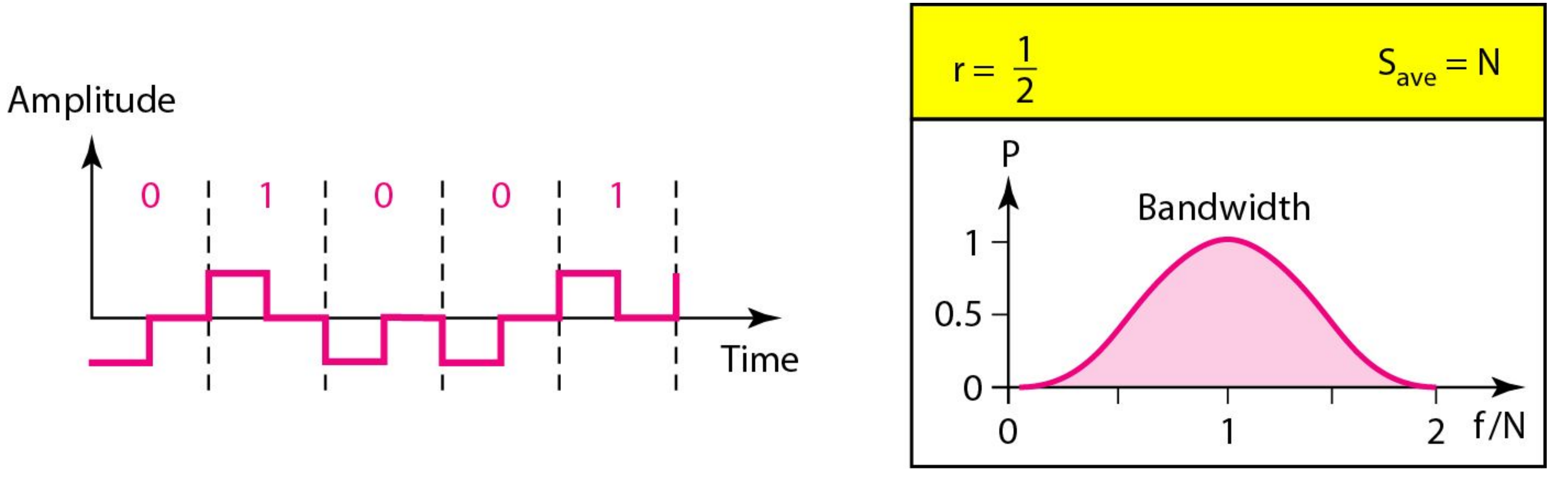
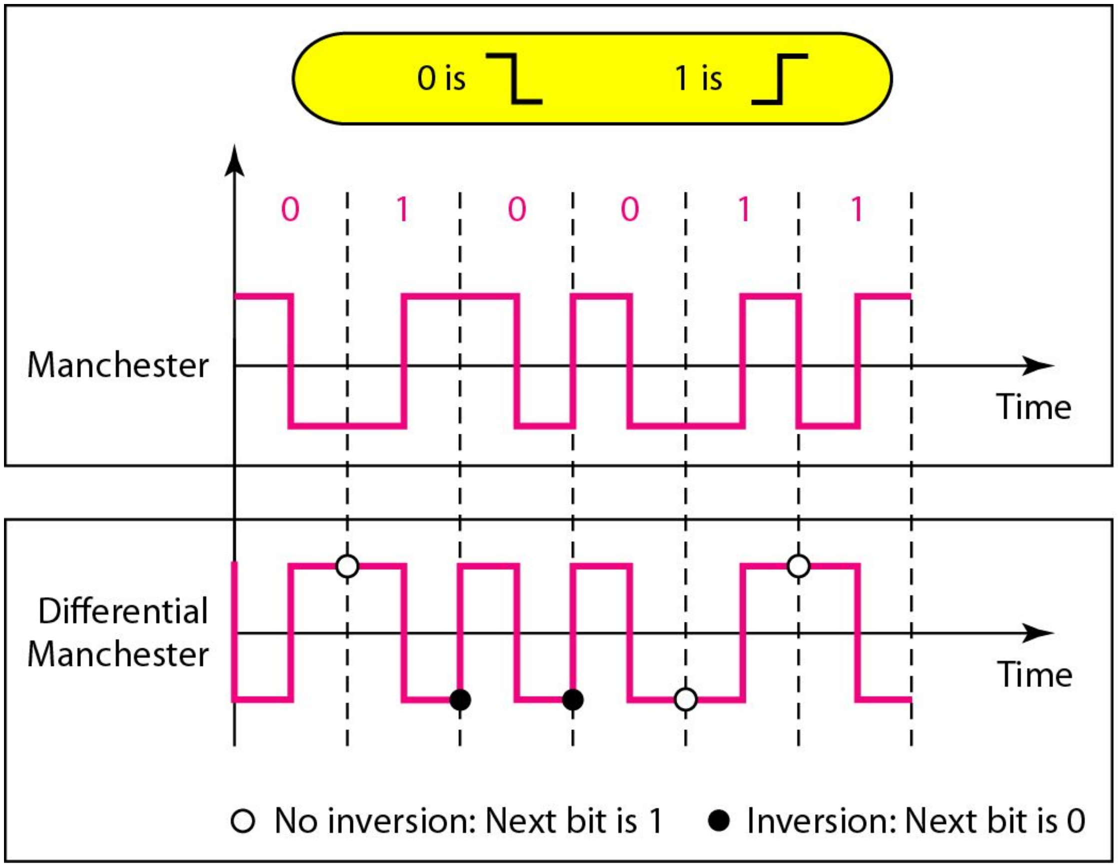
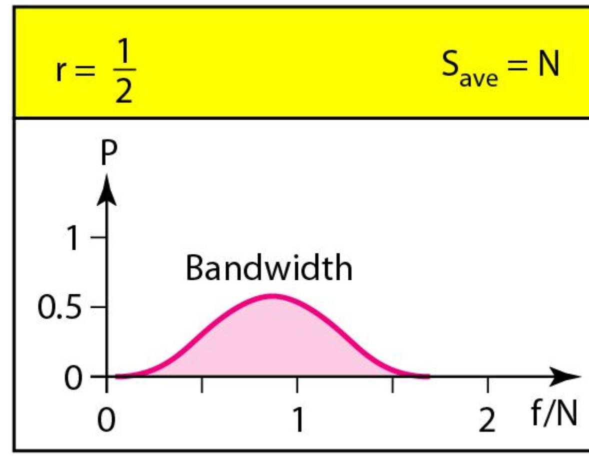


Figure 4.8 *Polar biphase: Manchester and differential Manchester schemes*





Note

In Manchester and differential Manchester encoding, the transition at the middle of the bit is used for synchronization.

Note

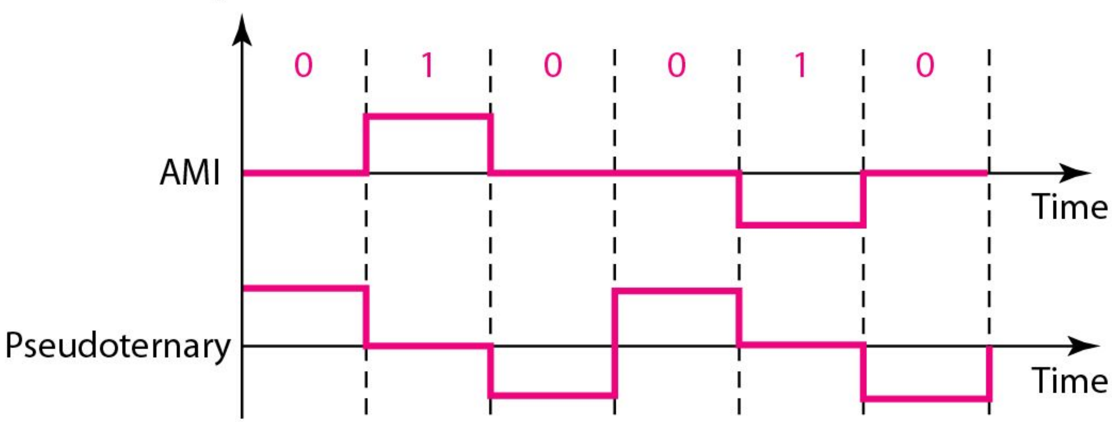
The minimum bandwidth of Manchester and differential Manchester is 2 times that of NRZ.

Note

In bipolar encoding, we use three levels: positive, zero, and negative.

Figure 4.9 *Bipolar schemes: AMI and pseudoternary*

- **Alternate Mark Inversion (AMI)** and **Pseudoternary.**
- Mark means 1. So AMI means alternate 1 inversion.
- A neutral zero voltage represents binary 0. Binary 1s are represented by alternating positive and negative voltages.
- A variation of AMI encoding is called pseudoternary in which the 1 bit is encoded as a zero voltage and the 0 bit is encoded as alternating positive and negative voltages.
- Same signal rate as NRZ, but there is no DC component.



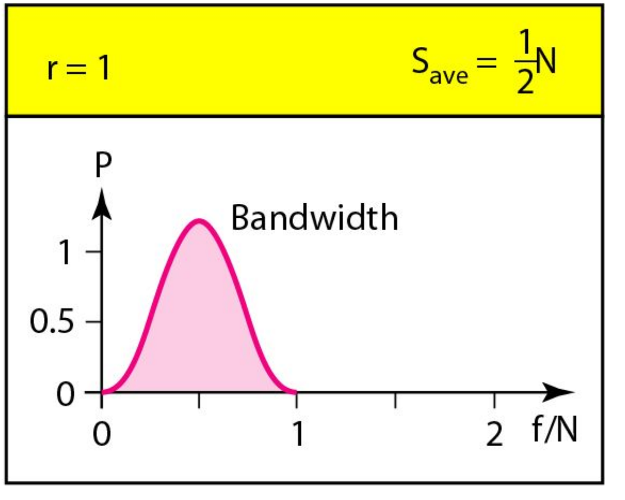
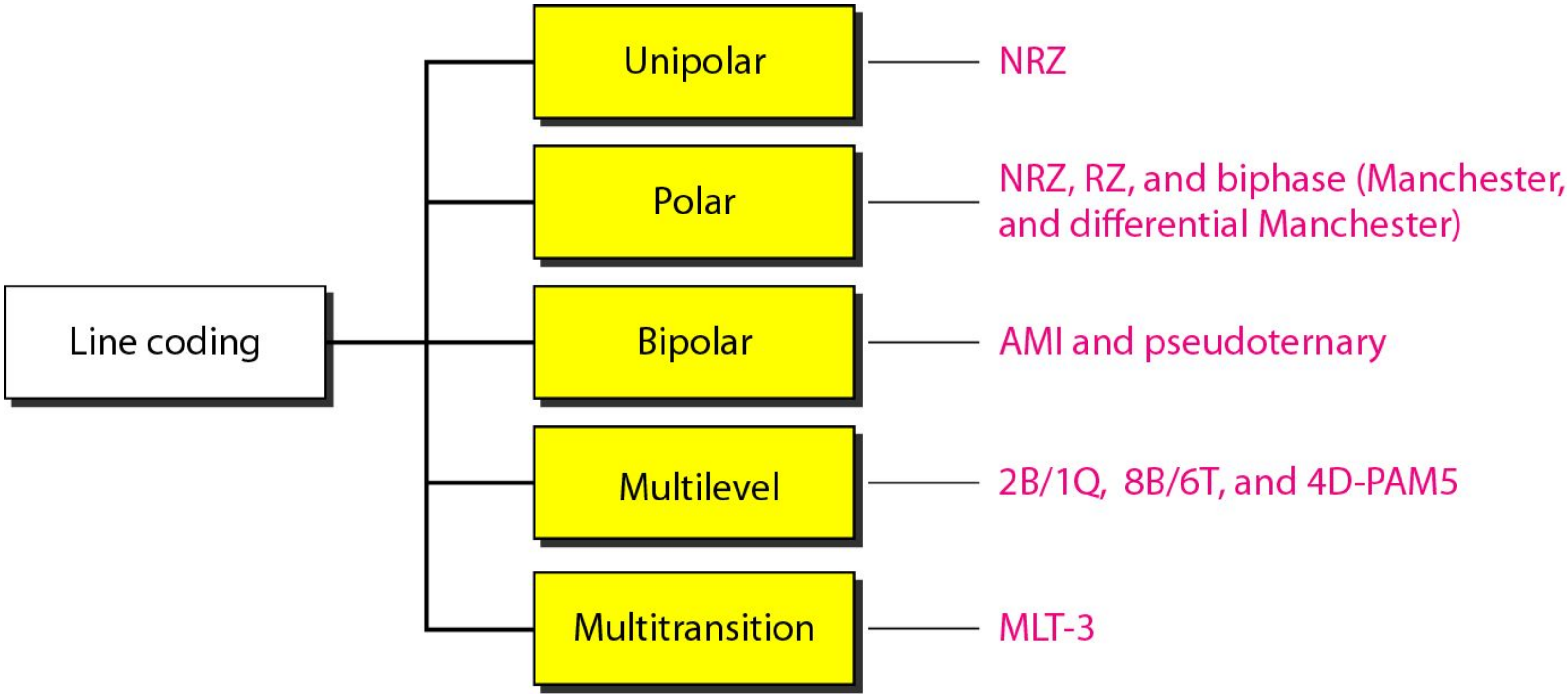


Figure 4.4 *Line coding schemes*



Note

In *m*B*n*L schemes, a pattern of *m* data elements is encoded as a pattern of *n* signal elements in which $2^m \le L^n$.

