Chapter 4

Digital Transmission

Components of Data Communication

- **Data**
  - **Analog: Continuous value data (sound, light, temperature)**
  - **Digital: Discrete value (text, integers, symbols)**
- **Signal**
  - **Analog: Continuously varying electromagnetic wave**
  - **Digital: Series of voltage pulses (square wave)**

Analog Data-->Signal Options

- Analog data to analog signal
  - Inexpensive, easy conversion (eg telephone)
  - Used in traditional analog telephony
- Analog data to digital signal
  - Requires a codec (encoder/decoder)
  - Allows use of digital telephony, voice mail

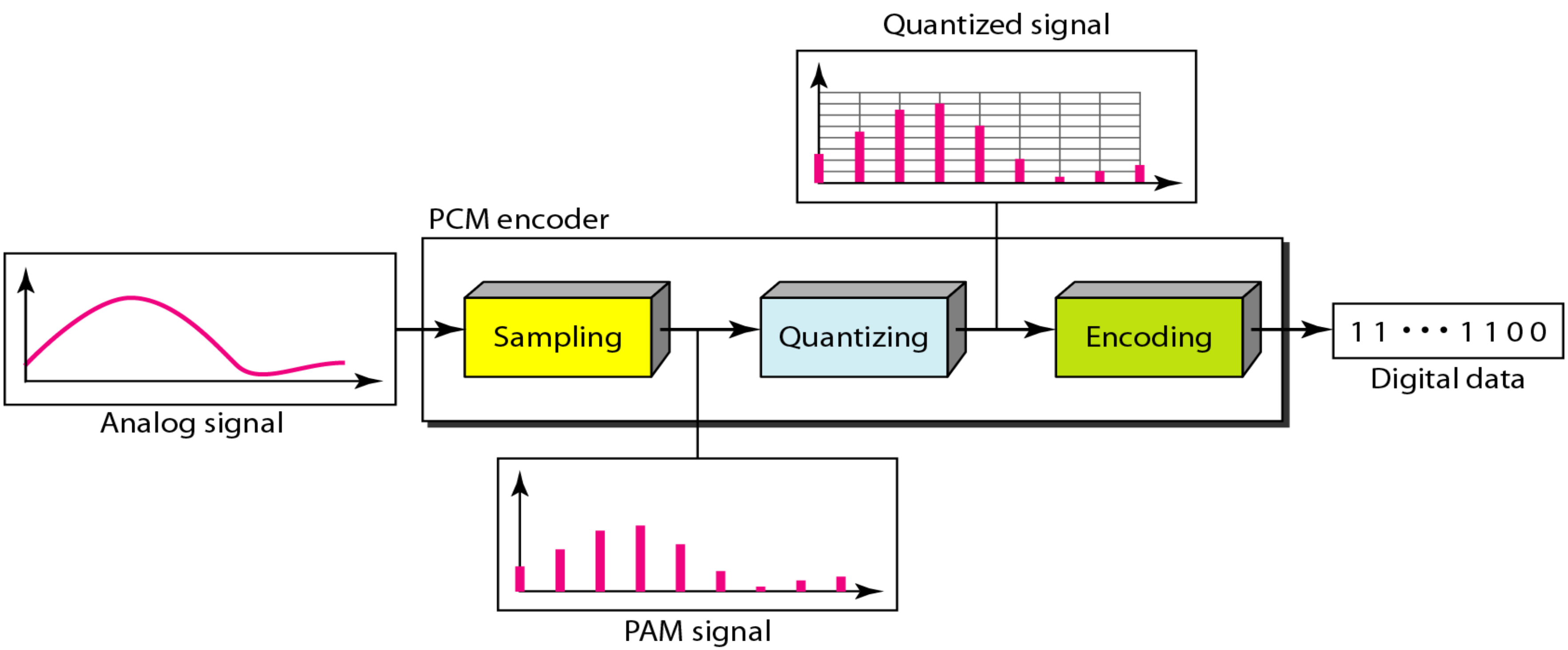
4-2 ANALOG-TO-DIGITAL CONVERSION

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*.

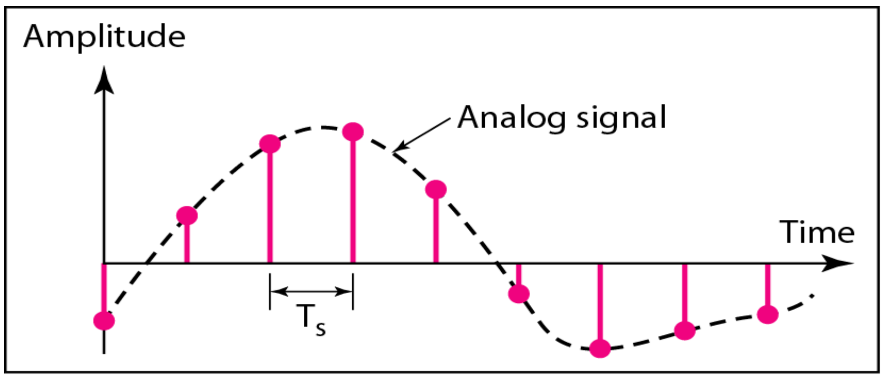
The most common technique to change an analog signal to digital data (digitization) is called pulse code modulation (PCM). A PCM encoder has three processes, as shown in Fig.

1. The analog signal is sampled.
2. The sampled signal is quantized.
3. The quantized values are encoded as streams of bits.

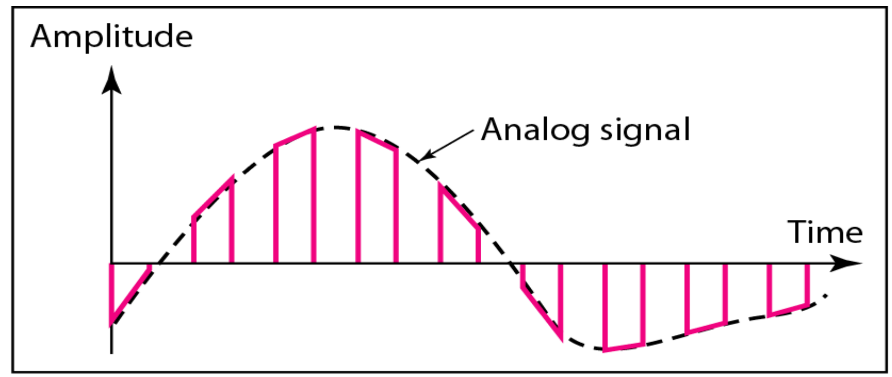


The first step in PCM is sampling.

