Analog Transmission

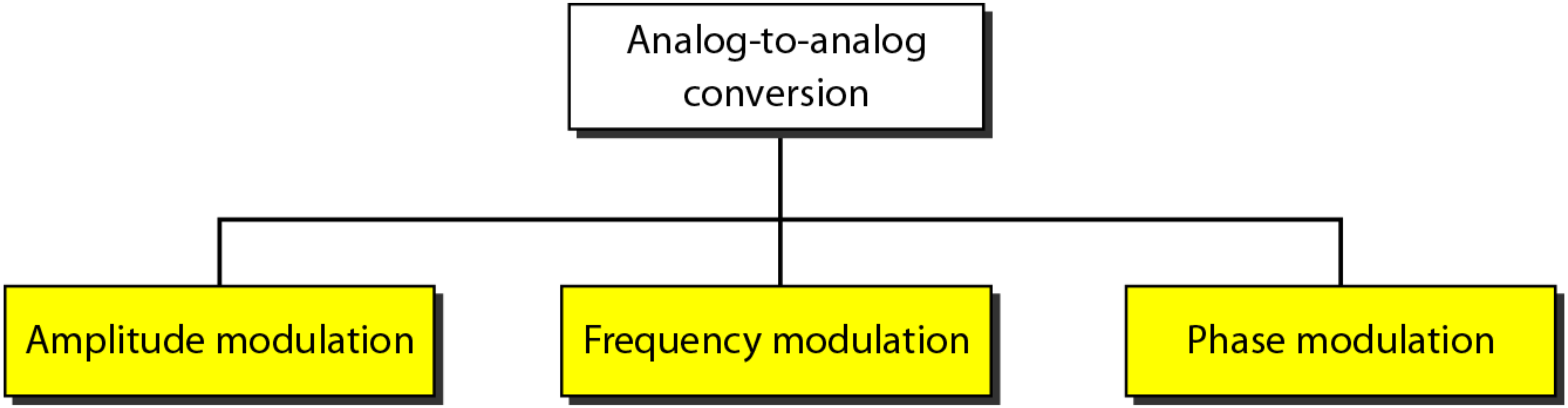
ANALOG AND ANALOG

Analog-to-analog conversion is the representation of analog information by an analog signal. One may ask why we need to modulate an analog signal; it is already analog. Modulation is needed if the medium is bandpass in nature or if only a bandpass channel is available to us.

Topics discussed in this section:

- **Amplitude Modulation**
- **Frequency Modulation**
- **Phase Modulation**

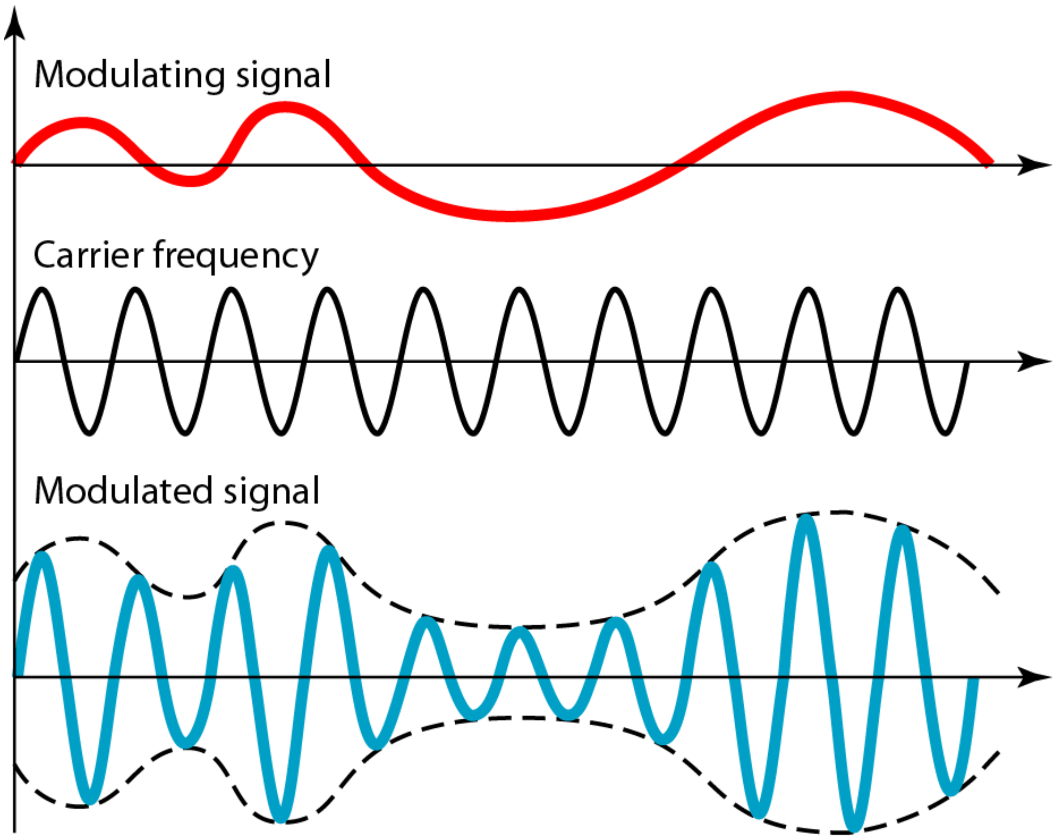
Types of analog-to-analog modulation

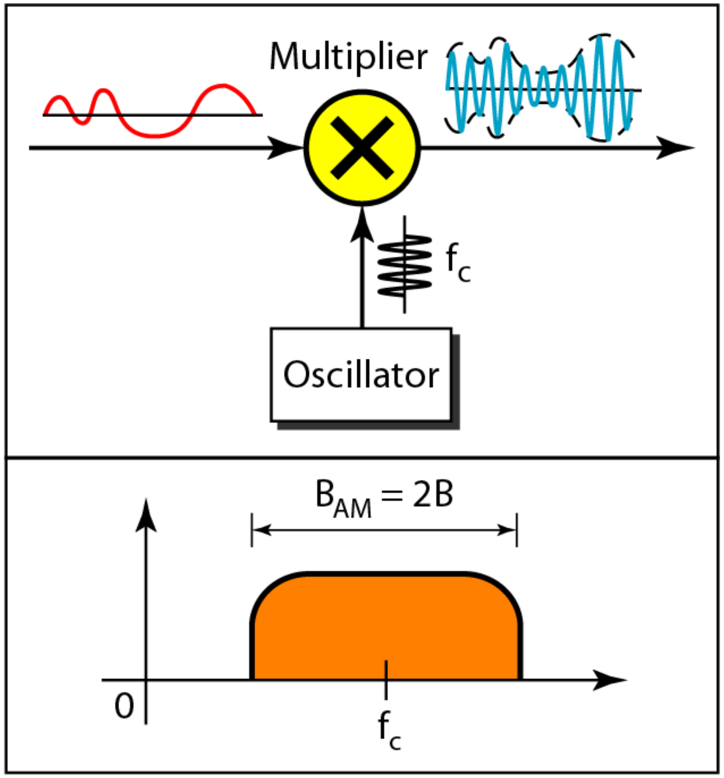


Amplitude Modulation

- A carrier signal is modulated only in amplitude value
- The modulating signal is the envelope of the carrier
- The required bandwidth is 2B, where B is the bandwidth of the modulating signal

Amplitude modulation





Note

The total bandwidth required for AM can be determined from the bandwidth of the audio signal: $B_{AM} = 2B$.