Figure 4.10 *Multilevel: 2B1Q scheme*

- The first *mBnL* scheme we discuss, **two binary, one quaternary (2B1Q),** uses data patterns of size 2 and encodes the 2-bit patterns as one signal element belonging to a four-level signal.
- In this type of encoding $m = 2$, $n = 1$, and $L = 4$ (quaternary).
- 2 times faster than by using NRZ-L
- There are no redundant signal patterns in this scheme because $2^2 = 4^1$.
- Used in DSL (Digital Subscriber Line) technology to provide a high-speed connection to the Internet by using subscriber telephone lines

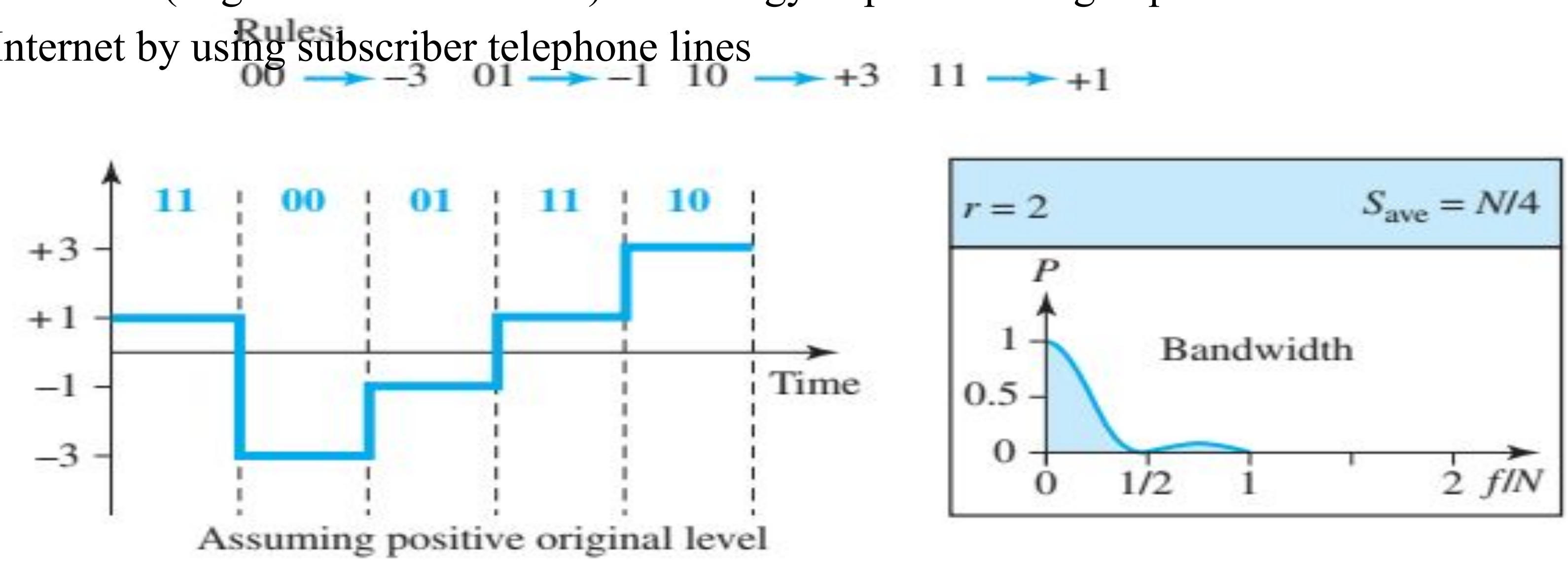
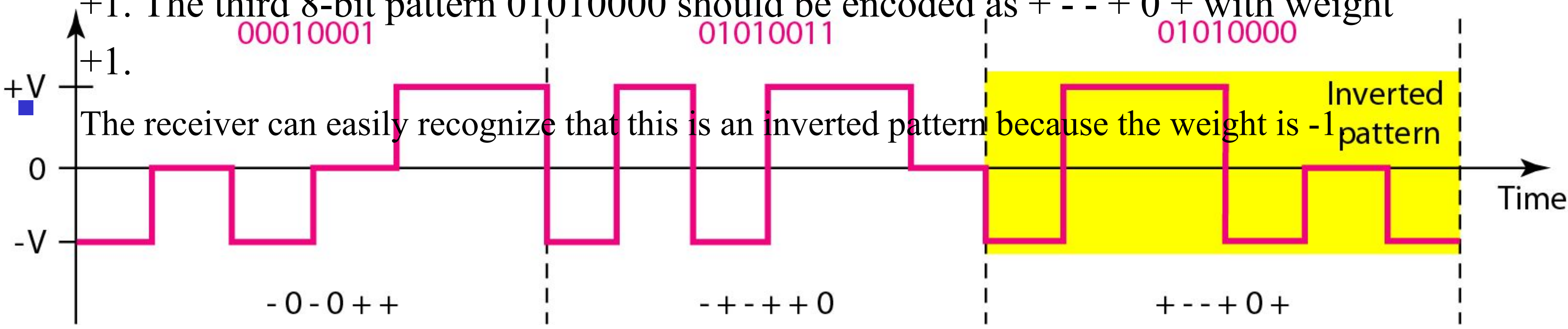


Figure 4.11 ***Multilevel: 8B6T scheme***

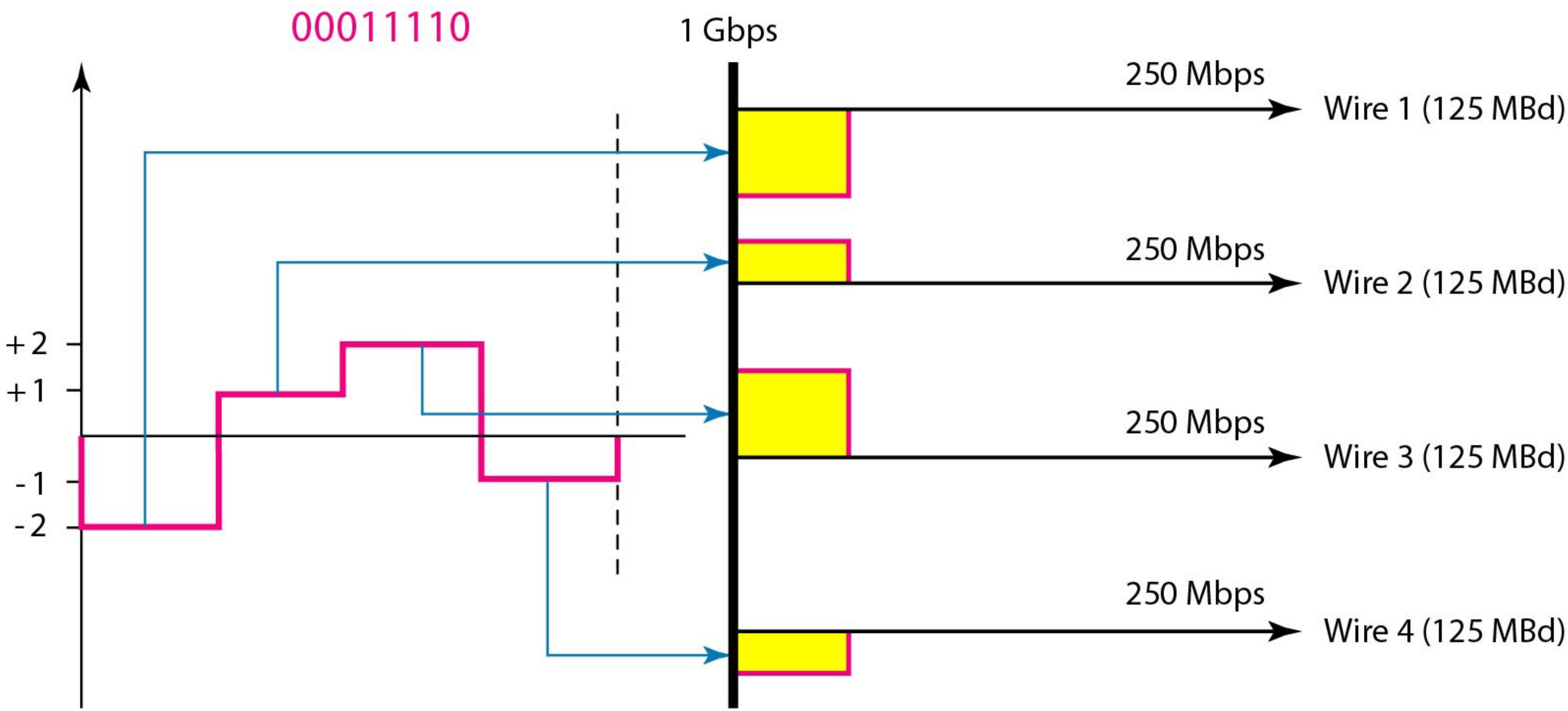
- The **eight binary, six ternary (8B6T)** is used with 100BASE-4T cable.
- Signal has three levels (ternary) 2^8 = 256 different data patterns and 3^6 = 729 different signal patterns.
- There are 729 - 256 = 473 redundant signal elements that provide synchronization, error detection and provide DC balance.
- The first 8-bit pattern 00010001 is encoded as the signal pattern - 0 - 0 + + with weight 0; the second 8-bit pattern 01010011 is encoded as - + - + + 0 with weight +1. The third 8-bit pattern 01010000 should be encoded as + - - + 0 + with weight +1.
- The receiver can easily recognize that this is an inverted pattern because the weight is -1



4D-PAM5

- Four-dimensional five level pulse amplitude modulation (4D-PAM5)
- The 4D means that data is sent over four wires at the same time. It uses five voltage levels, such as -2, -1, 0, 1, and 2.
- However, one level, level 0, is used only for forward error detection.
- Gigabit LANs use this technique to send 1-Gbps data over four copper cables that can handle 125 Mbaud.
- The extra signal patterns can be used for other purposes such as error detection.

Figure 4.12 *Multilevel: 4D-PAM5 scheme*



Multitransition: MLT-3

- The **multiline transmission, three-level (MLT-3) scheme uses** three levels (+*V*, 0, and -*V*) and three transition rules to move between the levels.
  1. If the next bit is 0, there is no transition.
  2. If the next bit is 1 and the current level is not 0, the next level is 0.
  3. If the next bit is 1 and the current level is 0, the next level is the opposite of the last nonzero level.
- The three voltage levels (-*V*, 0, and +*V*) are shown by three states (ovals).
- It turns out that the shape of the signal in this scheme helps to reduce the required bandwidth.
- MLT-3 a suitable choice when we need to send 100 Mbps on a copper wire that cannot support more than 32 MHz.
- 1 = level change. 0 = no change.

Figure 4.13 *Multitransition: MLT-3 scheme*

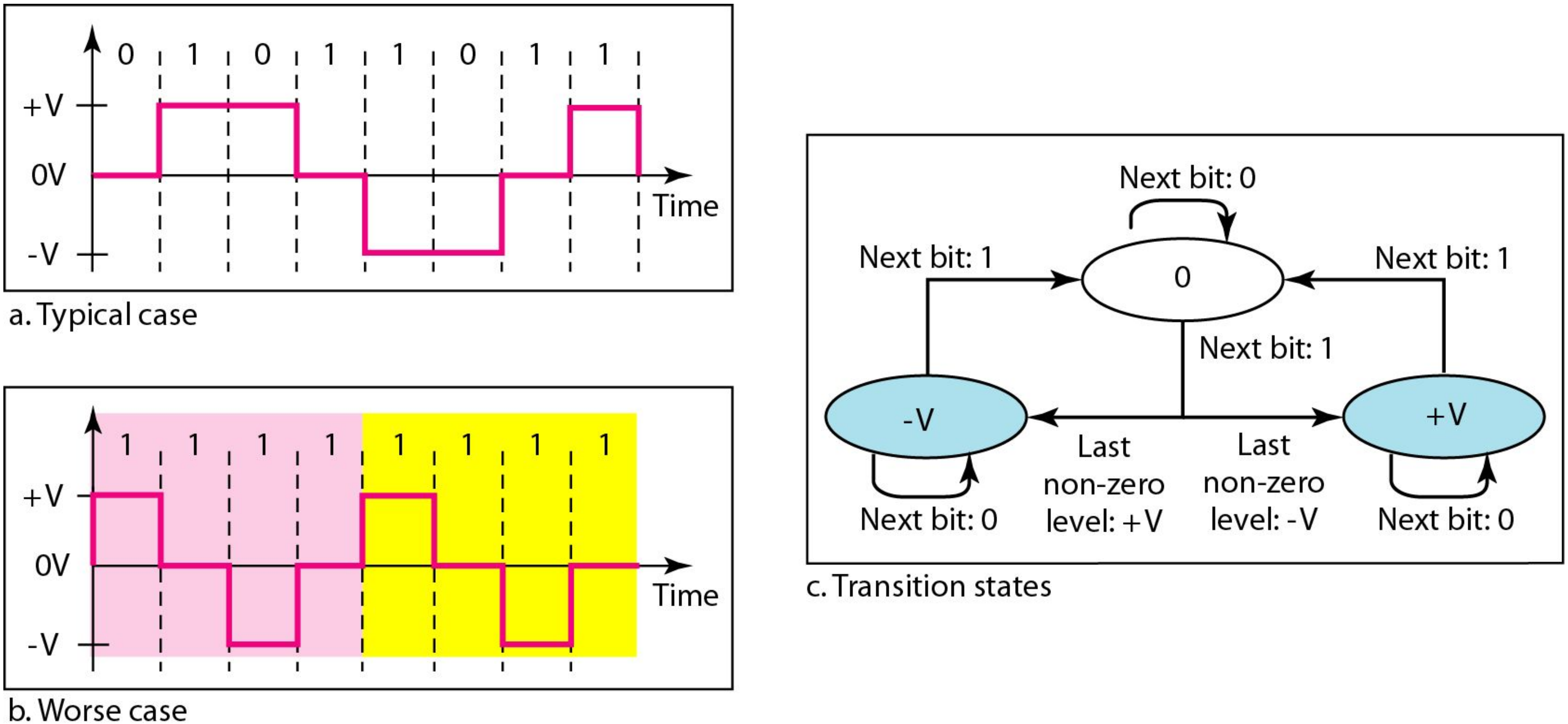


Table 4.1 *Summary of line coding schemes*

Category	*Scheme*	*Bandwidth (average)*	*Characteristics*
Unipolar	NRZ	$B = N/2$	Costly, no self-synchronization if long 0s or 1s, DC
Unipolar	NRZ-L	$B = N/2$	No self-synchronization if long 0s or 1s, DC
	NRZ-I	$B = N/2$	No self-synchronization for long 0s, DC
	Biphase	$B = N$	Self-synchronization, no DC, high bandwidth
Bipolar	AMI	$B = N/2$	No self-synchronization for long 0s, DC
Multilevel	2B1Q	$B = N/4$	No self-synchronization for long same double bits
	8B6T	$B = 3N/4$	Self-synchronization, no DC
	4D-PAM5	$B = N/8$	Self-synchronization, no DC
Multiline	MLT-3	$B = N/3$	No self-synchronization for long 0s

Note

Block coding is normally referred to as *m*B/*n*B coding; it replaces each *m*-bit group with an *n*-bit group.