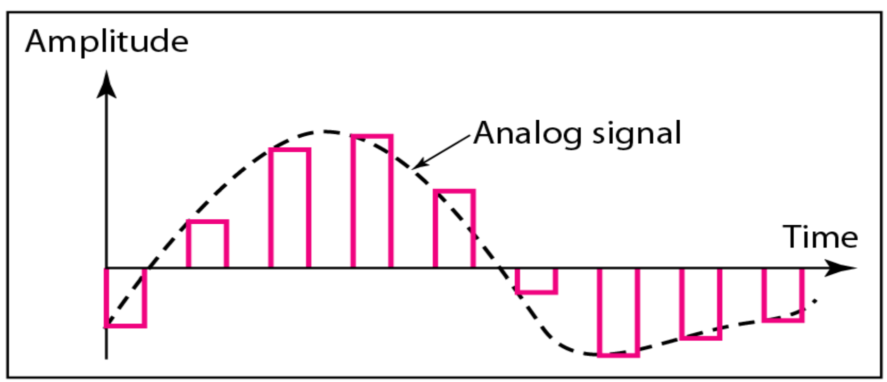
- Sampling is the reduction of continuous time signal to a discrete time signal.
- The analog signal is sampled every T s , where T s is the sample interval or period. The inverse of the sampling interval is called the sampling rate or sampling frequency and denoted by fs, where fs = 1/Ts.
- There are three sampling methods: ideal, natural, and flat-top-as shown in Fig.



a. Ideal sampling



b. Natural sampling



c. Flat-top sampling

In ***ideal sampling***, pulses from the analog signal are sampled. This is an ideal sampling method and cannot be easily implemented.

In natural sampling, a high-speed switch is turned on for only the small period of time when the sampling occurs.
The result is a sequence of samples that retains the shape of the analog signal.

The most common sampling method, called sample and hold, however, creates ***flat-top samples*** by using a circuit.

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

What can we get from this:

1. we can sample a signal only if the signal is band-limited

2. the sampling rate must be at least 2 times the highest frequency, not the bandwidth