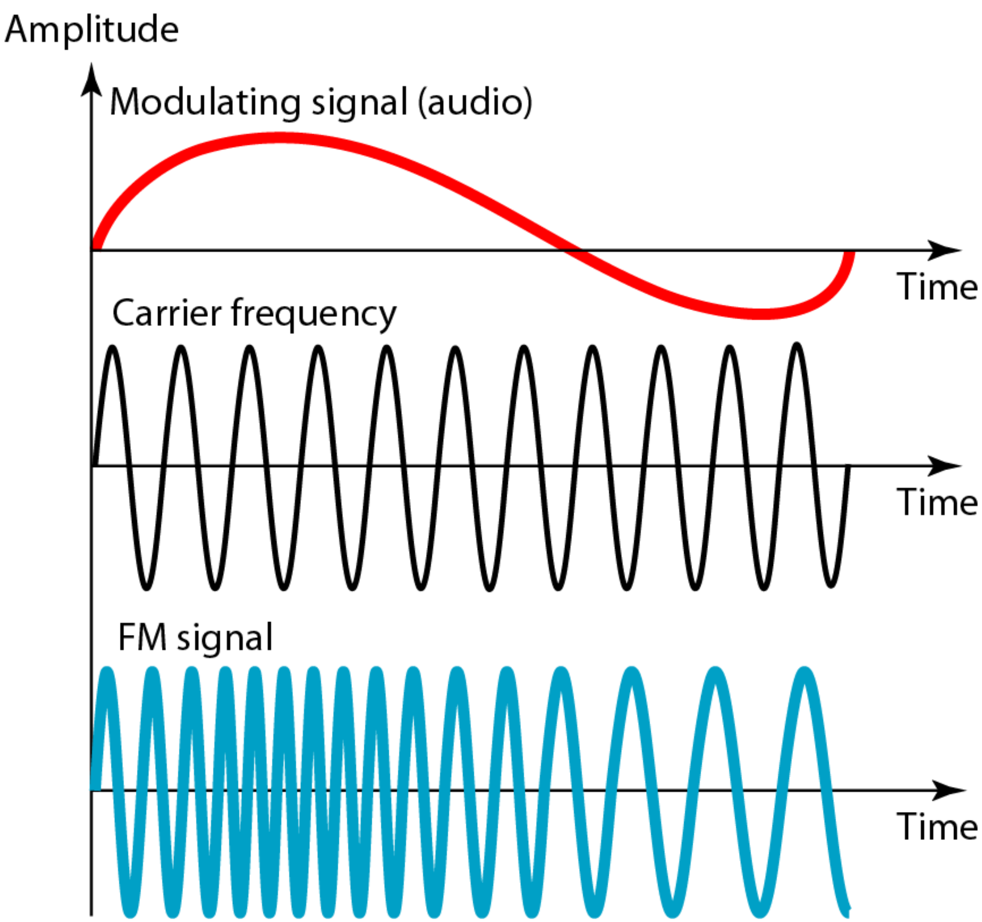
Frequency Modulation

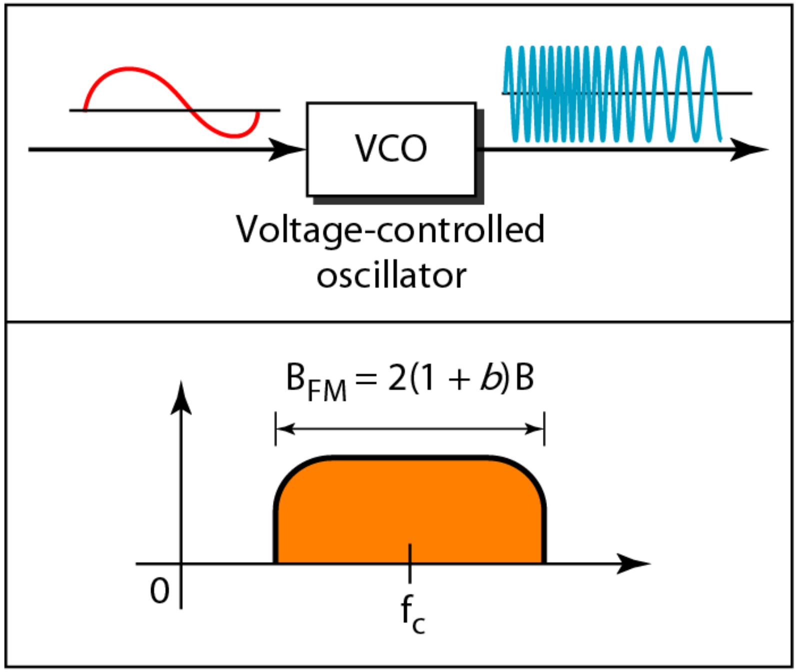
- The modulating signal changes the freq. f_c of the carrier signal
- The bandwidth for FM is high
- It is approx. 10x the signal frequency

Note

The total bandwidth required for FM can be determined from the bandwidth of the audio signal: $B_{FM} = 2(1 + \beta)B$. Where β is usually 4.