Figure 4.14 *Block coding concept*

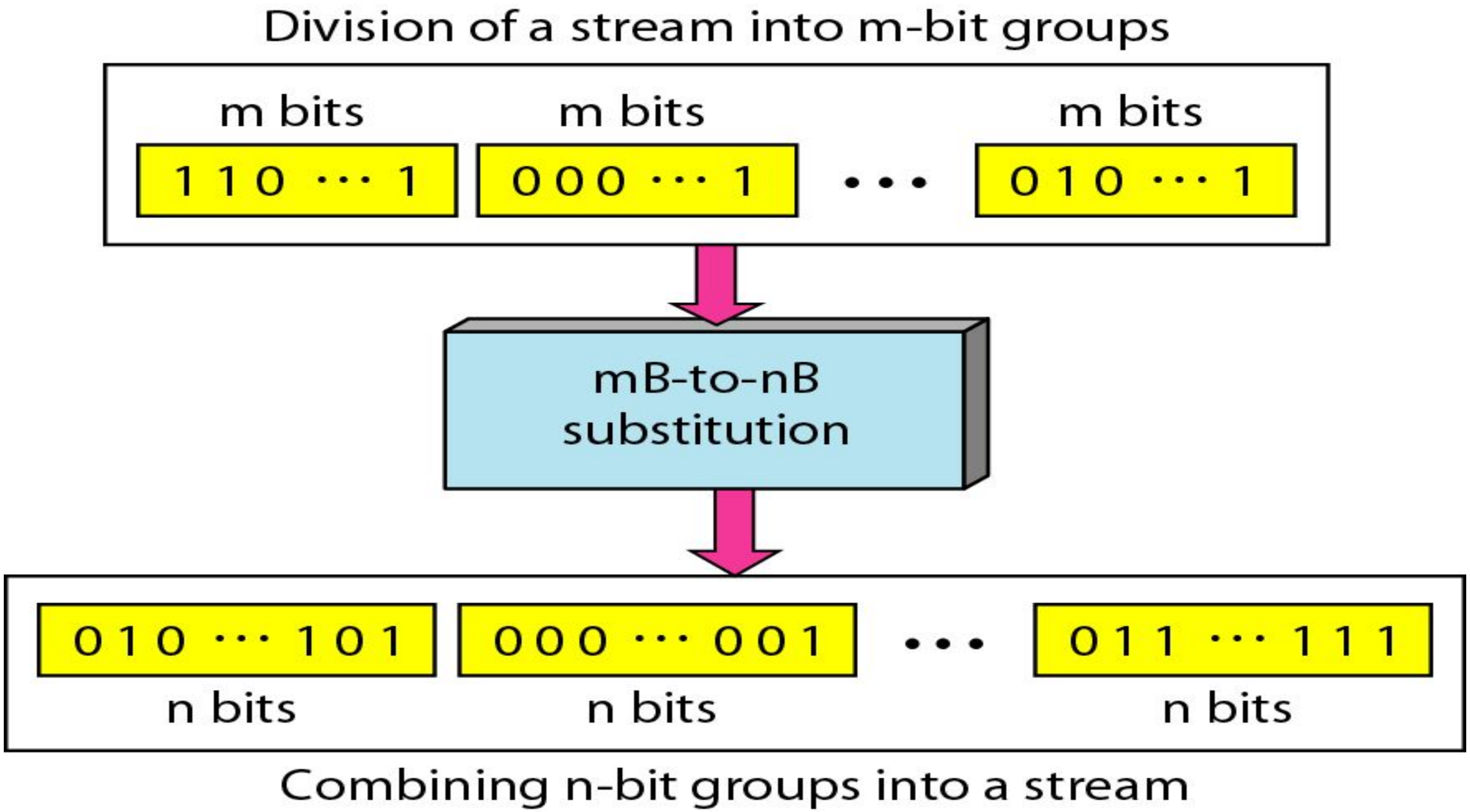


Figure 4.15 *Using block coding 4B/5B with NRZ-I line coding scheme*

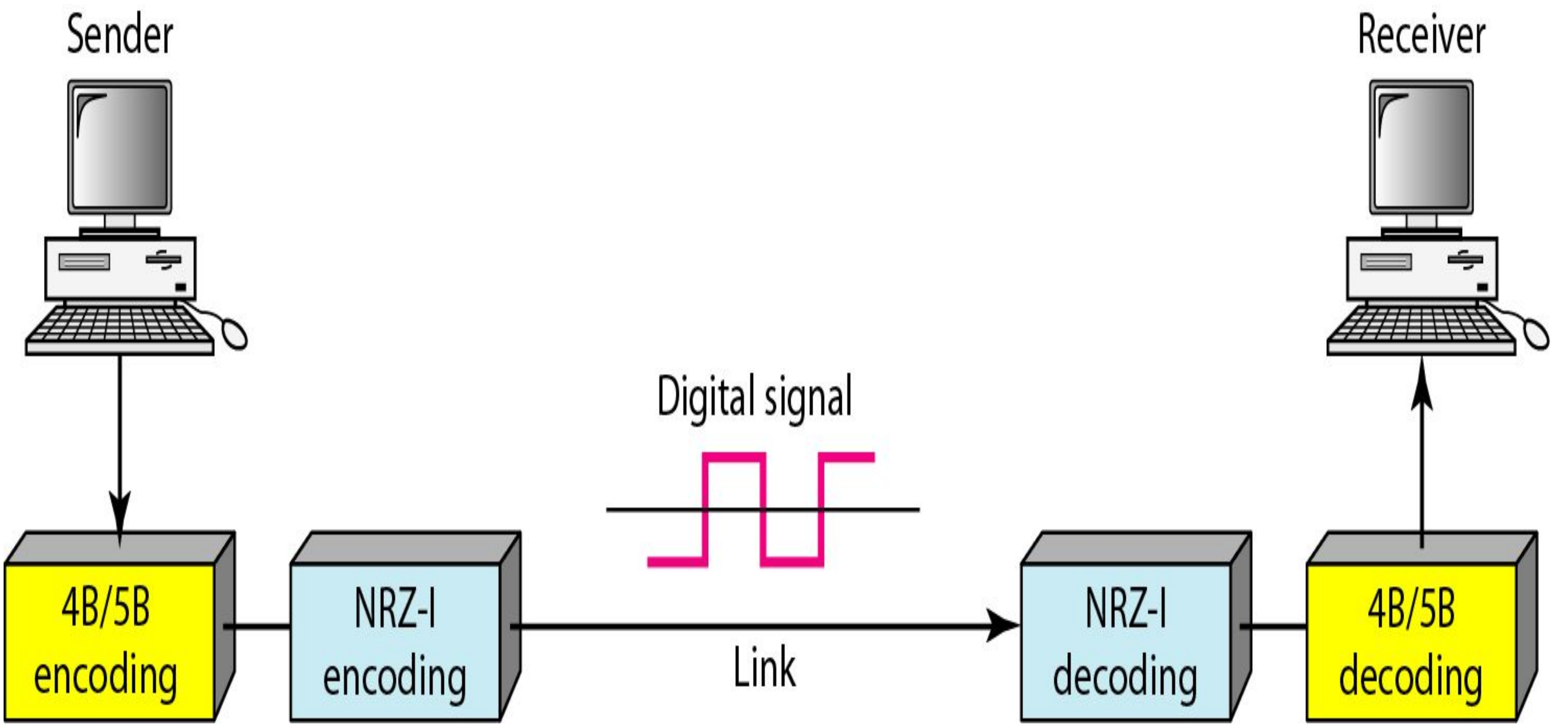


Table 4.2 *4B/5B mapping codes*

Data Sequence	*Encoded Sequence*	*Control Sequence*	*Encoded Sequence*
0000	11110	Q (Quiet)	00000
0001	01001	I (Idle)	11111
0010	10100	H (Halt)	00100
0011	10101	J (Start delimiter)	11000
0100	01010	K (Start delimiter)	10001
0101	01011	T (End delimiter)	01101
0110	01110	S (Set)	11001
0111	01111	R (Reset)	00111
1000	10010		
1001	10011		
1010	10110		
1011	10111		
1100	11010		
1101	11011		
1110	11100		
1111	11101		

Figure 4.16 *Substitution in 4B/5B block coding*

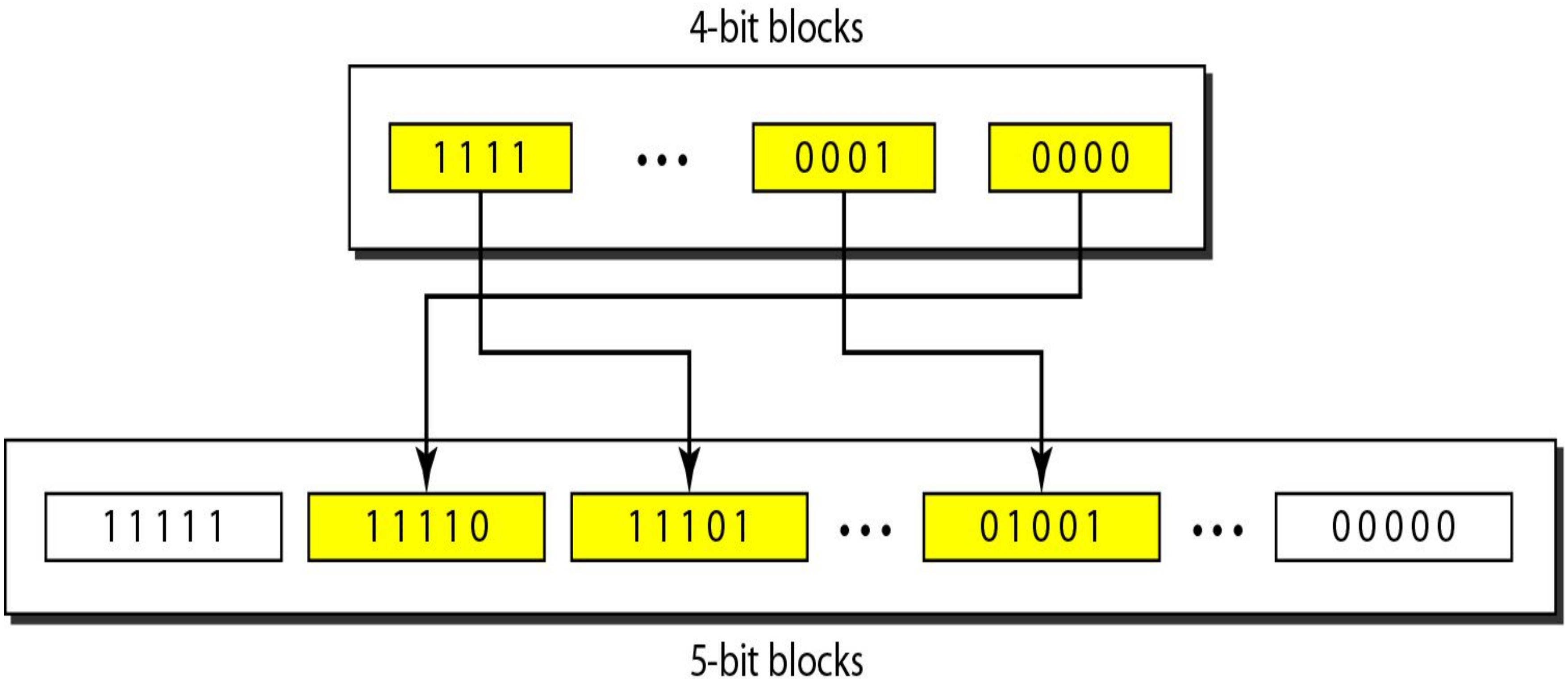
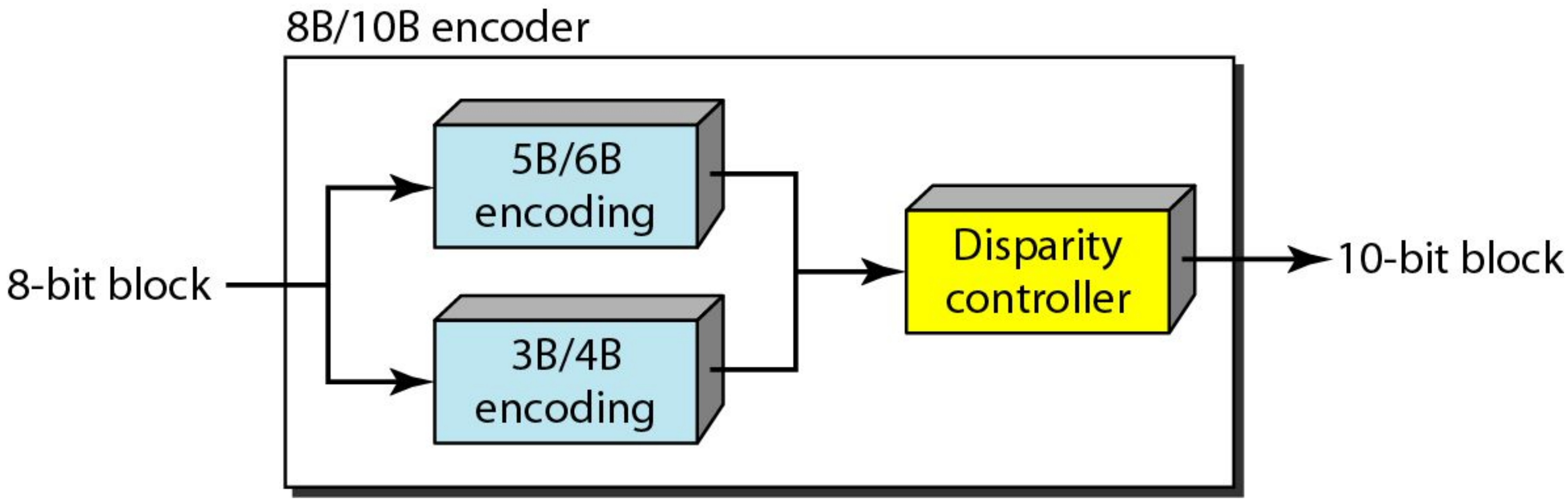