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

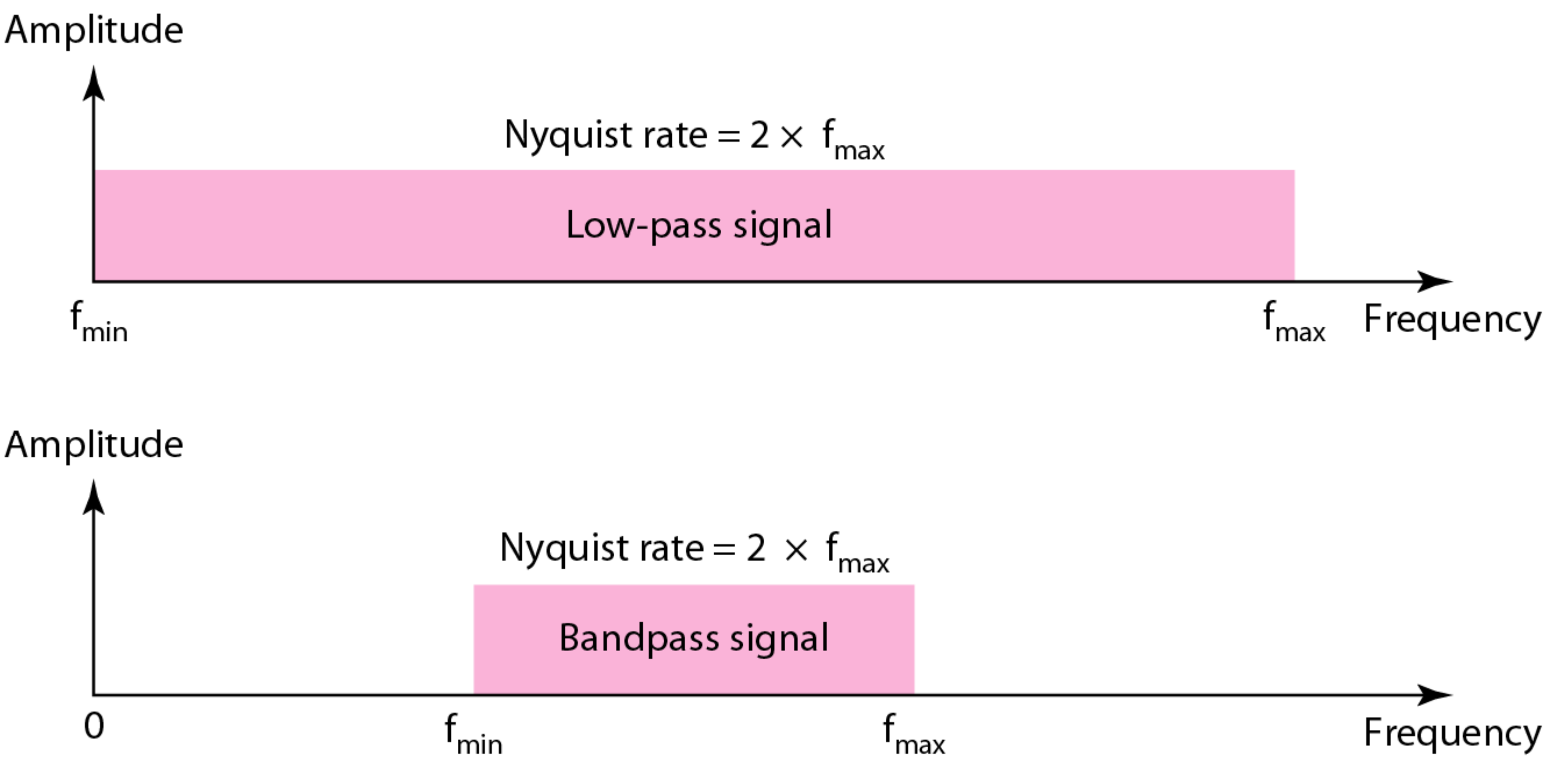


Figure 4.24 *Recovery of a sampled sine wave for different sampling rates*

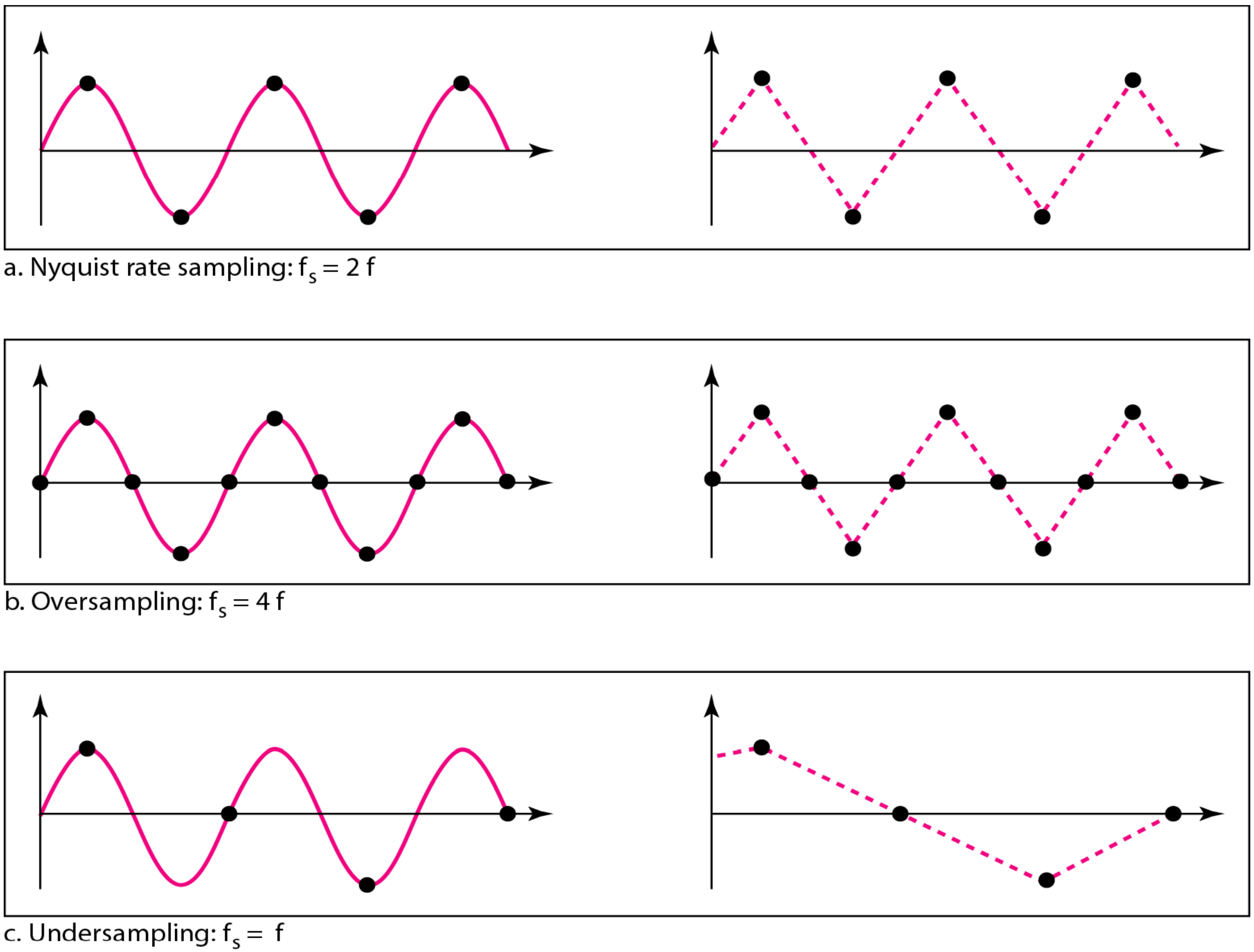
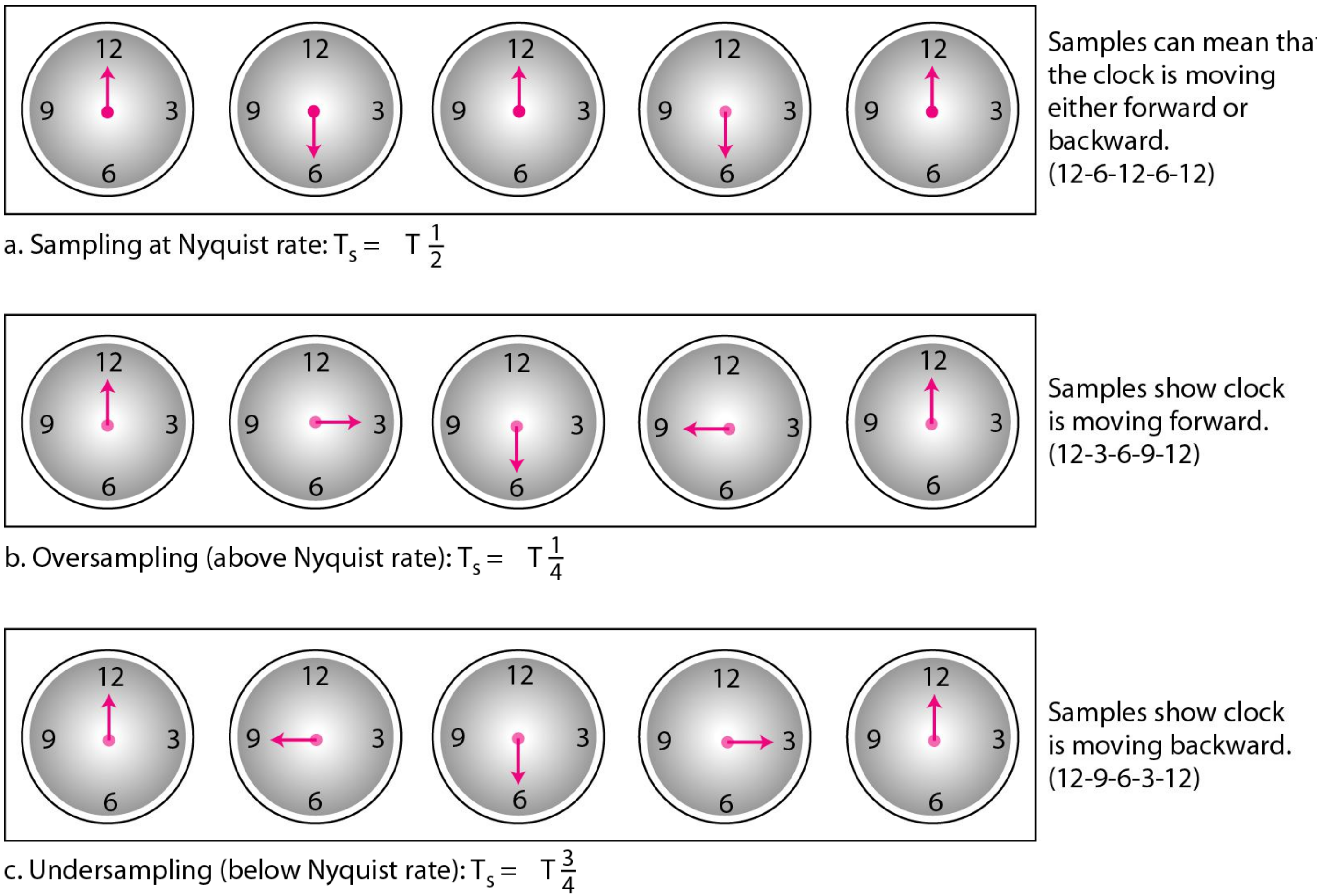


Figure 4.25 *Sampling of a clock with only one hand*



Example

An example related 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

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.