Frequency modulation

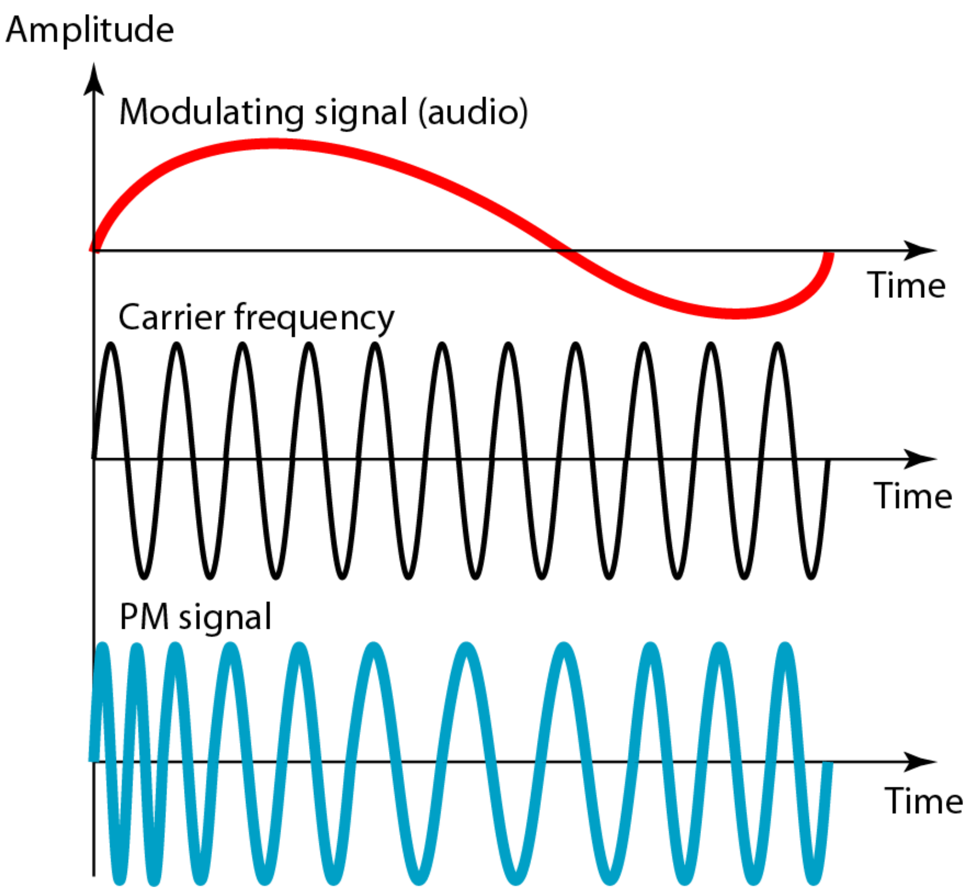


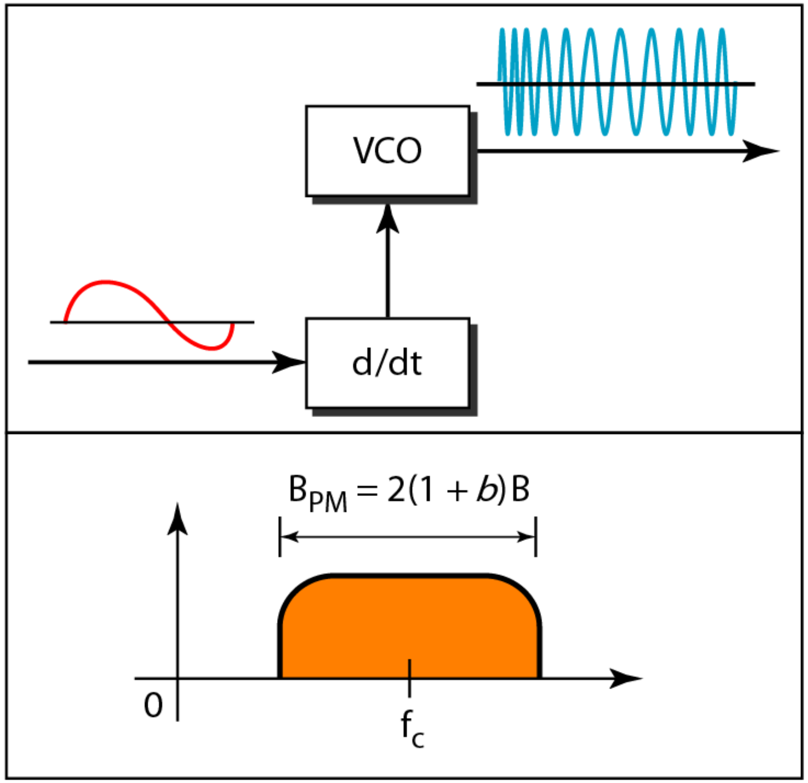


Phase Modulation (PM)

- The modulating signal only changes the phase of the carrier signal.
- The phase change manifests itself as a frequency change but the instantaneous frequency change is proportional to the derivative of the amplitude.
- The bandwidth is higher than for AM.

Phase modulation





Note

The total bandwidth required for PM can be determined from the bandwidth and maximum amplitude of the modulating signal:

$$B_{PM} = 2(1 + \beta)B.$$

Where $\beta = 2$ most often.

POLL 1

- Which of the following has the maximum Bandwidth

a) AM

b) FM

c) PM

d) None

POLL 2

- PM can be achieved using

a) VCO only

b) Derivative + VCO

c) Multiplier

d) None

Analog Transmission

DIGITAL-TO-ANALOG CONVERSION

Digital-to-analog conversion is the process of changing one of the characteristics of an analog signal based on the information in digital data.

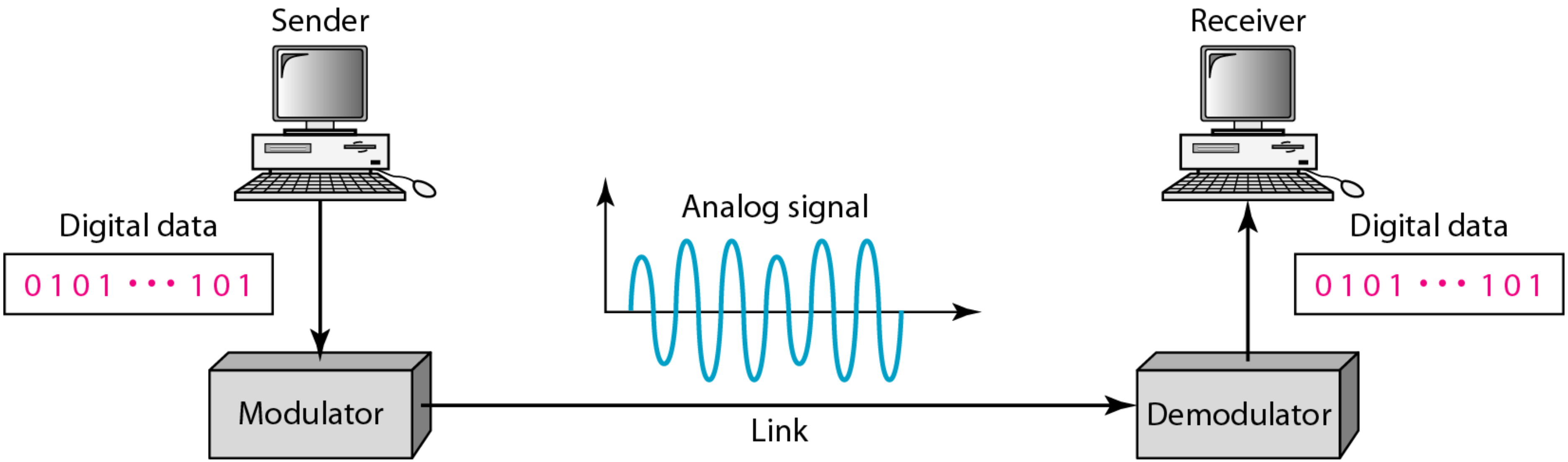
Topics discussed in this section:

- **Aspects of Digital-to-Analog Conversion**
- **Amplitude Shift Keying**
- **Frequency Shift Keying**
- **Phase Shift Keying**
- **Quadrature Amplitude Modulation**

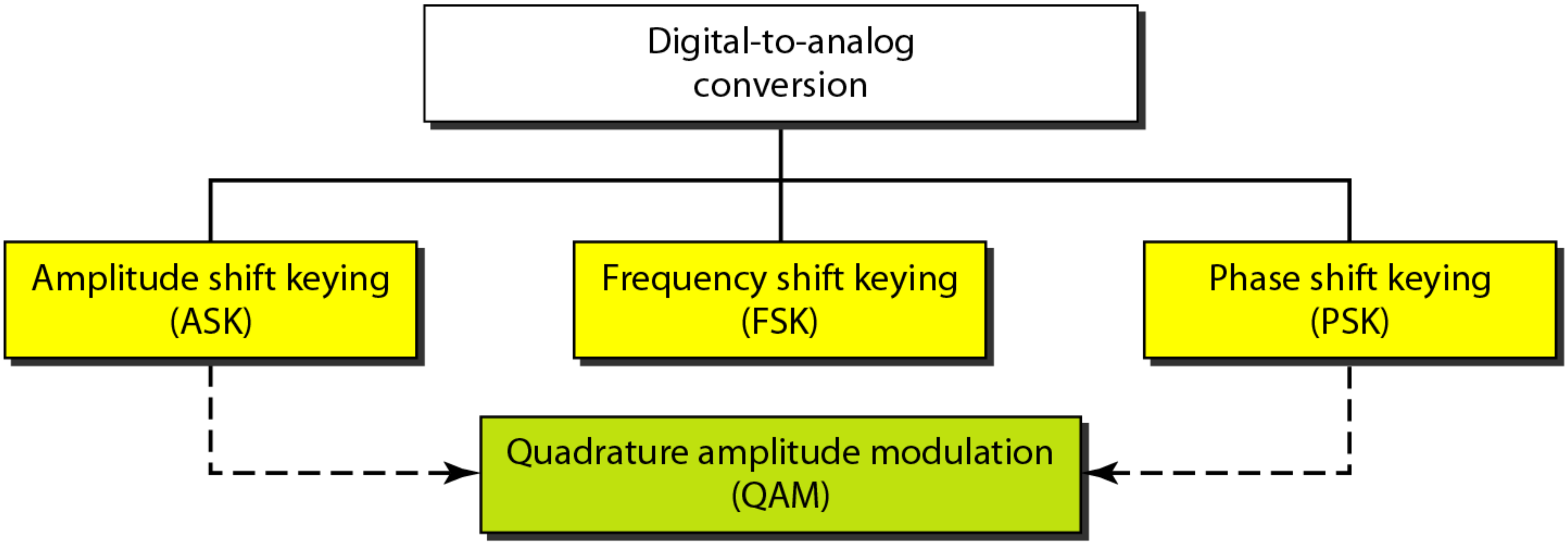
Digital to Analog Conversion

- Digital data needs to be carried on an analog signal.
- A carrier signal (frequency f_c) performs the function of transporting the digital data in an analog waveform.
- The analog carrier signal is manipulated to uniquely identify the digital data being carried.