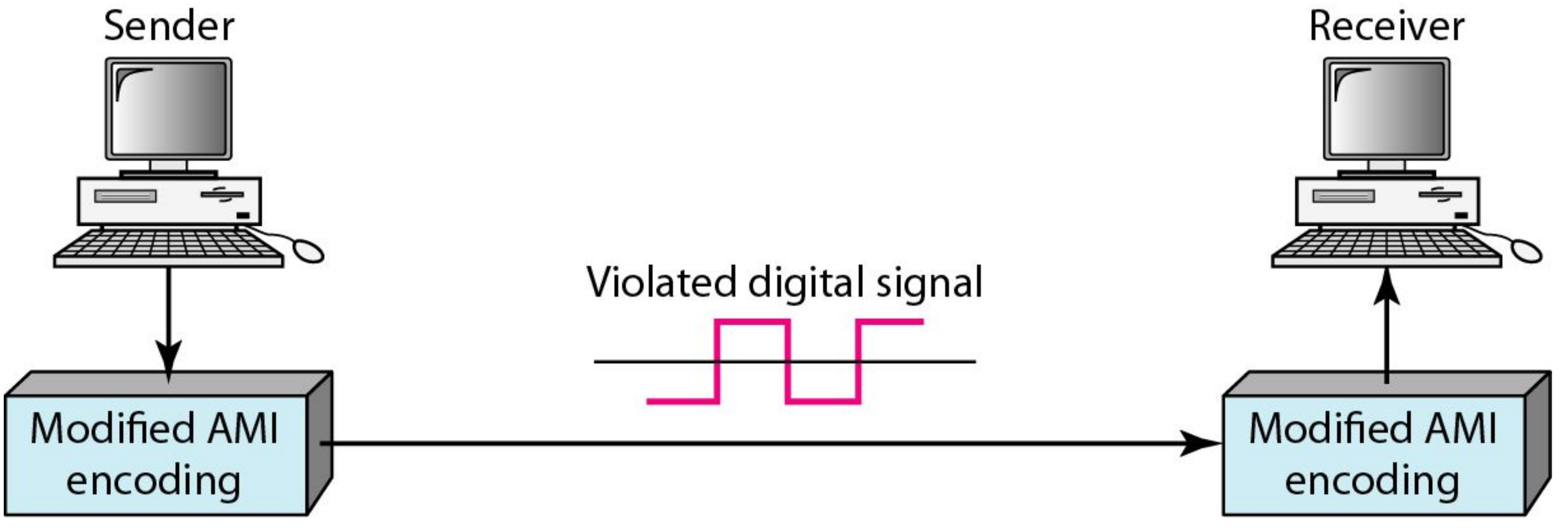
Figure 4.17 *8B/10B block encoding*



Scrambling

- We are looking for a technique that does not increase the number of bits and does provide synchronization.
- We are looking for a solution that substitutes long zero-level pulses with a combination of other levels to provide synchronization.
- One solution is called **scrambling.**
- It is done at the same time when encoding.
- Two common scrambling techniques are B8ZS and HDB3.
- **Bipolar with 8-zero substitution (B8ZS)**: In this technique, eight consecutive zero-level voltages are replaced by the sequence **000VB0VB.**
- **High-density bipolar 3-zero (HDB3)** : Two rules
- **1.** If the number of nonzero pulses after the last substitution is odd, the substitution pattern will be **000V**, which makes the total number of nonzero pulses even.
  2. If the number of nonzero pulses after the last substitution is even, the substitution pattern will be **B00V**, which makes the total number of nonzero pulses even.

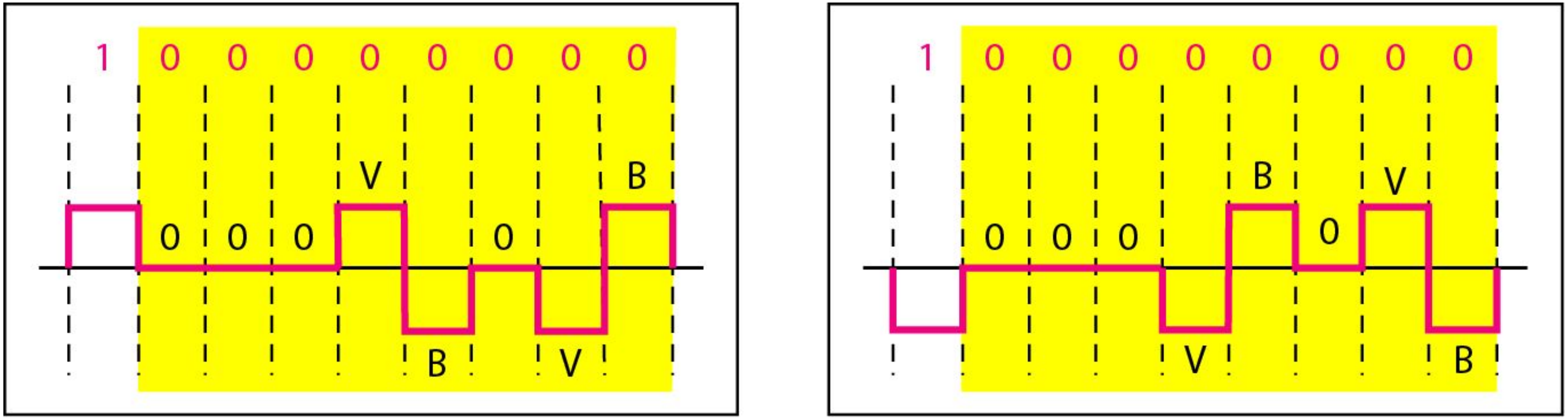
Figure 4.18 *AMI used with scrambling*



Note

B8ZS substitutes eight consecutive zeros with 000VB0VB.

Figure 4.19 *Two cases of B8ZS scrambling technique*



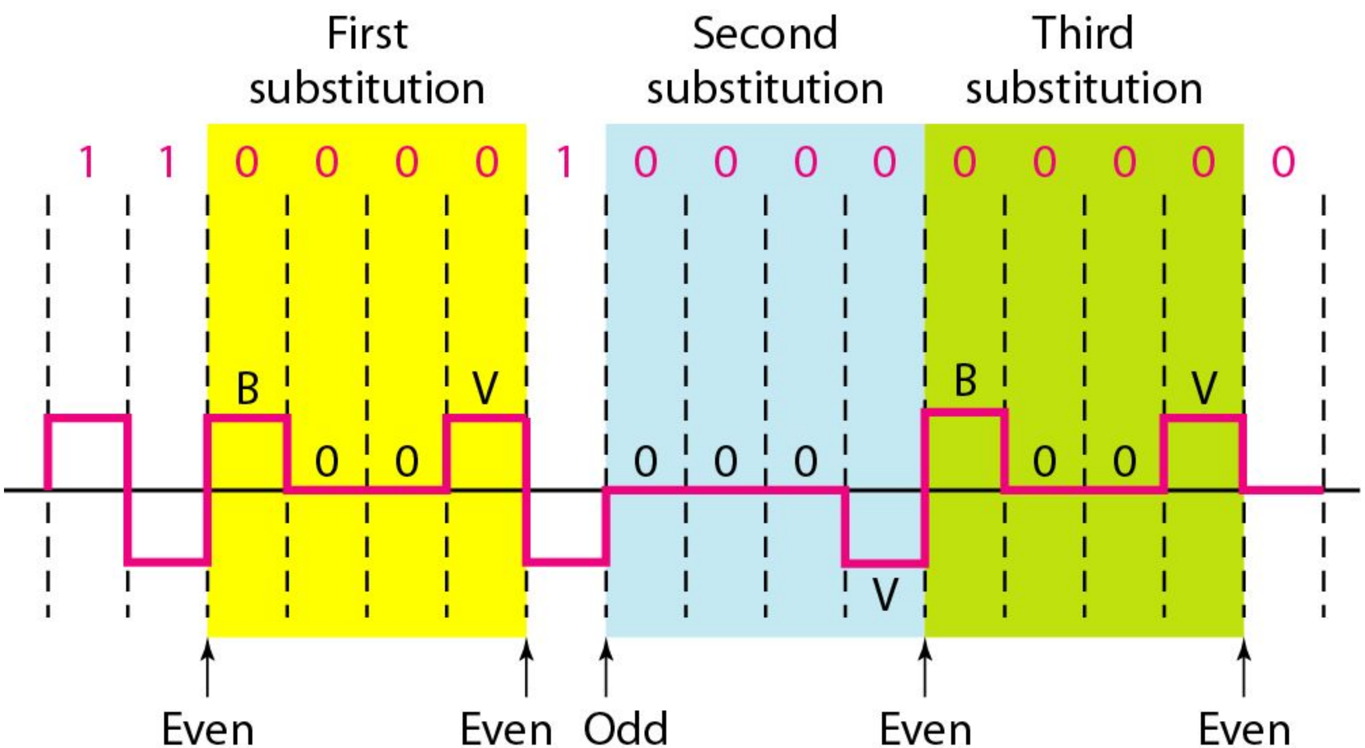
a. Previous level is positive.

b. Previous level is negative.

Note

HDB3 substitutes four consecutive zeros with 000V or B00V depending on the number of nonzero pulses after the last substitution.

Figure 4.20 *Different situations in HDB3 scrambling technique*



4-2 ANALOG-TO-DIGITAL CONVERSION

We have seen in Chapter 3 that a digital signal is superior to an analog signal. The tendency today is to change an analog signal to digital data. In this section we describe two techniques, pulse code modulation and delta modulation.

Topics discussed in this section:

Pulse Code Modulation (PCM)
Delta Modulation (DM)

Figure 4.21 *Components of PCM encoder*

Pulse Code Modulation.

Analog to digital conversion. Digitization.