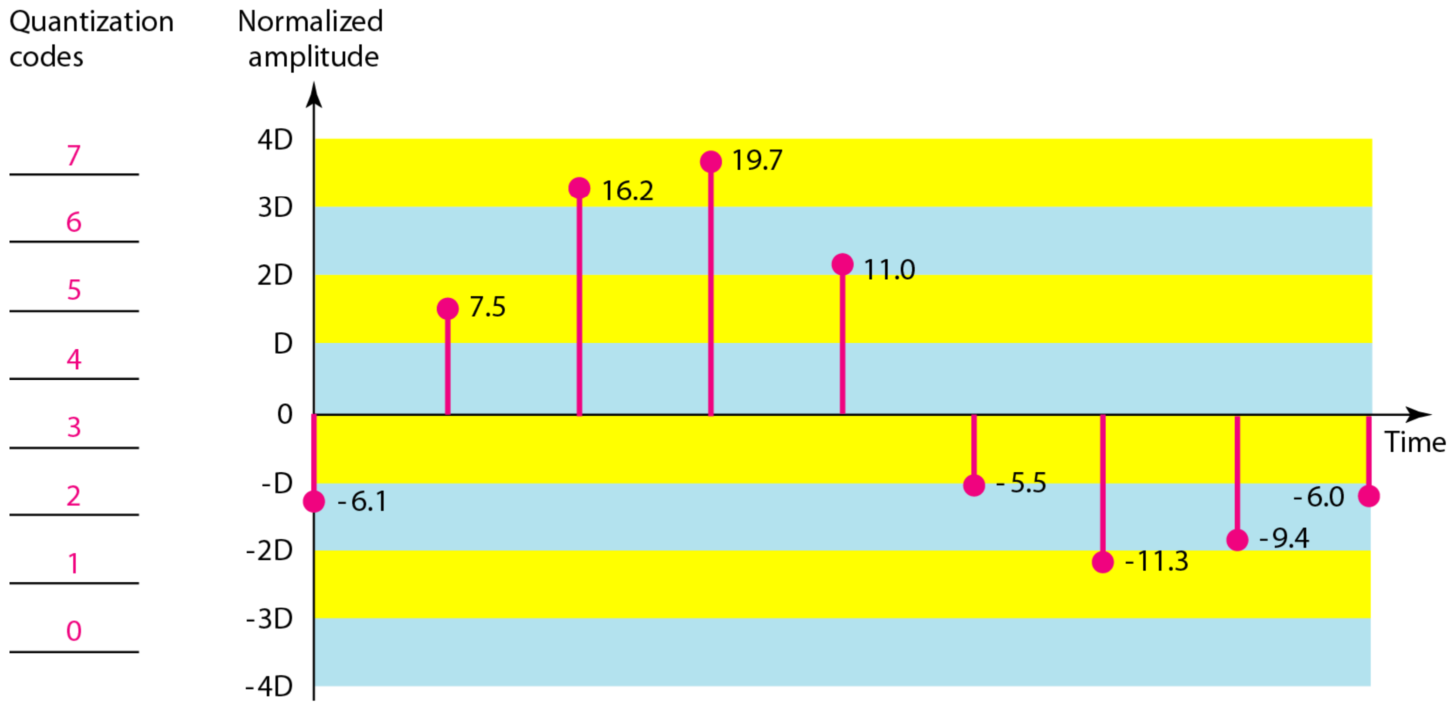
Example

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.

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

Contribution of the quantization error to SNR_{db}

$SNR_{db} = 6.02 n_b + 1.76$ dB

n_b: bits per sample (related to the number of level L)

What is the SNR_{dB} in the example of Figure 4.26?

Solution

We have eight levels and 3 bits per sample, so

$$SNR_{dB} = 6.02 \times 3 + 1.76 = 19.82 \text{ dB}$$

Increasing the number of levels increases the SNR.