Digital-to-analog conversion



Types of digital-to-analog conversion



POLL 3

- QAM is a combination of

a) ASK + PSK

b) ASK + FSK

c) PSK + FSK

d) none

Note

Bit rate, N, is the number of bits per second (bps). Baud rate is the number of signal elements per second (bauds).

In the analog transmission of digital data, the signal or baud rate is less than or equal to the bit rate.

S=Nx1/r bauds

Where r is the number of data bits per signal element.

Solution

An analog signal carries 4 bits per signal element. If 1000 signal elements are sent per second, find the bit rate.

Solution

In this case, r = 4, S = 1000, and N is unknown. We can find the value of N from

$$S = N \times \frac{1}{r} \qquad \text{or} \qquad N = S \times r = 1000 \times 4 = 4000 \text{ bps}$$

An analog signal has a bit rate of 8000 bps and a baud rate of 1000 baud. How many data elements are carried by each signal element? How many signal elements do we need?

Example

An analog signal has a bit rate of 8000 bps and a baud rate of 1000 baud. How many data elements are carried by each signal element? How many signal elements do we need?

Solution

In this example, S = 1000, N = 8000, and r and L are unknown. We find first the value of r and then the value of L.

$$S = N \times \frac{1}{r} \quad \Rightarrow \quad r = \frac{N}{S} = \frac{8000}{1000} = 8 \text{ bits/baud}$$

$$r = \log_2 L \quad \Rightarrow \quad L = 2^r = 2^8 = 256$$

Amplitude Shift Keying (ASK)

- ASK is implemented by changing the amplitude of a carrier signal to reflect amplitude levels in the digital signal.
- For example: a digital “1” could not affect the signal, whereas a digital “0” would, by making it zero.
- The line encoding will determine the values of the analog waveform to reflect the digital data being carried.

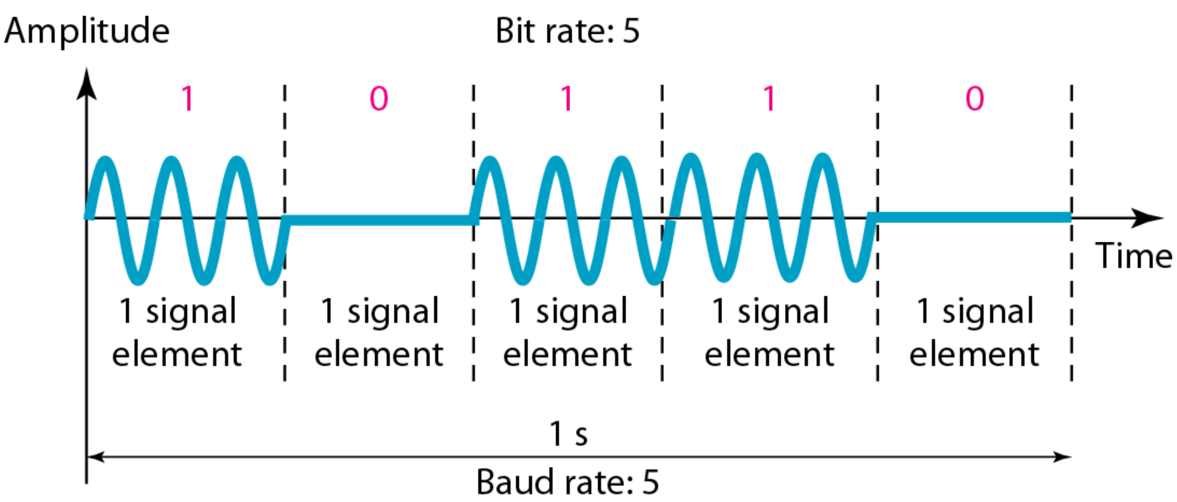
Bandwidth of ASK

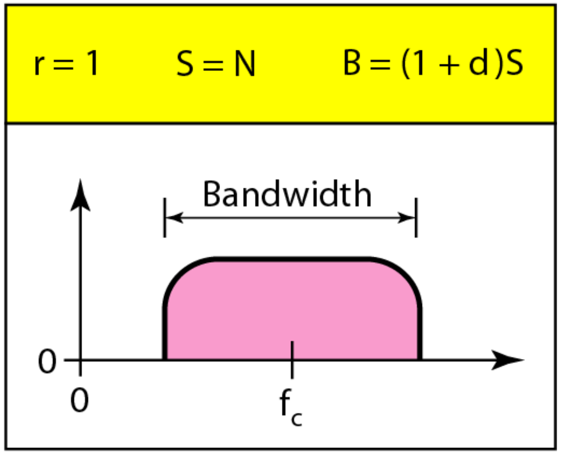
- The bandwidth B of ASK is proportional to the signal rate S.

$$B = (1+d)S$$

- “d” is due to modulation and filtering, lies between 0 and 1.

Binary amplitude shift keying

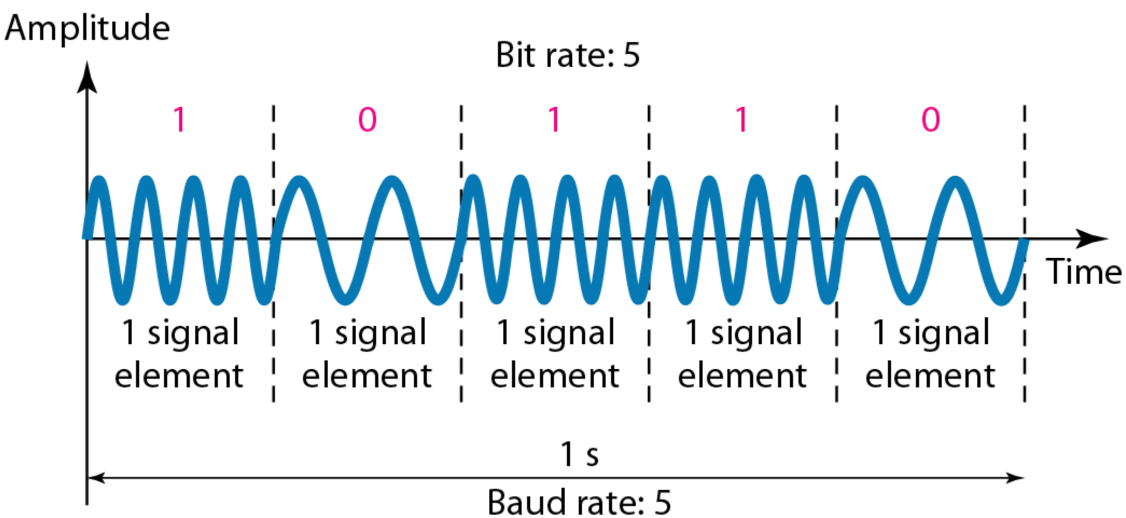


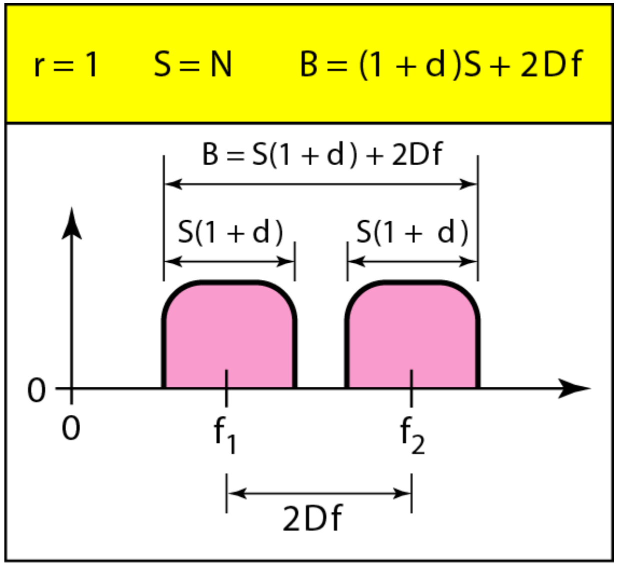


Frequency Shift Keying

- The digital data stream changes the frequency of the carrier signal, f_c.
- For example, a “1” could be represented by $f_1=f_c+\Delta f$, and a “0” could be represented by $f_2=f_c-\Delta f$.