1. The analog signal is sampled.

2. The sampled signal is quantized.

3. The quantized values are encoded as streams of bits.

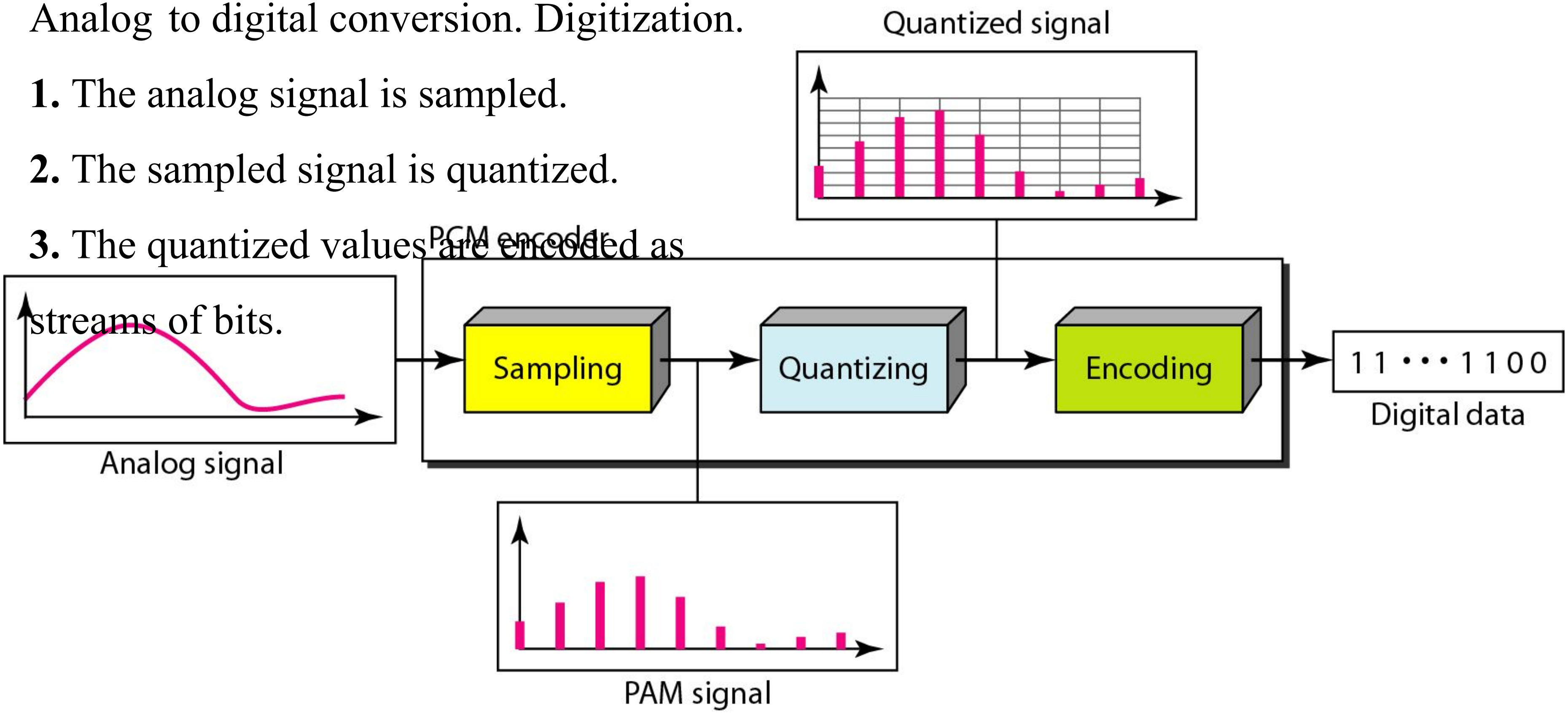
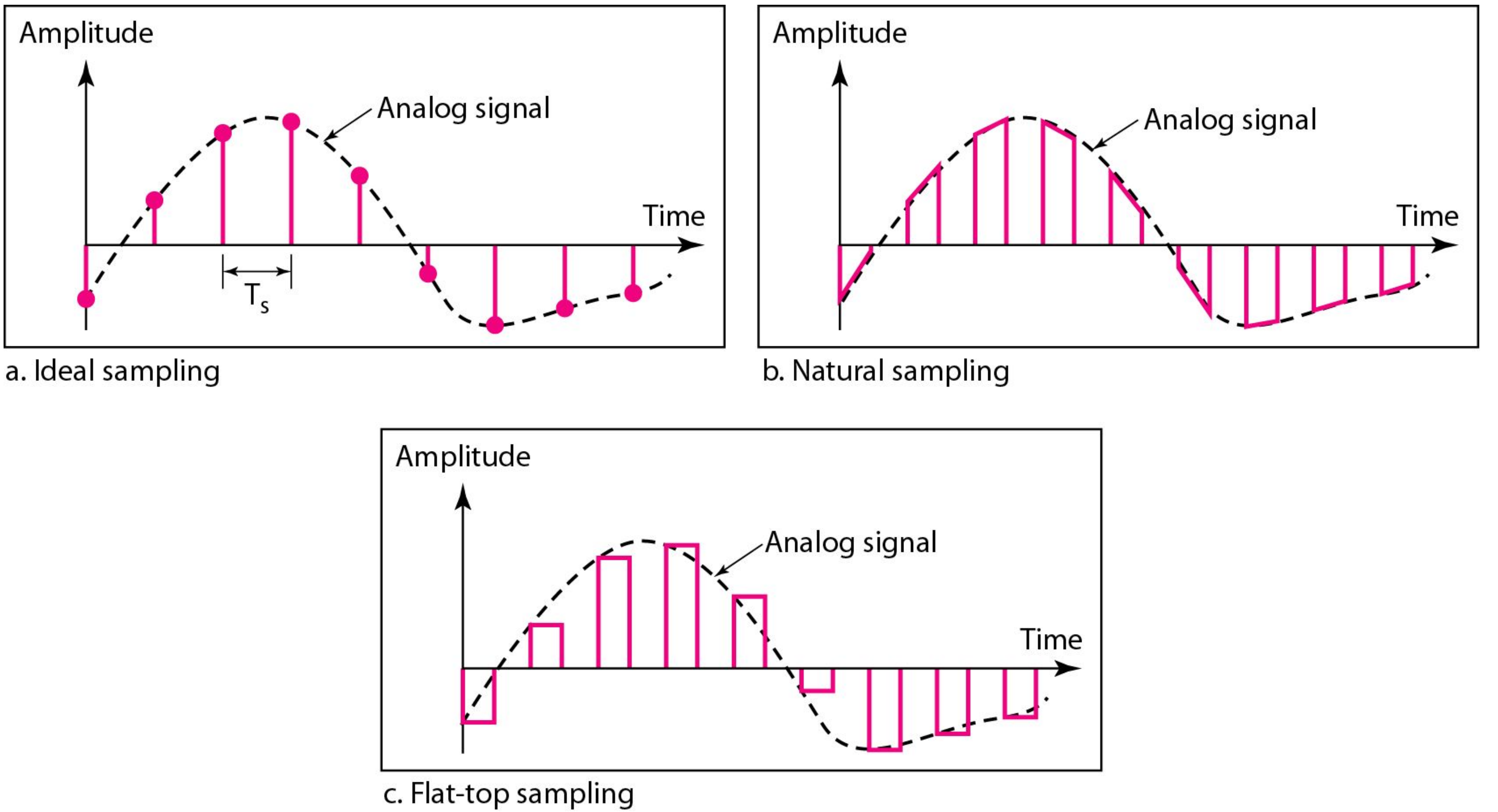


Figure 4.22 *Three different sampling methods for PCM - Pulse Code Modulation, PAM – Pulse Amplitude Modulation = Sampling*

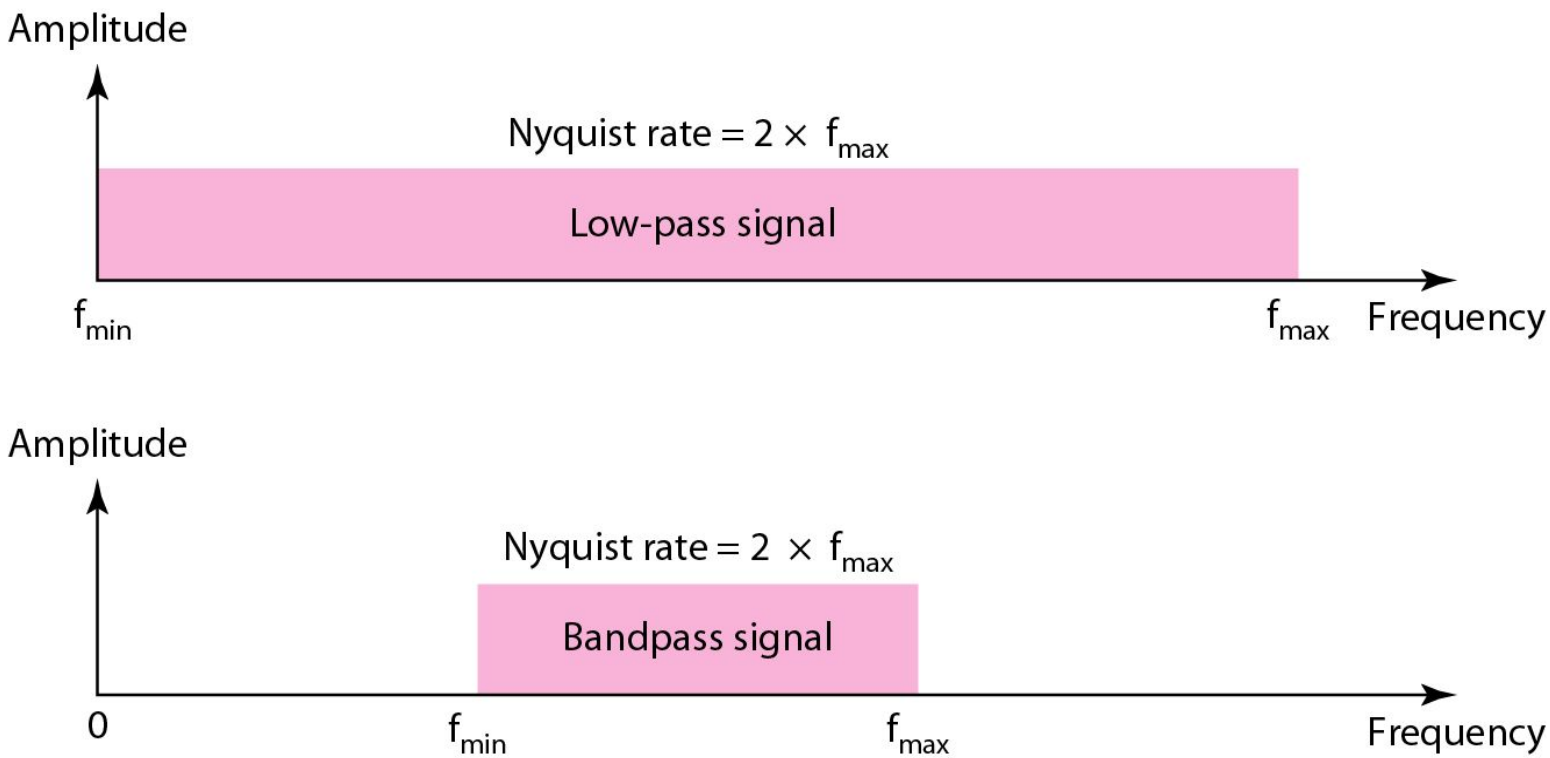


Note

According to the Nyquist theorem, the sampling rate must be at least 2 times the highest frequency contained in the signal.

Figure 4.23 *Nyquist sampling rate for low-pass and bandpass signals*

Signal with infinite band cannot be sampled. Sampling Rate must be 2 times higher than frequency. If analog signal is bandpass, bandwidth is lower than frequency.

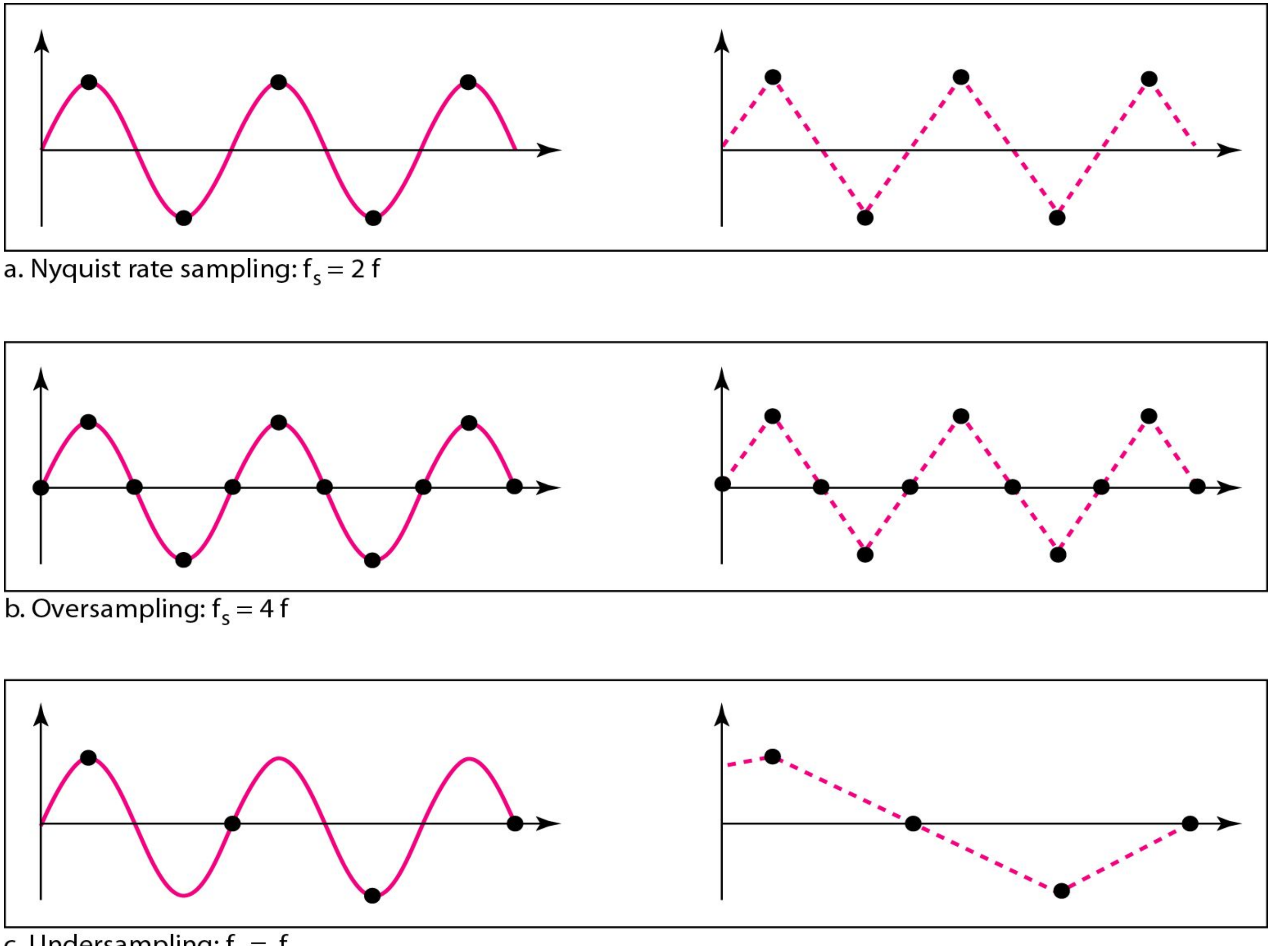


Example 4.6

For an intuitive example of the Nyquist theorem, let us sample a simple sine wave at three sampling rates: $f_s = 4f$ (2 times the Nyquist rate), $f_s = 2f$ (Nyquist rate), and $f_s = f$ (one-half the Nyquist rate). Figure 4.24 shows the sampling and the subsequent recovery of the signal.