Example

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

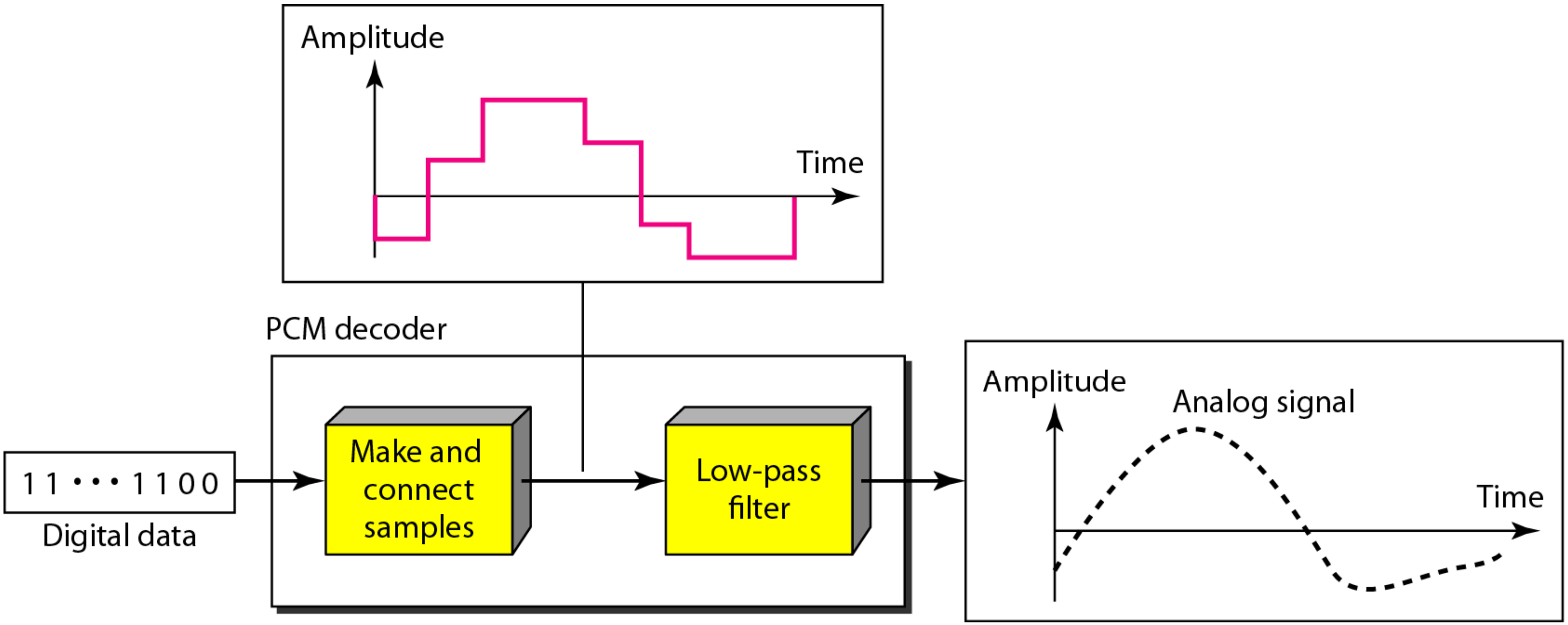
Solution

We can calculate the number of bits as

$$\boxed{\mathrm{SNR_{dB}} = 6.02n_b + 1.76 = 40 \longrightarrow n = 6.35}$$

Telephone companies usually assign 7 or 8 bits per sample.

PCM decoder: recovers the original signal



The minimum bandwidth of the digital signal is n_b times greater than the bandwidth of the analog signal.

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

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.