Binary frequency shift keying





Bandwidth of FSK

- If the difference between the two frequencies (f_1 and f_2) is $2\Delta f$, then the required BW B will be:

$$B = (1+d)xS + 2\Delta f$$

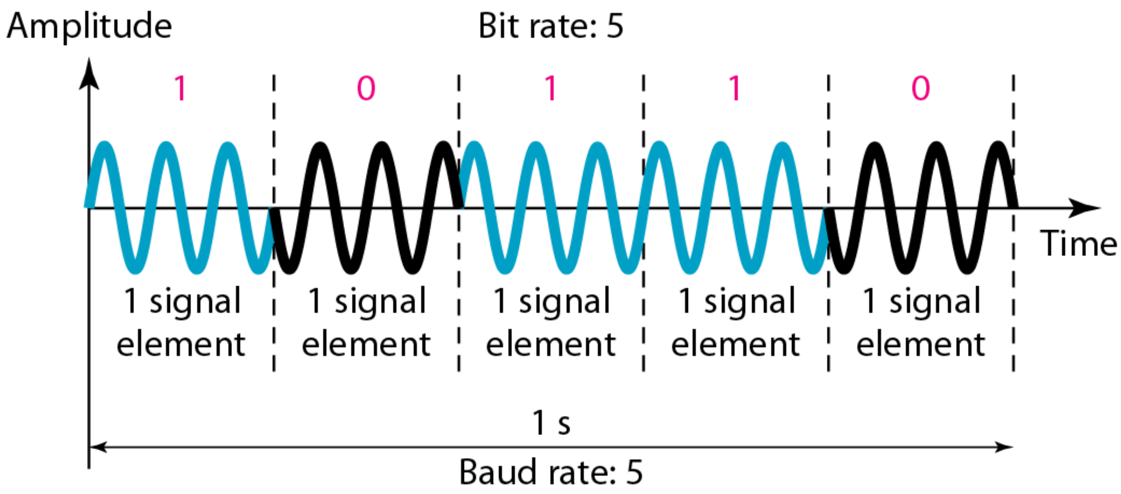
Phase Shift Keying

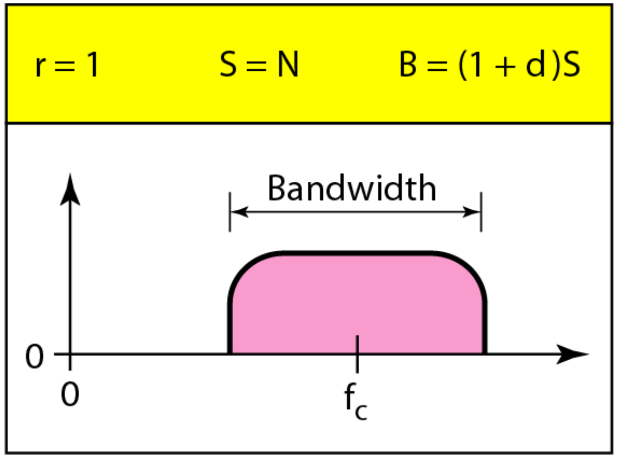
- We vary the phase shift of the carrier signal to represent digital data.
- The bandwidth requirement, B is:

  $$B = (1+d)xS$$

- PSK is much more robust than ASK as it is not that vulnerable to noise, which changes amplitude of the signal.

Binary phase shift keying





POLL 4

BW of binary FSK is

a) S(1+d)

b) S(1+d) + 2Df

c) S(1-d)

d) S(1+d) – 2Df

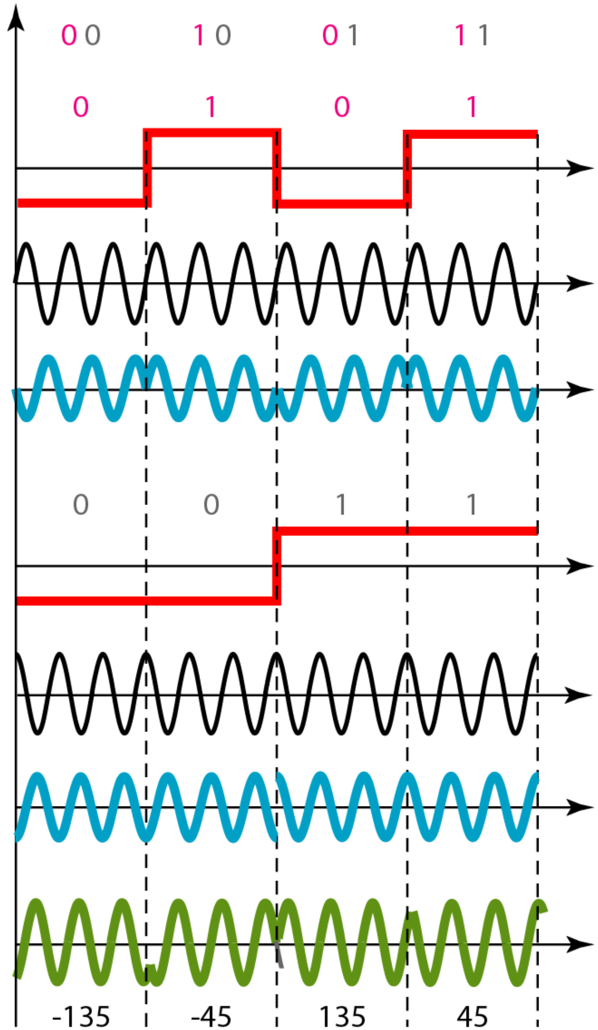
Quadrature PSK

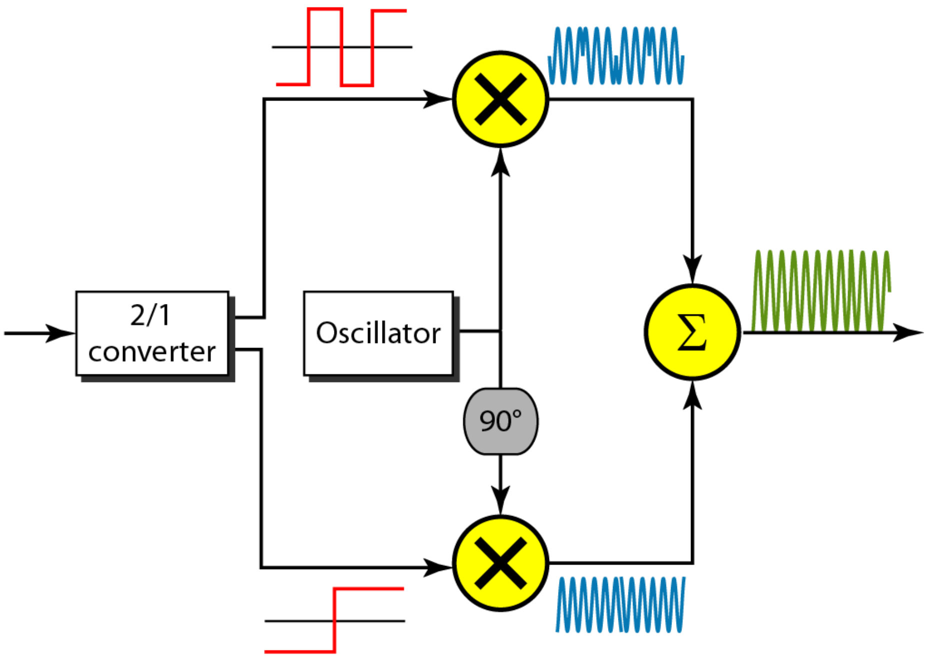
- To increase the bit rate, we can code 2 or more bits onto one signal element.