It can be seen that sampling at the Nyquist rate can create a good approximation of the original sine wave (part a). Oversampling in part b can also create the same approximation, but it is redundant and unnecessary. Sampling below the Nyquist rate (part c) does not produce a signal that looks like the original sine wave.

Figure 4.24 *Recovery of a sampled sine wave for different sampling rates*



a. Nyquist rate sampling: $f_s = 2f$

b. Oversampling: $f_s = 4f$

c. Undersampling: $f_s = f$

Example 4.7

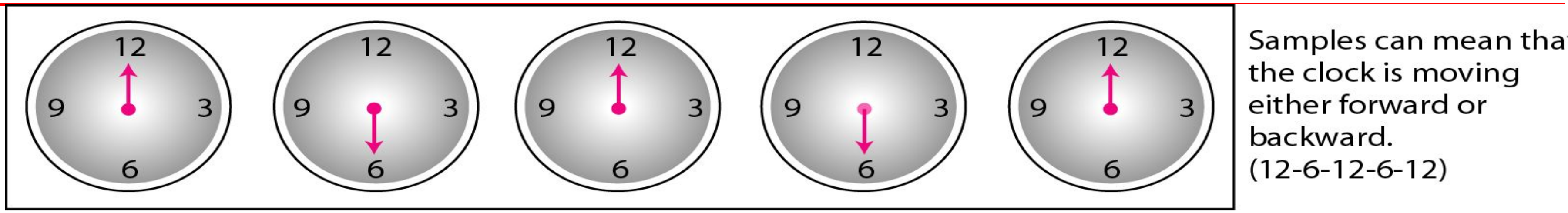
Consider the revolution of a hand of a clock. The second hand of a clock has a period of 60 s. According to the Nyquist theorem, we need to sample the hand every 30 s ($T_s = T$ or $f_s = 2f$).

In Figure 4.25a, the sample points, in order, are 12, 6, 12, 6, 12, and 6. The receiver of the samples cannot tell if the clock is moving forward or backward.

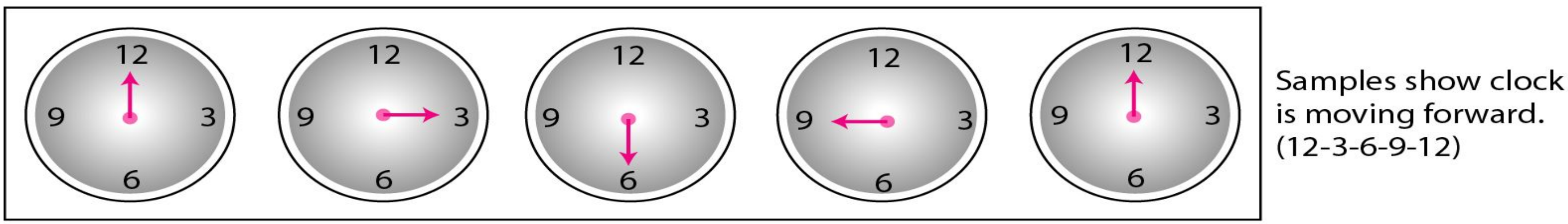
In part b, we sample at double the Nyquist rate (every 15 s). The sample points are 12, 3, 6, 9, and 12. The clock is moving forward.

In part c, we sample below the Nyquist rate ($T_s = T$ or $f_s = f$). The sample points are 12, 9, 6, 3, and 12. Although the clock is moving forward, the receiver thinks that the clock is moving backward.

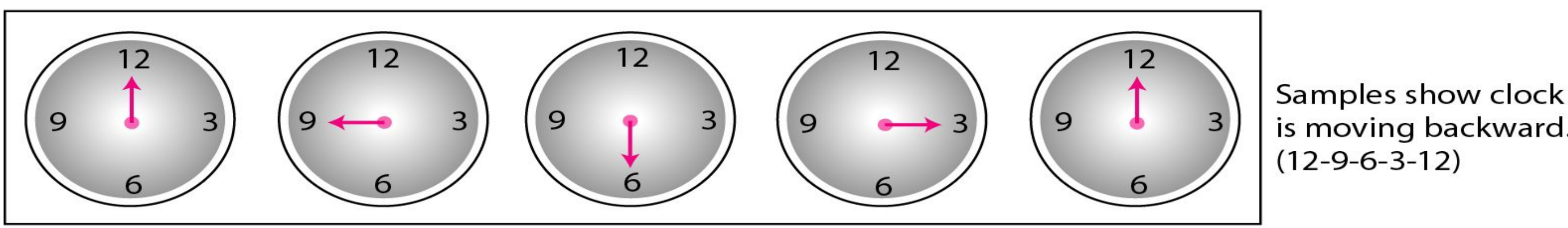
Figure 4.25 *Sampling of a clock with only one hand*



a. Sampling at Nyquist rate: $T_s = T\frac{1}{2}$



b. Oversampling (above Nyquist rate): $T_s = T\frac{1}{4}$



c. Undersampling (below Nyquist rate): $T_s = T\frac{3}{4}$

Example 4.8

An example related to Example 4.7 is the seemingly backward rotation of the wheels of a forward-moving car in a movie.
This can be explained by under-sampling.
A movie is filmed at 24 frames per second.
If a wheel is rotating more than 12 times per second, the under-sampling creates the impression of a backward rotation.

Example 4.9

Telephone companies digitize voice by assuming a maximum frequency of 4000 Hz. The sampling rate therefore is 8000 samples per second.

Example 4.10

A complex low-pass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?

Solution

The bandwidth of a low-pass signal is between 0 and f, where f is the maximum frequency in the signal. Therefore, we can sample this signal at 2 times the highest frequency (200 kHz). The sampling rate is therefore 400,000 samples per second.

Example 4.11

A complex bandpass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?

Solution

We cannot find the minimum sampling rate in this case because we do not know where the bandwidth starts or ends. We do not know the maximum frequency in the signal.

Quantization

- Sampling results in pulses with infinite amplitude, which cannot be used for encoding.
- So, we need Quantization.
- Steps for Quantization.

1. We assume that the original analog signal has instantaneous amplitudes between *V*min and *V*max.

2. We divide the range into *L* zones, each of height Δ (delta).

$$\Delta = Vmax – Vmin / L$$

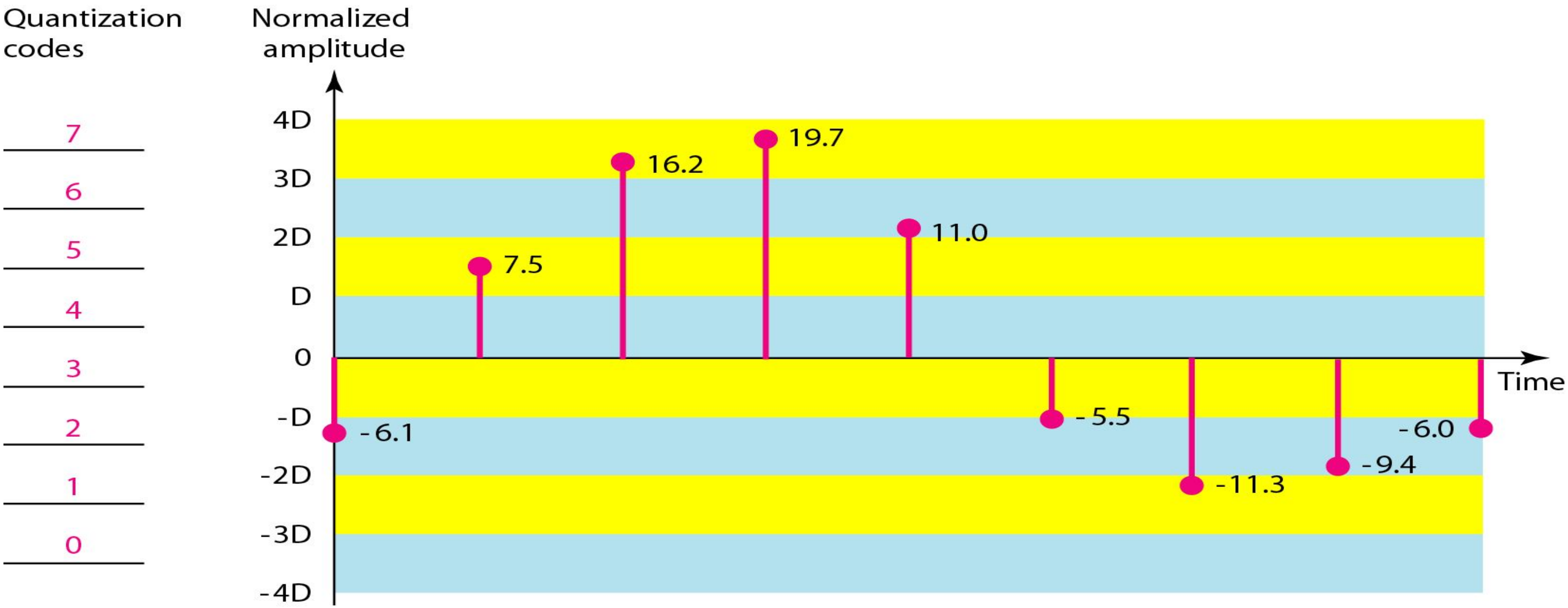
3. We assign quantized values of 0 to *L* - 1 to the midpoint of each zone.

4. We approximate the value of the sample amplitude to the quantized values.

Quantization

- Consider, sampled signal and the sample amplitudes are between -20 and +20 V.
- We decide to have eight levels (L = 8). This means that Δ = 5 V.
- We have shown only nine samples using ideal sampling.
- Actual amplitude is shown in the graph.
- Normalized value for each sample is calculated for actual amplitude/Δ.
- The quantization process selects the quantization value from the middle of each zone. This means that the normalized quantized values (second row).
- The difference is called the *normalized error* (third row).
- The fourth row is the quantization code for each sample based on the quantization levels at the left of the graph.
- The encoded words (fifth row) are the final products of the conversion to binary.

Figure 4.26 *Quantization and encoding of a sampled signal*



Normalized PAM values	-1.22	1.50	3.24	3.94	2.20	-1.10	-2.26	-1.88	-1.20
Normalized quantized values	-1.50	1.50	3.50	3.50	2.50	-1.50	-2.50	-1.50	-1.50
Normalized error	-0.38	0	+0.26	-0.44	+0.30	-0.40	-0.24	+0.38	-0.30
Quantization code	2	5	7	7	6	2	1	2	2
Encoded words	010	101	111	111	110	010	001	010	010

Quantization

- In audio digitizing, *L* is normally chosen to be 256; in video it is normally thousands. Choosing lower values of *L* increases the quantization error if there is a lot of fluctuation in the signal.
- Quantization is an approximation process.
- Input is real value and output is approximation.
- Error occurs only when the input value is not the middle of the level.
- The quantization error changes the signal-to-noise ratio of the signal, which in turn **reduces the upper limit capacity according to Shannon**.
- **Quantization error** to the SNRdB of the signal depends on the number of quantization levels *L*, or the bits per sample *nb*, with formula.

$$\mathrm{SNR_{dB}} = 6.02n_b + 1.76 \quad \mathrm{dB}$$

Example 4.12

What is the SNR_{dB} in the example of Figure 4.26? Means, if we have eight levels and 3 bits per sample what will be the SNR_{dB} ?

Solution

We can use the formula to find the quantization. We have eight levels and 3 bits per sample, so

$$SNR_{dB} = 6.02(3) + 1.76 = 19.82\ dB$$

Increasing the number of levels increases the SNR.

Example 4.13

A telephone subscriber line must have an SNR_{dB} above 40. What is the minimum number of bits per sample?

Solution

We can calculate the number of bits as

$$\boxed{SNR_{dB} = 6.02n_b + 1.76 = 40 \quad \Rightarrow \quad n = 6.35}$$

Telephone companies usually assign 7 or 8 bits per sample.

Encoding

- The last step in PCM is encoding. After each sample is quantized and the number of bits per sample is decided, each sample can be changed to an *nb*-bit code word.
- Last row in the figure of quantization.
- A quantization code of 2 is encoded as 010; 5 is encoded as 101; and so on.
- If the number of quantization levels is *L*, the number of bits is ***nb* = log2 *L*.**
- The bit rate can be found from the formula:

$$\text{Bit rate} = \text{sampling rate} \times \text{number of bits per sample} = f_s \times n_b$$

Example 4.14

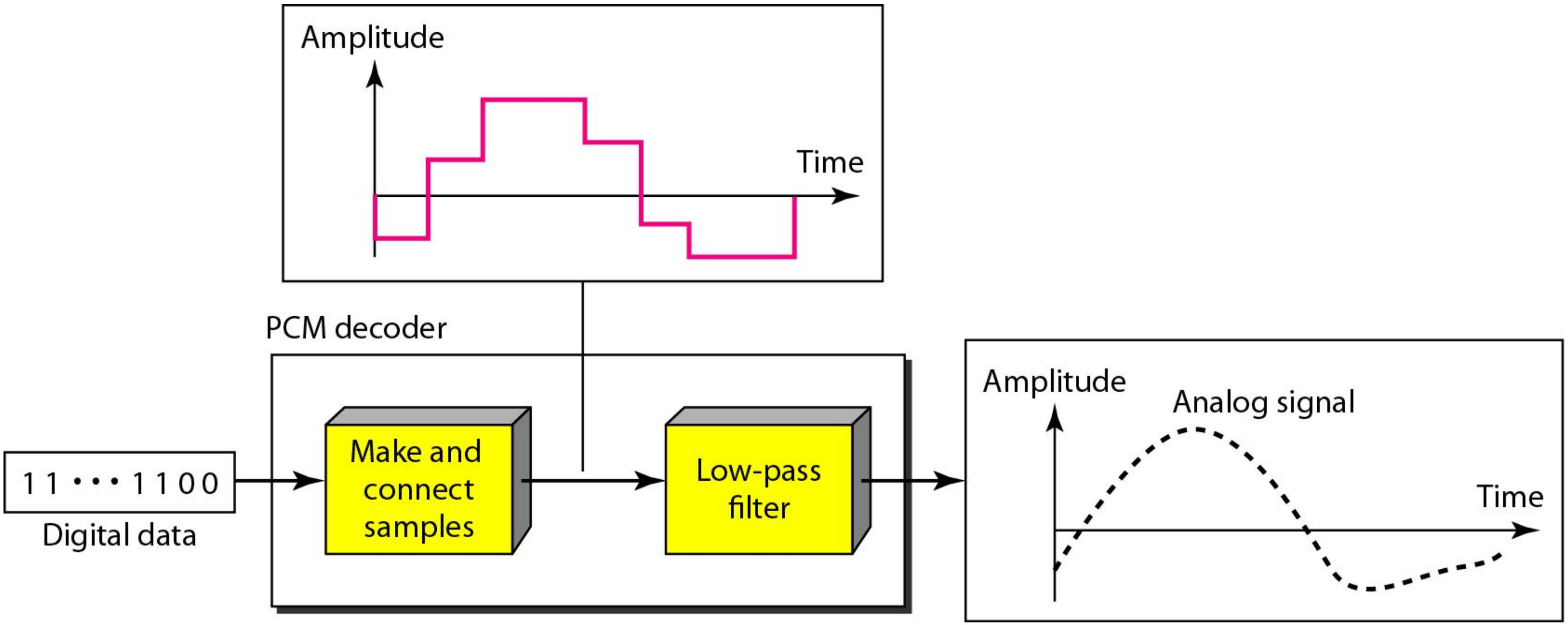
We want to digitize the human voice. What is the bit rate, assuming 8 bits per sample?