DM **(delta modulation) finds the change from the previous sample**

Next bit is 1, if amplitude of the analog signal is larger

Next bit is 0, if amplitude of the analog signal is smaller

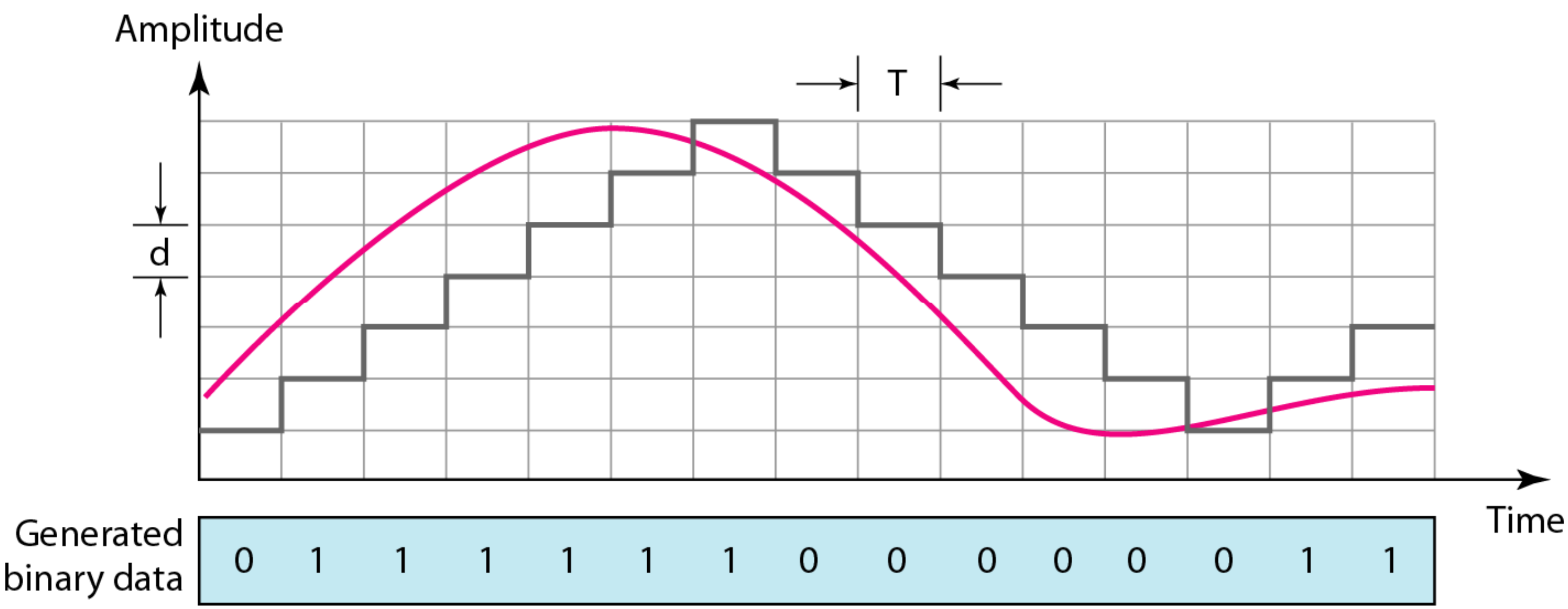


Figure 4.29 ***Delta modulation components***

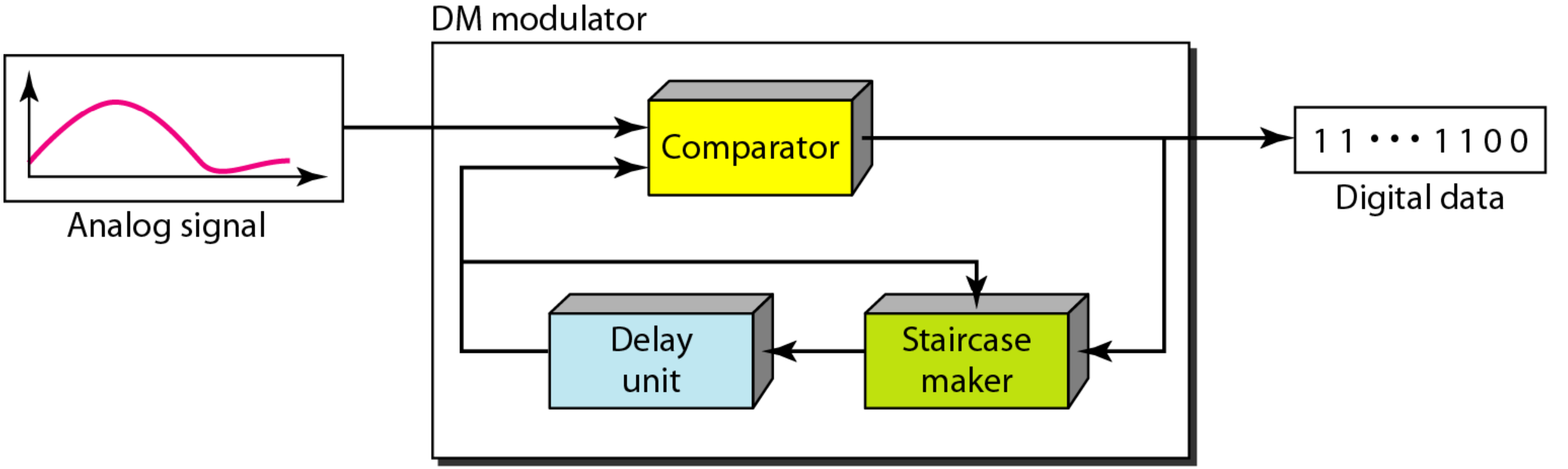
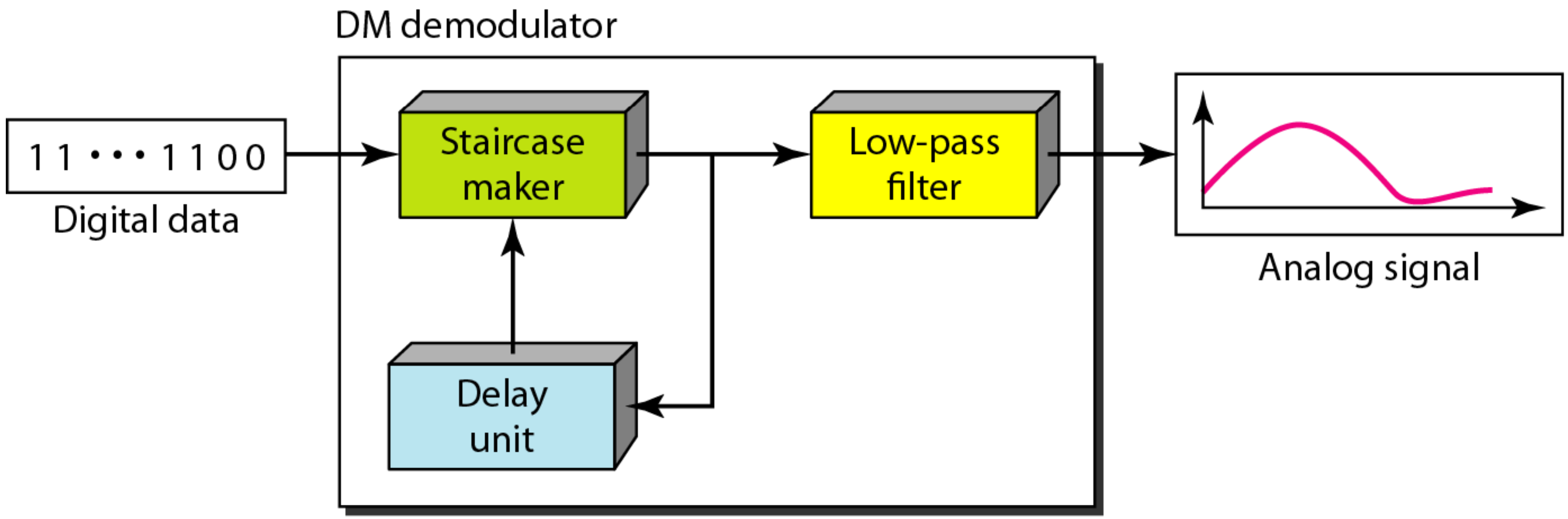


Figure 4.30 *Delta demodulation components*



4-3 TRANSMISSION MODES

1. The transmission of binary data across a link can be accomplished in either parallel or serial mode.
2. In parallel mode, multiple bits are sent with each clock tick.
3. In serial mode, 1 bit is sent with each clock tick.
4. there are three subclasses of serial transmission: asynchronous, synchronous, and isochronous.