- In QPSK, we parallelize the bit stream so that every two incoming bits are split up and PSK a carrier frequency. One carrier frequency is phase shifted 90^o from the other - in quadrature.

- The two PSKed signals are then added to produce one of 4 signal elements. L = 4 here.

QPSK and its implementation





Note

Quadrature amplitude modulation is a combination of ASK and PSK.

Digital Transmission

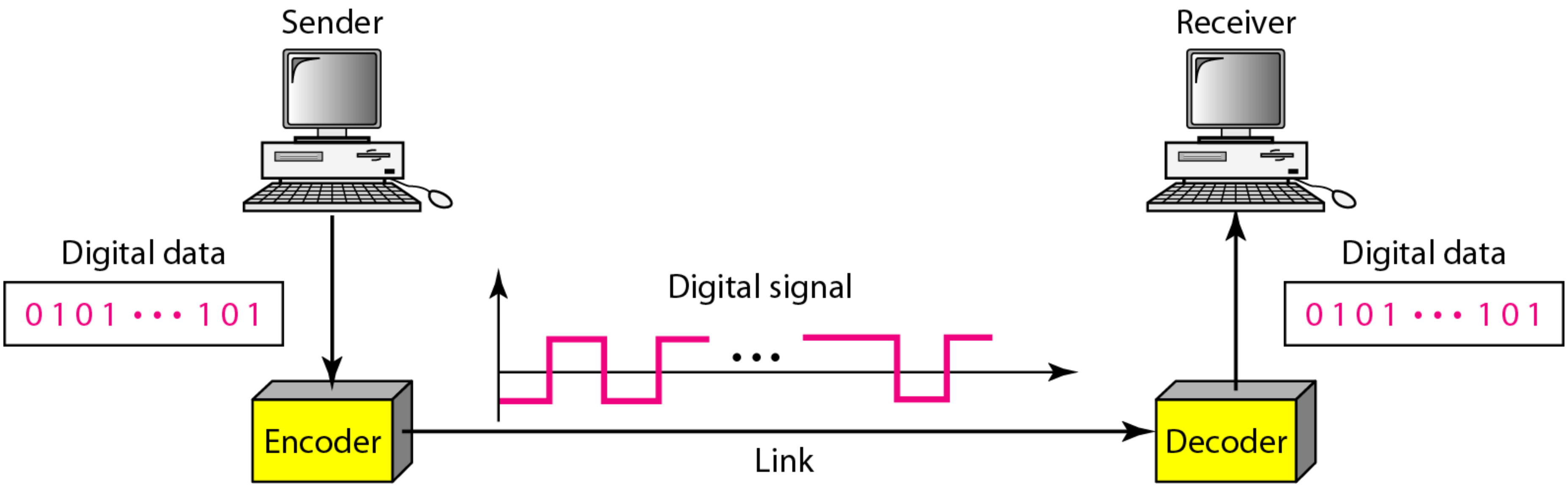
DIGITAL-TO-DIGITAL CONVERSION

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.

Line Coding

- Converting a string of 1’s and 0’s (digital data) into a sequence of signals that denote the 1’s and 0’s.
- For example a high voltage level (+V) could represent a “1” and a low voltage level (0 or -V) could represent a “0”.

Line coding and decoding



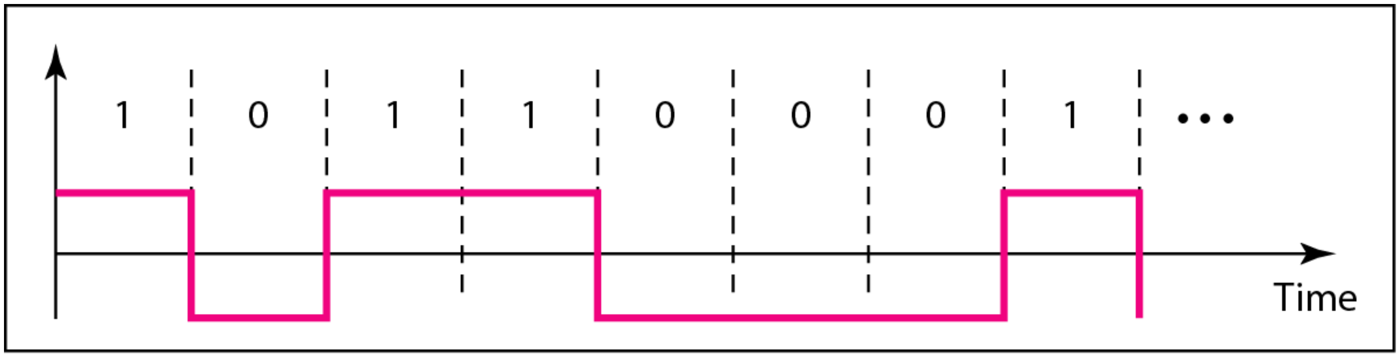
Line encoding C/Cs

- DC components - when the voltage level remains constant for long periods of time, there is an increase in the low frequencies of the signal. Most channels are bandpass and may not support the low frequencies.
- This will require the removal of the dc component of a transmitted signal.

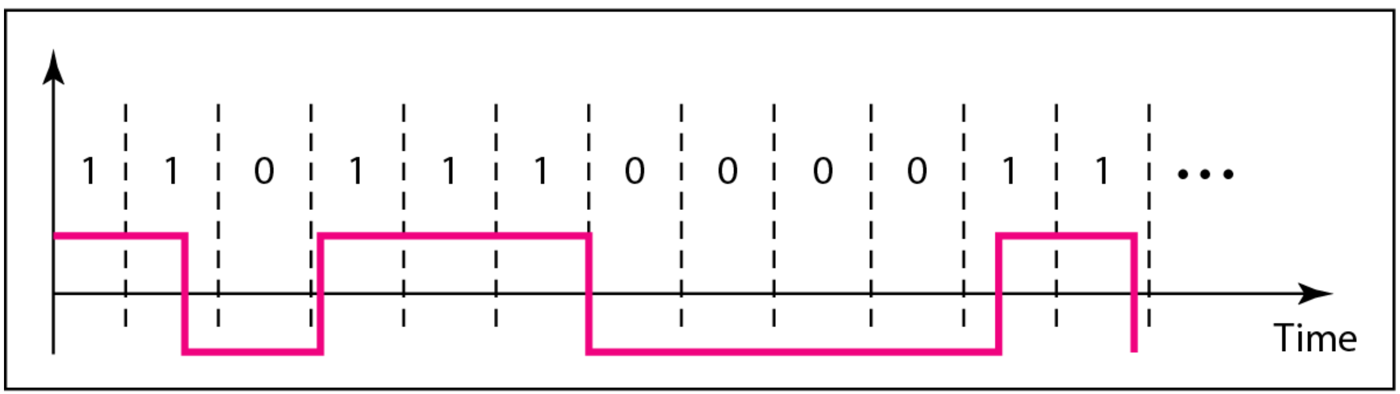
Line encoding C/Cs

- Self synchronization - the clocks at the sender and the receiver must have the same bit interval.
- If the receiver clock is faster or slower it will misinterpret the incoming bit stream.