Solution

The human voice normally contains frequencies from 0 to 4000 Hz. So the sampling rate and bit rate are calculated as follows:

$$\text{Sampling rate} = 4000 \times 2 = 8000 \text{ samples/s}$$
$$\text{Bit rate} = 8000 \times 8 = 64{,}000 \text{ bps} = 64 \text{ kbps}$$

Figure 4.27 *Components of a PCM decoder*



PCM Bandwidth, Maximum Data Rate of a Channel & Minimum Required Bandwidth

$$B_{\min} = n_b \times B_{\text{analog}}$$

$$N_{\max} = 2 \times B \times \log_2 L \quad \text{bps}$$

$$B_{\min} = \frac{N}{(2 \times \log_2) L} \quad \text{Hz}$$

Example 4.15

We have a low-pass analog signal of 4 kHz. If we send the analog signal, we need a channel with a minimum bandwidth of 4 kHz. If we digitize the signal and send 8 bits per sample, we need a channel with a minimum bandwidth of 8 × 4 kHz = 32 kHz.

Delta Modulator

- PCM is a very complex technique. Other techniques have been developed to reduce the complexity of PCM.
- The simplest is **delta modulation.**
- PCM finds the value of the signal amplitude for each sample; DM finds the change from the previous sample.
- Note that there are no code words here; bits are sent one after another.

Figure 4.28 *The process of delta modulation*

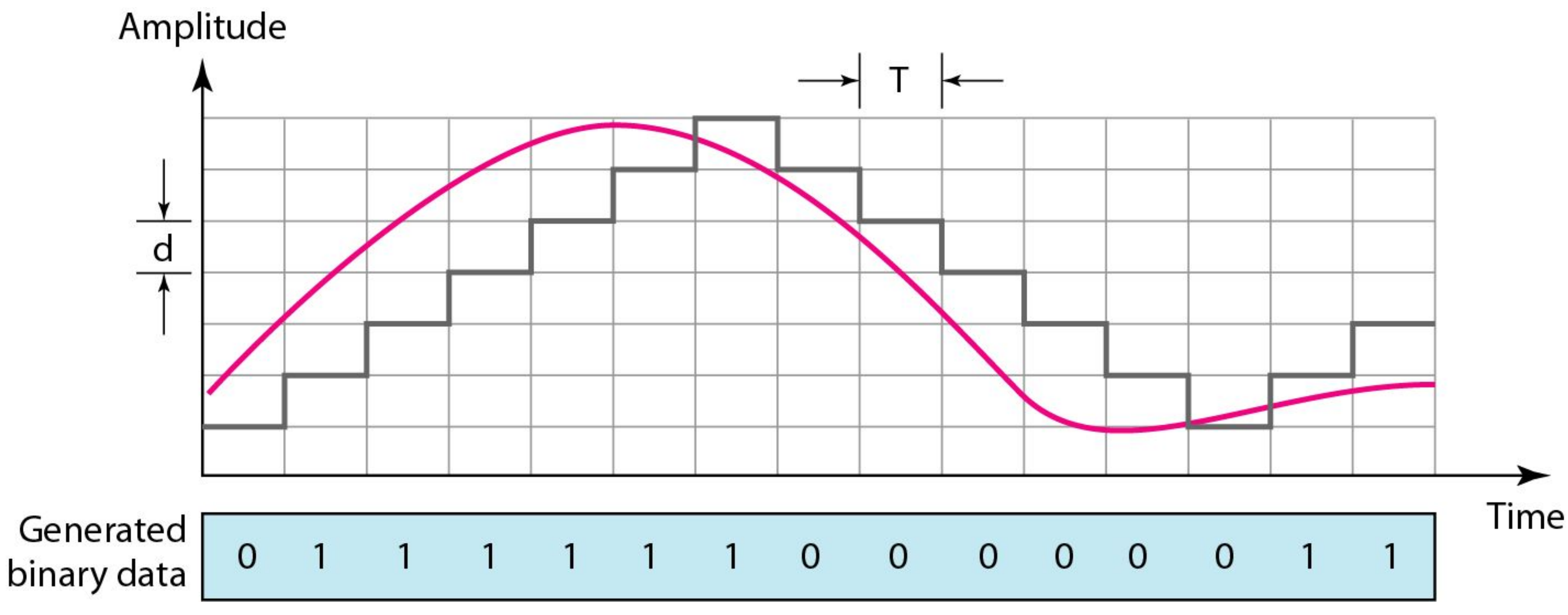


Figure 4.29 *Delta modulation components*

- Modulator: is used at the sender site to create a stream of bits from an analog signal.
- If the delta is positive, the process records a 1; if it is negative, the process records a 0.
- Base of comparison is required. Which is done by Staircase Maker.
- The modulator, at each sampling interval, compares the value of the analog signal with the last value of the staircase signal.
- Note that we need a delay unit to hold the staircase function for a period between two comparisons.

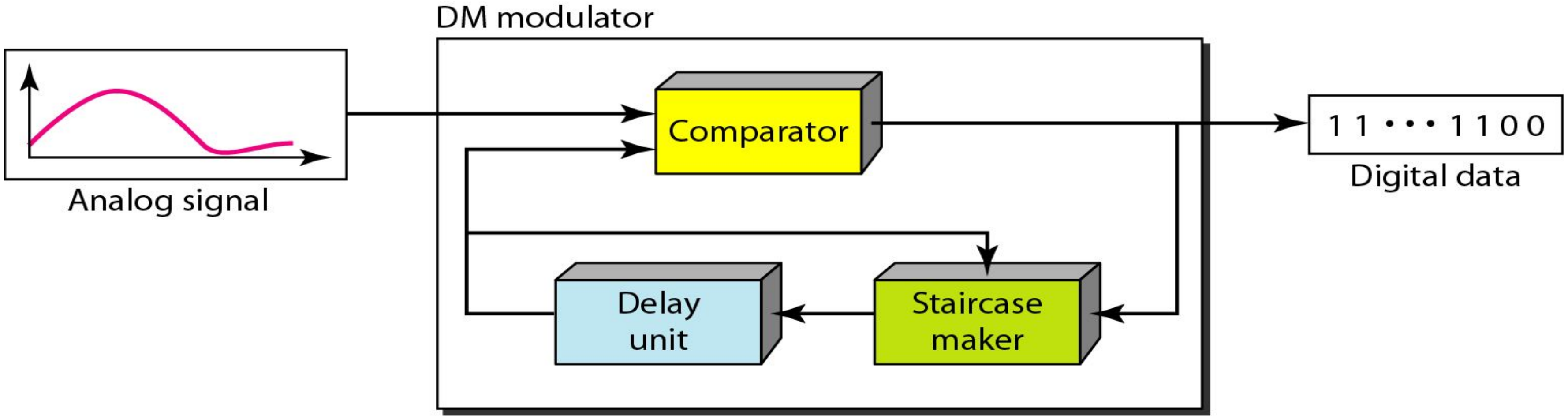
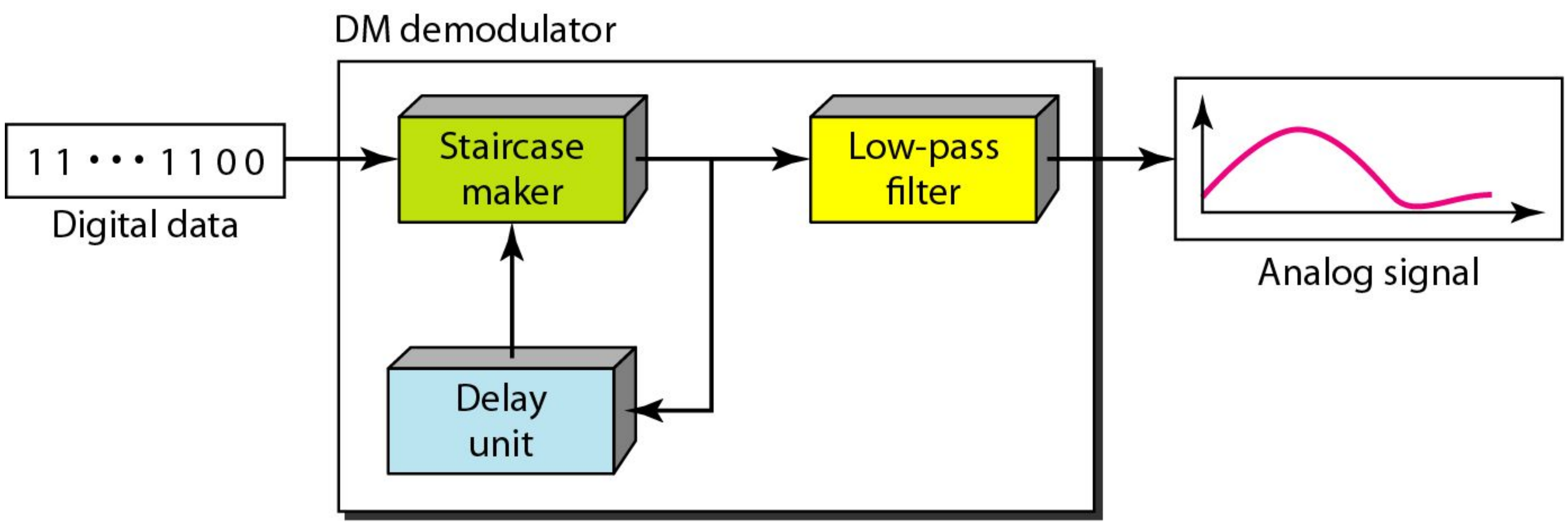


Figure 4.30 *Delta demodulation components*

- Demodulator: The demodulator takes the digital data and, using the staircase maker and the delay unit, creates the analog signal.
- Low-pass filter is used for smoothing.
- Adaptive DM: A better performance can be achieved if the value of δ is not fixed. In adaptive delta modulation, the value of δ changes according to the amplitude of the analog signal.
- Quantization Error: DM is not perfect. Quantization error is always introduced in the process. The quantization error of DM, however, is much less than that for PCM.



4-3 TRANSMISSION MODES

The transmission of binary data across a link can be accomplished in either Parallel or Serial mode. In parallel mode, multiple bits are sent with each clock tick. In serial mode, 1 bit is sent with each clock tick. While there is only one way to send parallel data, there are three subclasses of Serial Transmission: Asynchronous, Synchronous, and Isochronous.