Figure 4.31 *Data transmission and modes*

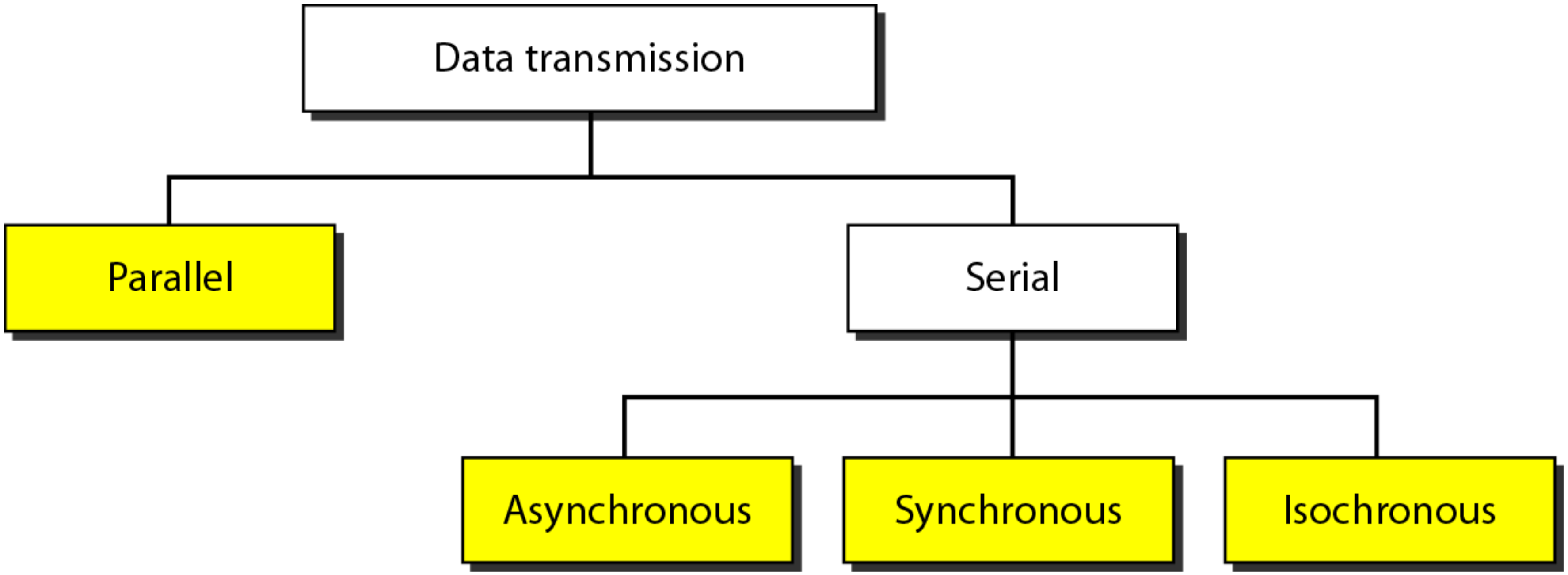


Figure 4.32 *Parallel transmission*

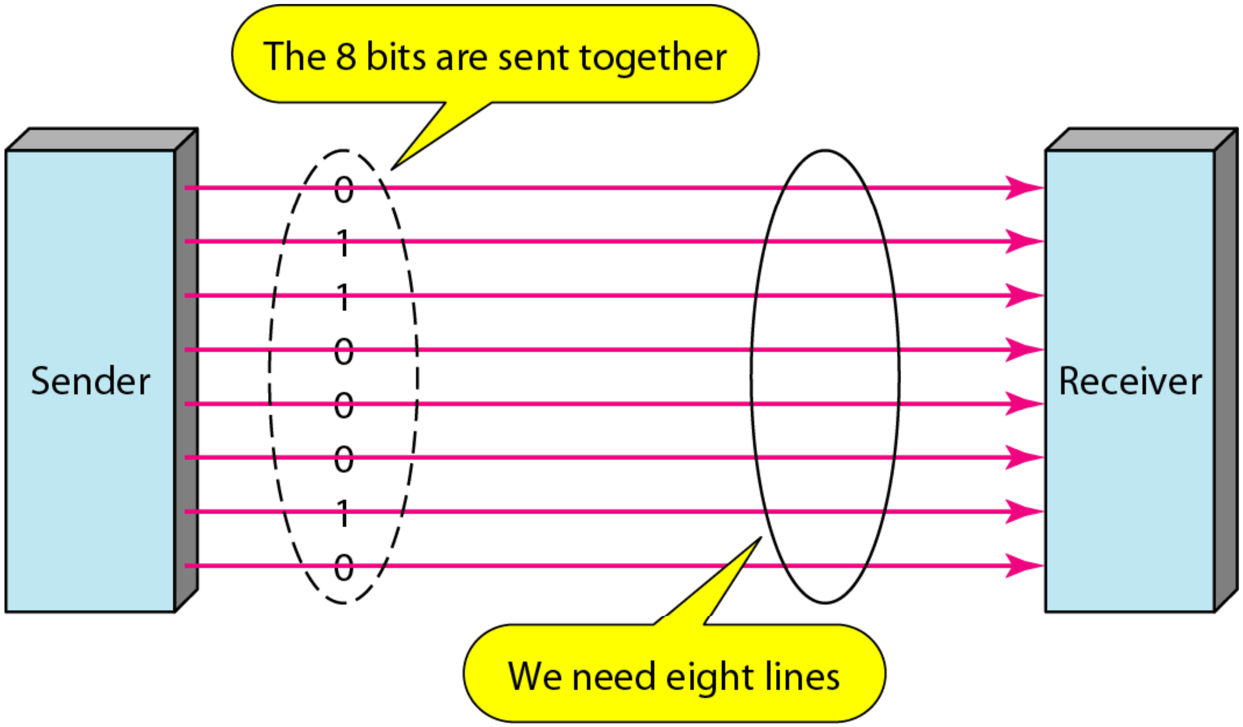
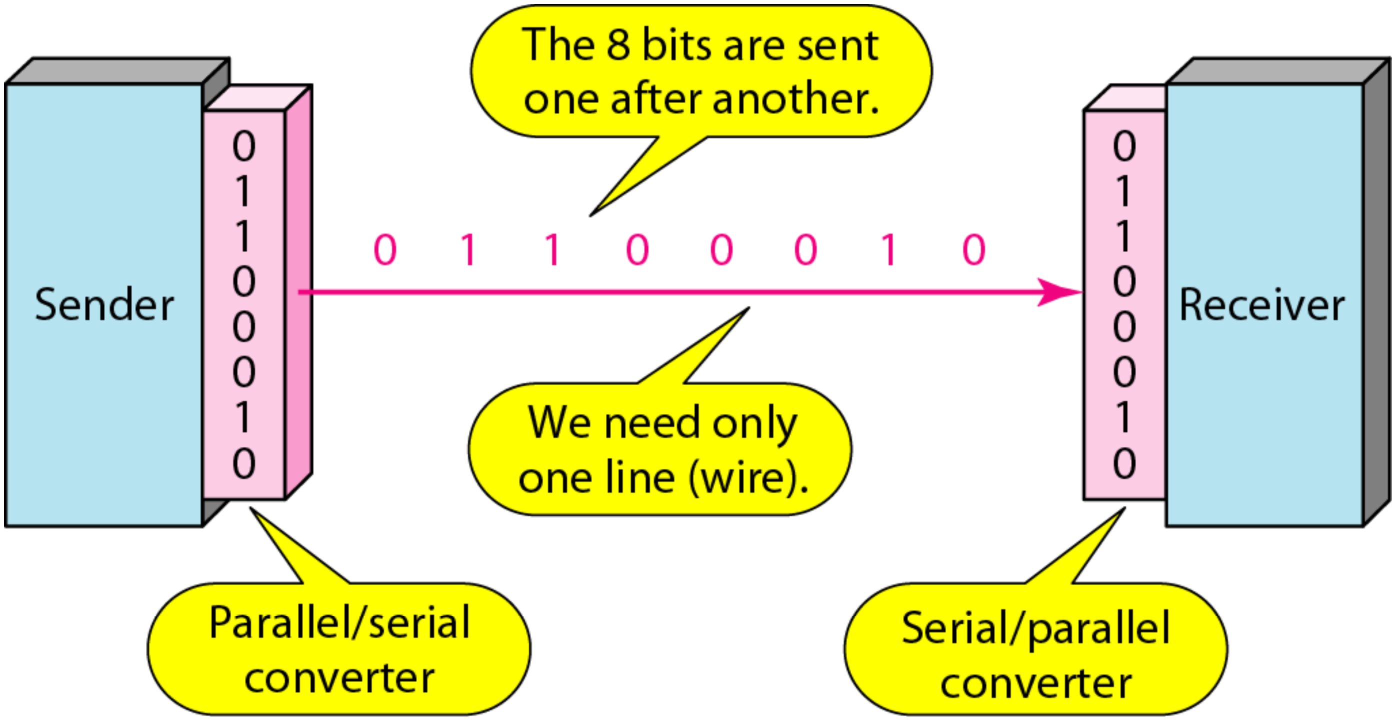
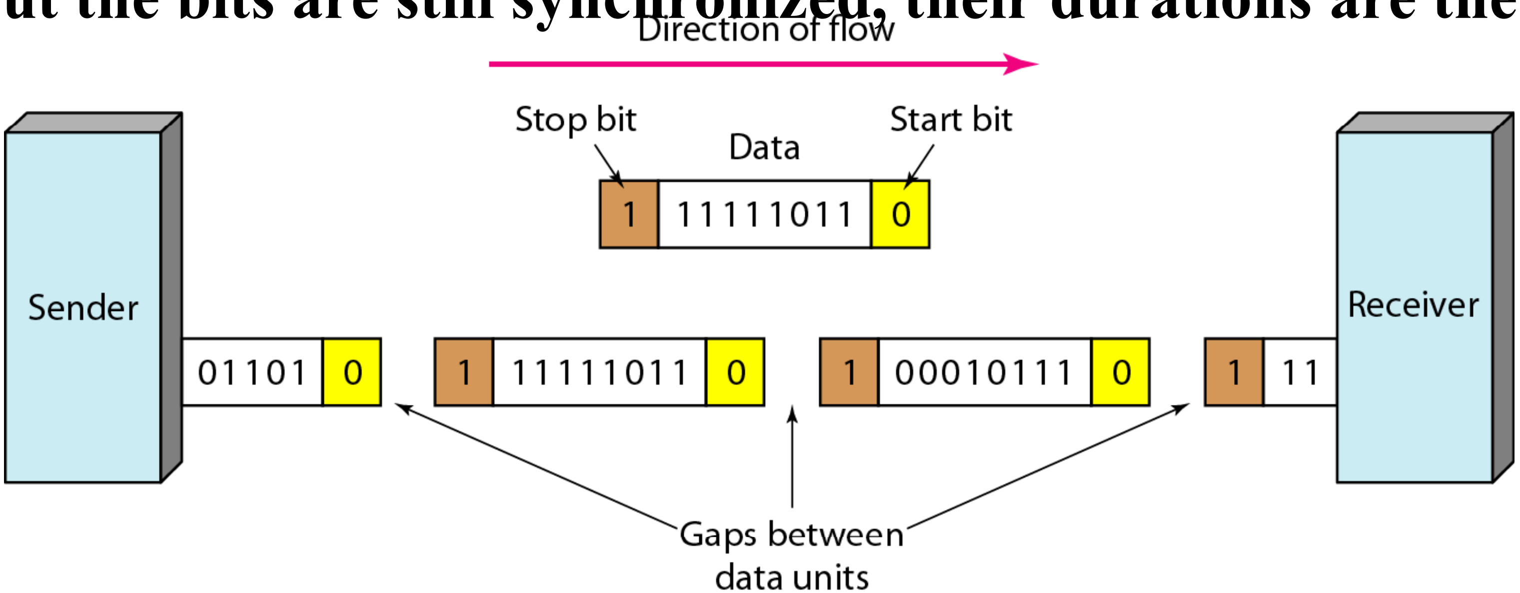