Figure ***Effect of lack of synchronization***



a. Sent



b. Received

Example

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

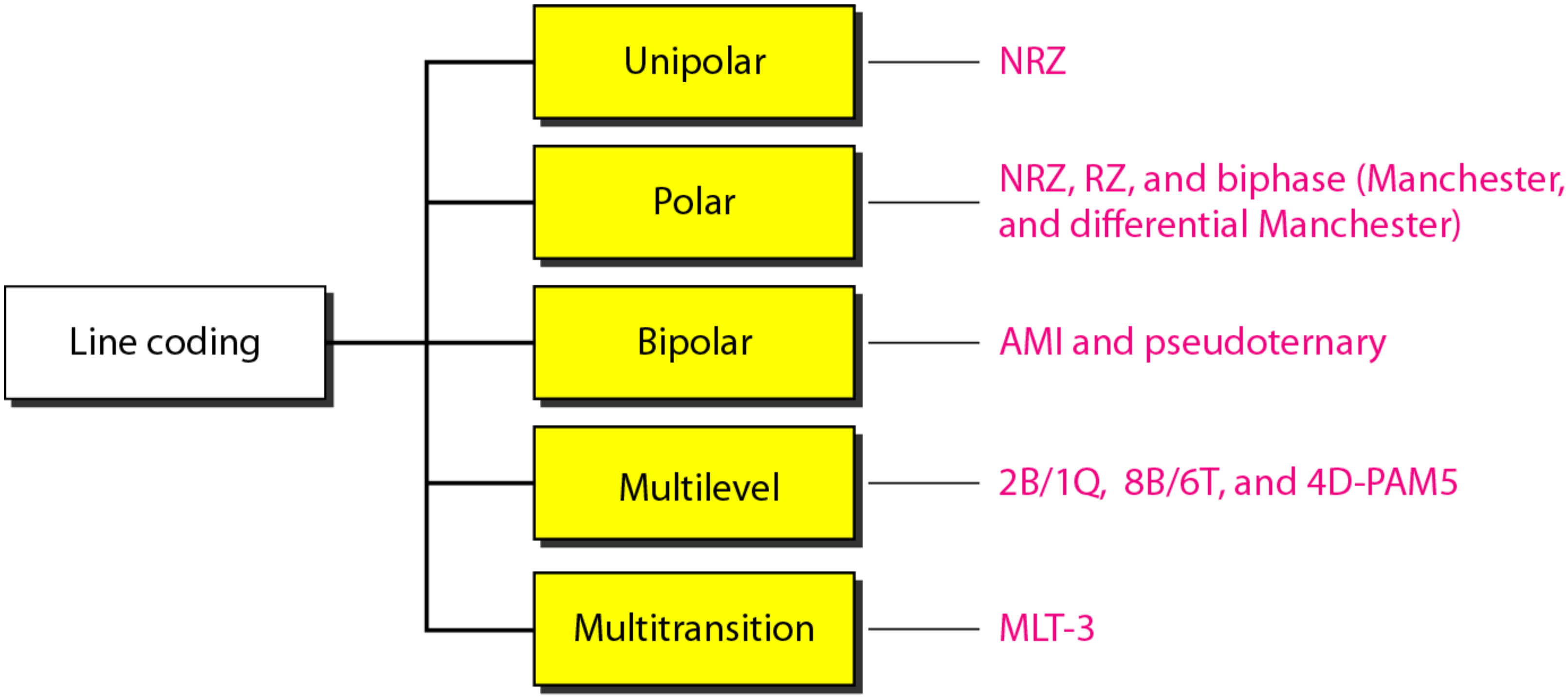
At 1 kbps, the receiver receives 1001 bps instead of 1000 bps.

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 *Line coding schemes*



POLL 5

Which of the following is NOT a type of Line Coding

a) Unipolar

b) Bipolar

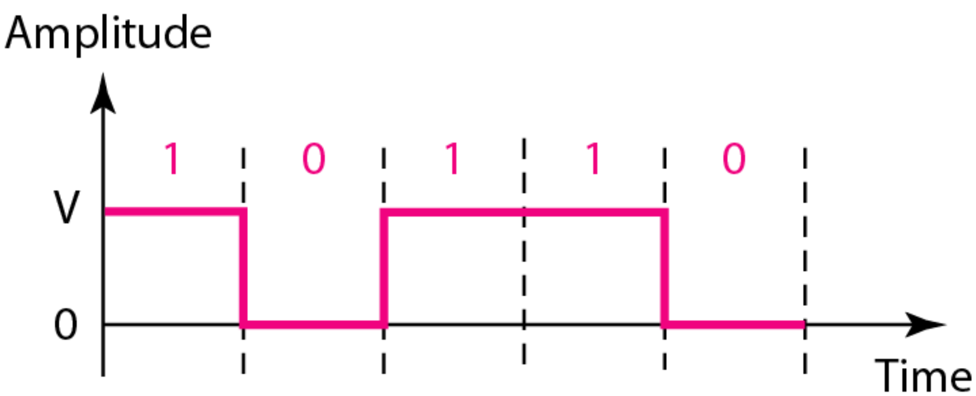
c) Polar

d) Multipolar

Unipolar

- All signal levels are on one side of the time axis - either above or below
- NRZ - Non Return to Zero scheme is an example of this code. The signal level does not return to zero during a symbol transmission.
- Scheme is prone to DC components. It has no synchronization or any error detection. It is simple but costly in power consumption.

Figure *Unipolar NRZ scheme*

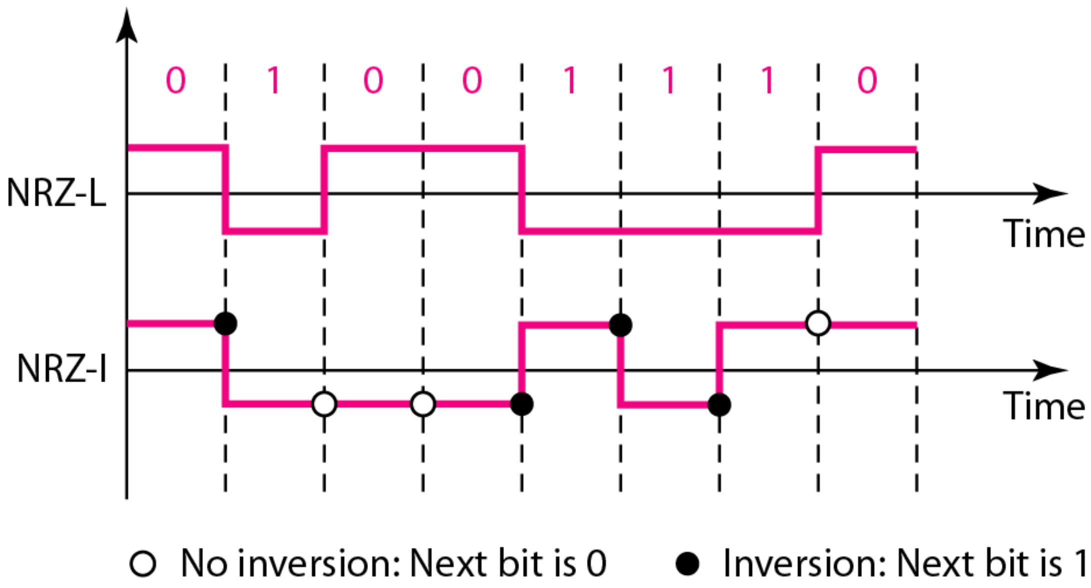


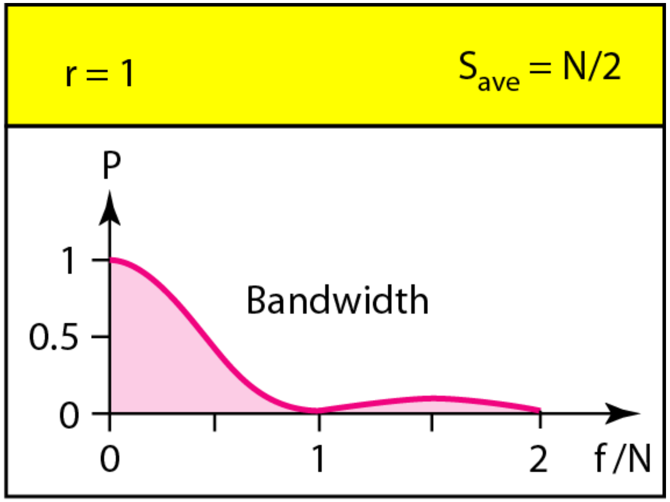
$$\frac{1}{2}V^2 + \frac{1}{2}(0)^2 = \frac{1}{2}V^2$$

Polar - NRZ

- The voltages are on both sides of the time axis.
- Polar NRZ scheme can be implemented with two voltages. E.g. +V for 1 and -V for 0.
- There are two versions:
  - NZR - Level (NRZ-L) - positive voltage for one symbol and negative for the other
  - NRZ - Inversion (NRZ-I) - the change or lack of change in polarity determines the value of a symbol. E.g. a “1” symbol inverts the polarity a “0” does not.