Topics discussed in this section:

Parallel Transmission
Serial Transmission

Figure 4.31 *Data transmission and modes*

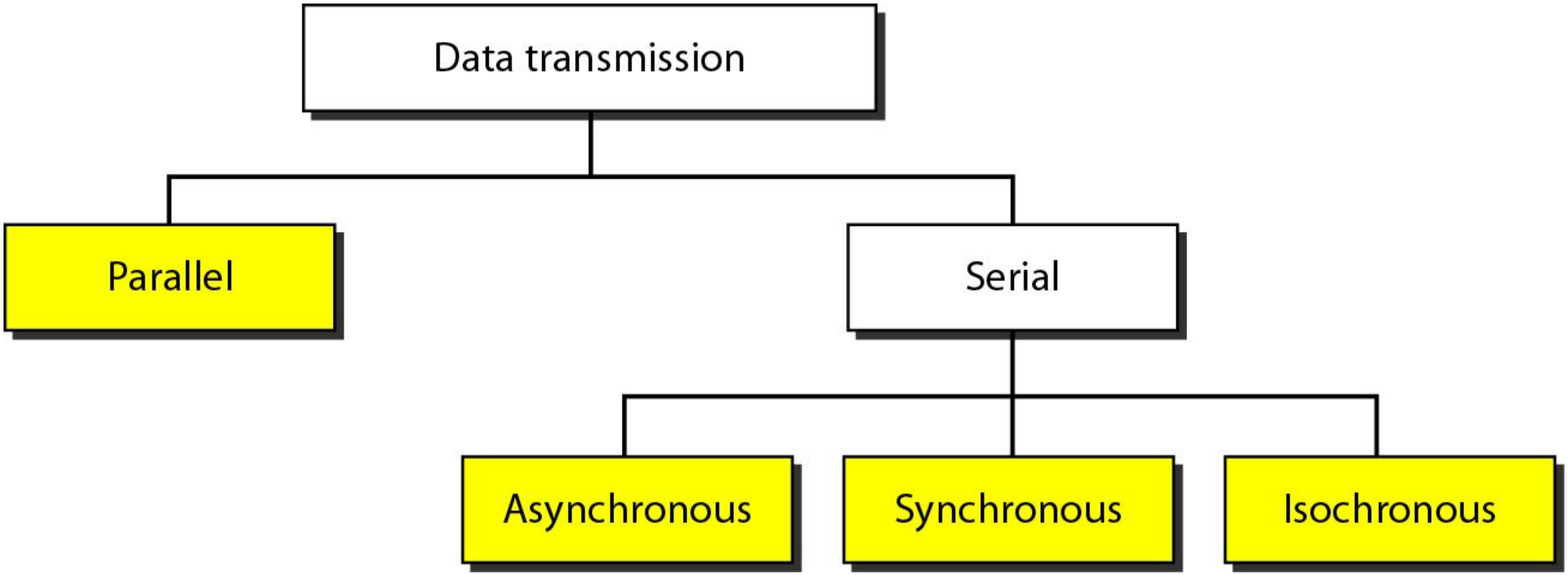


Figure 4.32 *Parallel transmission*

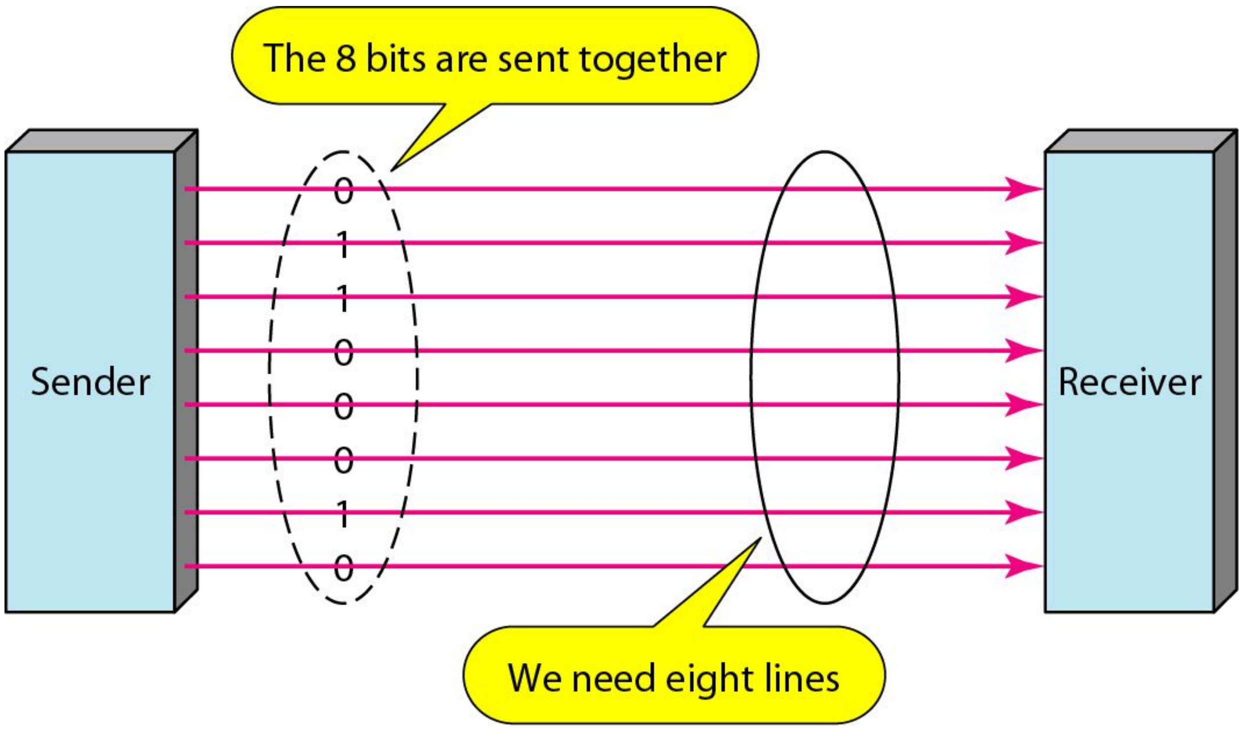
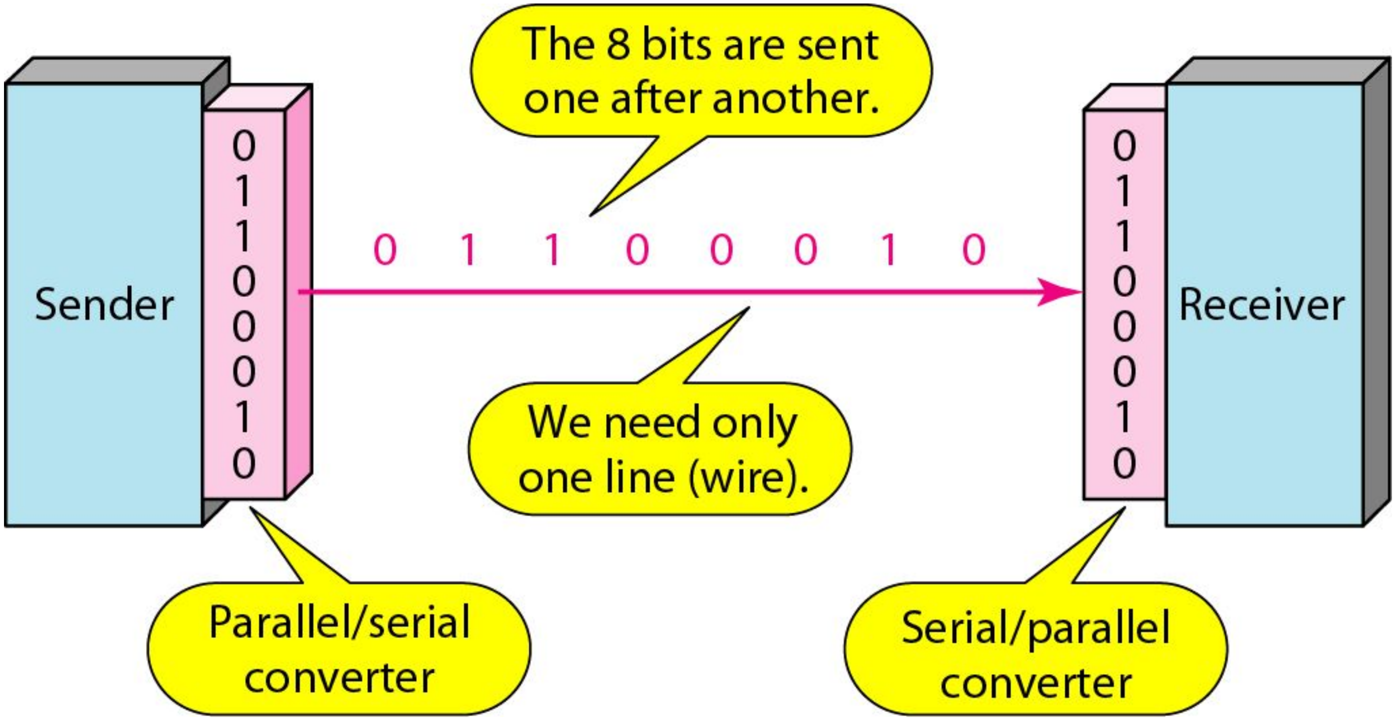


Figure 4.33 *Serial transmission*



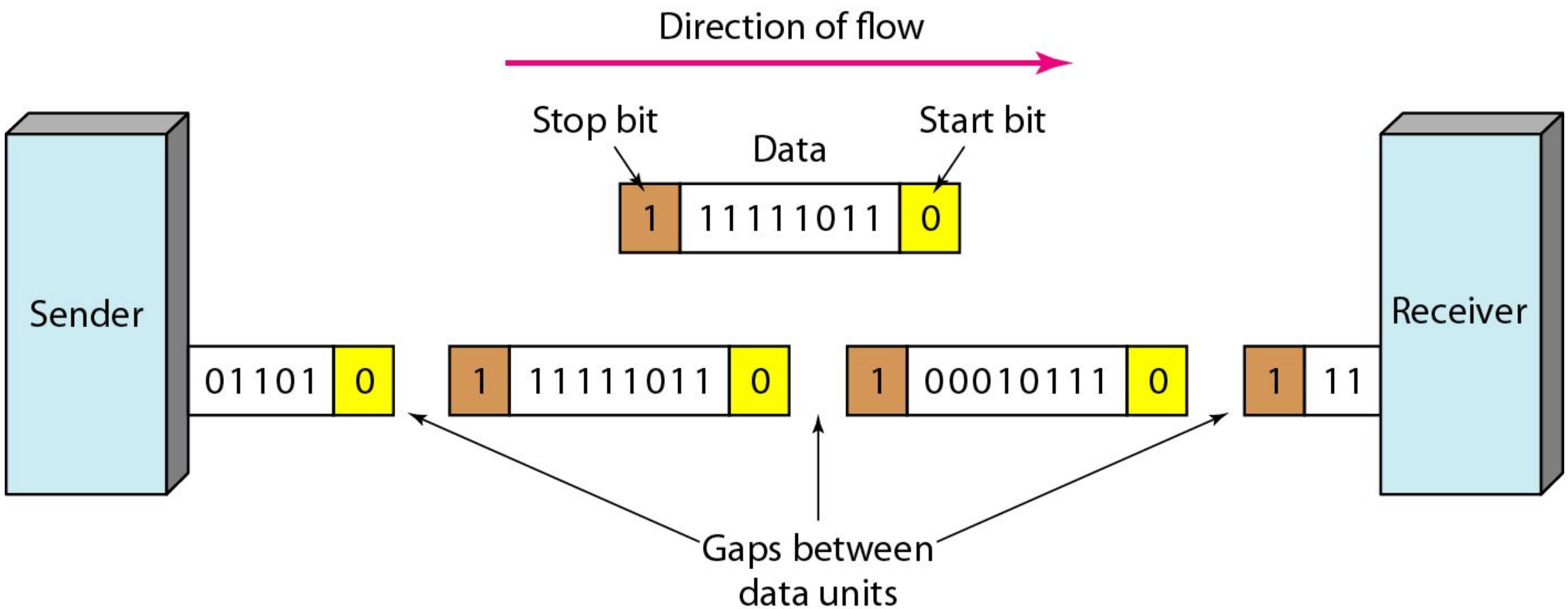
Note

In asynchronous transmission, we send 1 start bit (0) at the beginning and 1 or more stop bits (1s) at the end of each byte. There may be a gap between each byte.

Note

Asynchronous here means "asynchronous at the byte level," but the bits are still synchronized; their durations are the same.

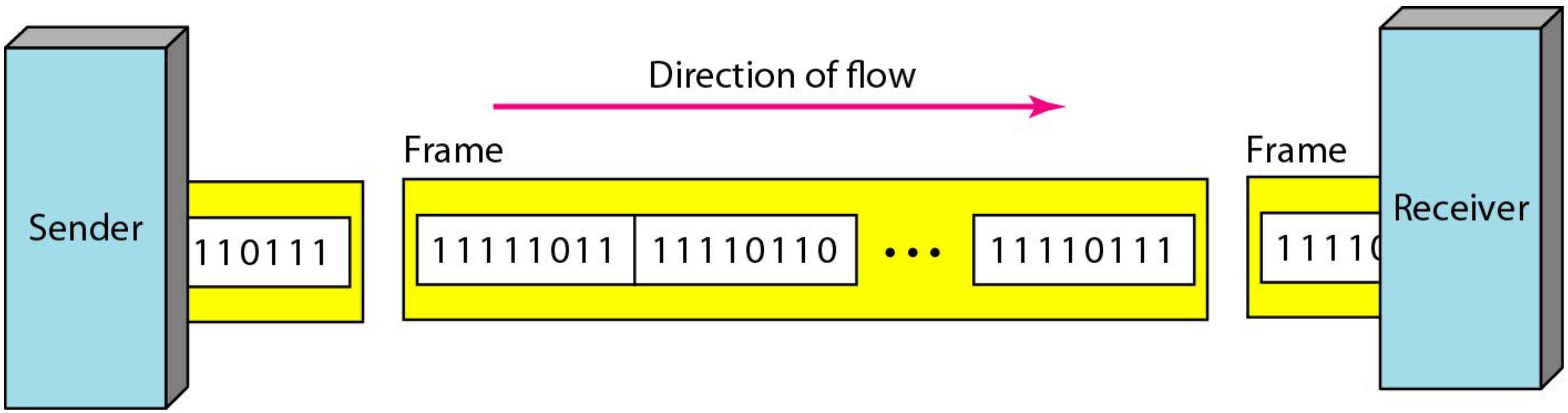
Figure 4.34 *ASYNCHRONOUS TRANSMISSION*



Note

In synchronous transmission, we send bits one after another without start or stop bits or gaps. It is the responsibility of the receiver to group the bits.

Figure 4.35 *SYNCHRONOUS TRANSMISSION*



ISOCHRONOUS TRANSMISSION

- In real-time audio and video, in which uneven delays between frames are not acceptable, synchronous transmission fails.
  - ✔ For example, TV images are broadcast at the rate of 30 images per second; they must be viewed at the same rate. If each image is sent by using one or more frames, there should be no delays between frames.
- Synchronization between characters is not enough; the entire stream of bits must be synchronized. The **isochronous transmission guarantees that the data arrive at** a fixed rate.