Figure 4.33 ***Serial transmission***



Asynchronous transmission

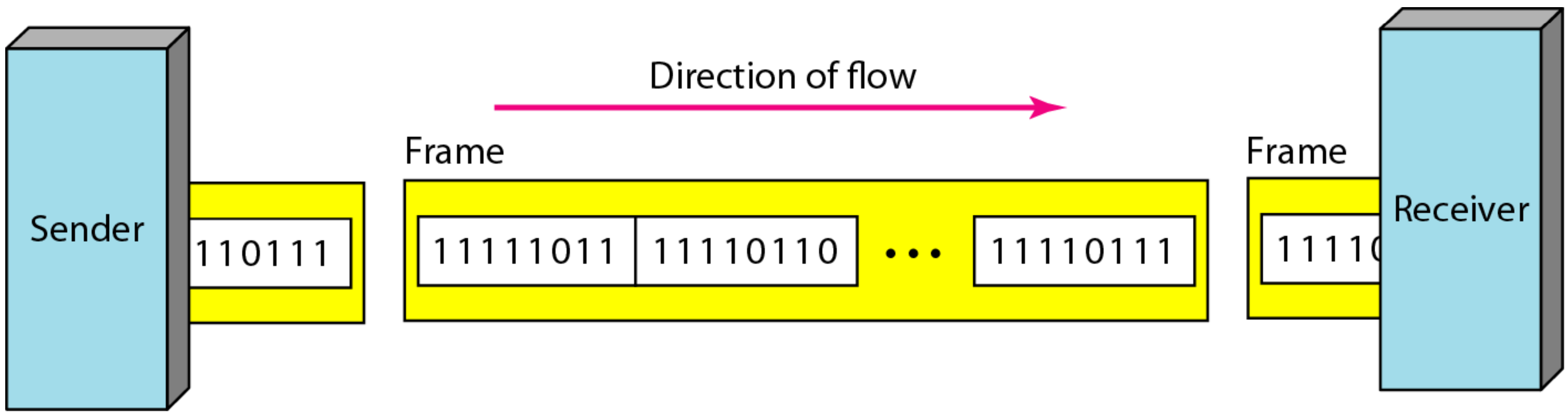
1. We send 1 start bit (0) at the beginning and 1 or more stop bits (1s) at the end of each byte.
2. There may be a gap between each byte.
3. Extra bits and gaps are used to alert the receiver, and allow it to synchronize with the data stream.
4. Asynchronous here means "asynchronous at the byte level," but the bits are still synchronized, their durations are the same.



For example, the connection of a keyboard to a computer is a natural application for asynchronous transmission.
A user types only one character at a time, types extremely slowly in data processing terms, and leaves unpredictable gaps of time between each character.

Synchronous transmission

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.



The advantage of synchronous transmission is speed. With no extra bits or gaps to introduce at the sending end and remove at the receiving end, and, by extension, with fewer bits to move across the link, synchronous transmission is faster than asynchronous transmission. For this reason, it is more useful for high-speed applications such as the transmission of data from one computer to another. Byte synchronization is accomplished in the data link layer.

Isochronous

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 flames, there should be no delays between frames. For this type of application, 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.