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





Note

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.

Note

NRZ-L and NRZ-I both have an average signal rate of N/2 Bd.

Note

NRZ-L and NRZ-I both have a DC component problem, it is worse for NRZ-L. Both have no self synchronization &no error detection. Both are relatively simple to implement.

POLL 6

NRZ-L and NRZ-I both have

a) DC component problem

b) AC component Problem

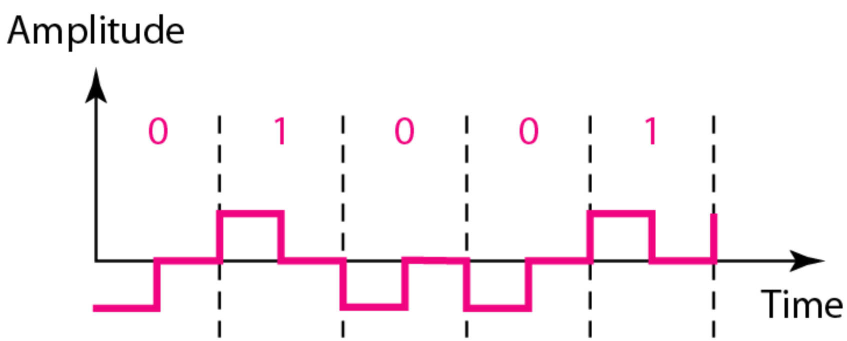
c) Self Synchronization

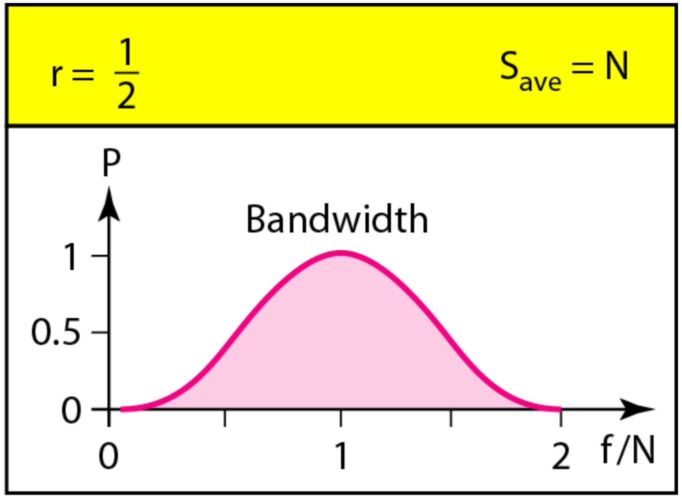
d) Error Detection

Polar - RZ

- The Return to Zero (RZ) scheme uses three voltage values. +, 0, -.
- Each symbol has a transition in the middle. Either from high to zero or from low to zero.
- This scheme has more signal transitions (two per symbol) and therefore requires a wider bandwidth.
- No DC components or baseline wandering.
- Self synchronization - transition indicates symbol value.
- More complex as it uses three voltage level. It has no error detection capability.

Figure *Polar RZ scheme*

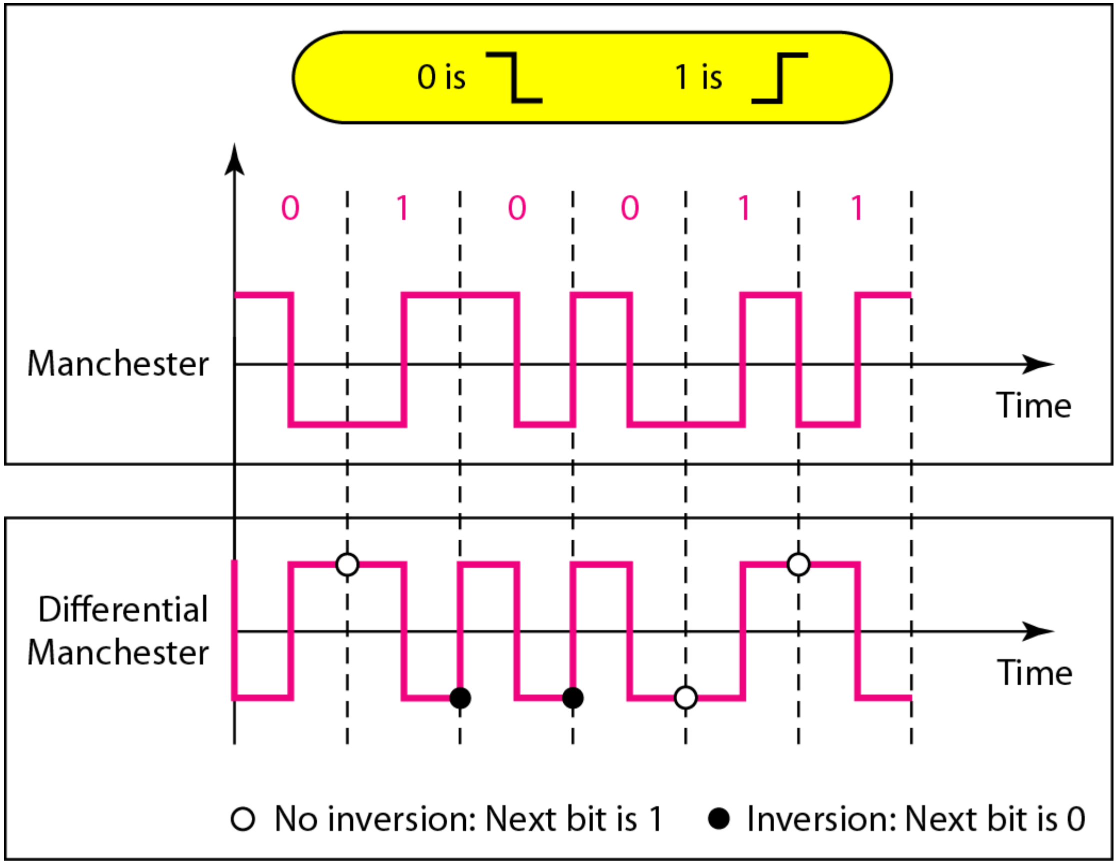


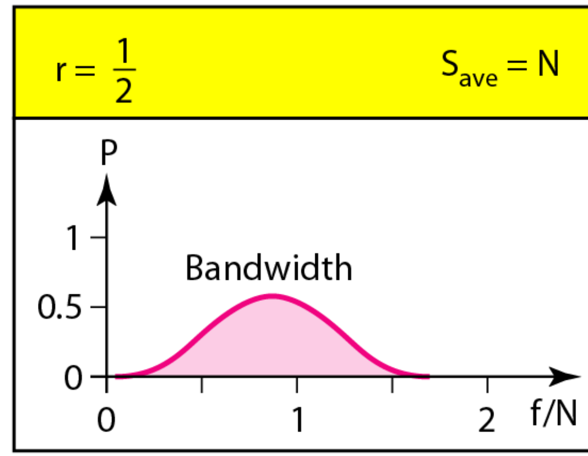


Polar - Biphase: Manchester and Differential Manchester

- Manchester coding consists of combining the NRZ-L and RZ schemes.
  - Every symbol has a level transition in the middle: from high to low or low to high. Uses only two voltage levels.
- Differential Manchester coding consists of combining the NRZ-I and RZ schemes.
  - Every symbol has a level transition in the middle. But the level at the beginning of the symbol is determined by the symbol value. One symbol causes a level change the other does not.

Figure ***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. The is no DC component. None of these codes has error detection.

Digital Transmission

ANALOG-TO-DIGITAL CONVERSION

A digital signal is superior to an analog signal because it is more robust to noise and can easily be recovered, corrected and amplified. For this reason, 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)**

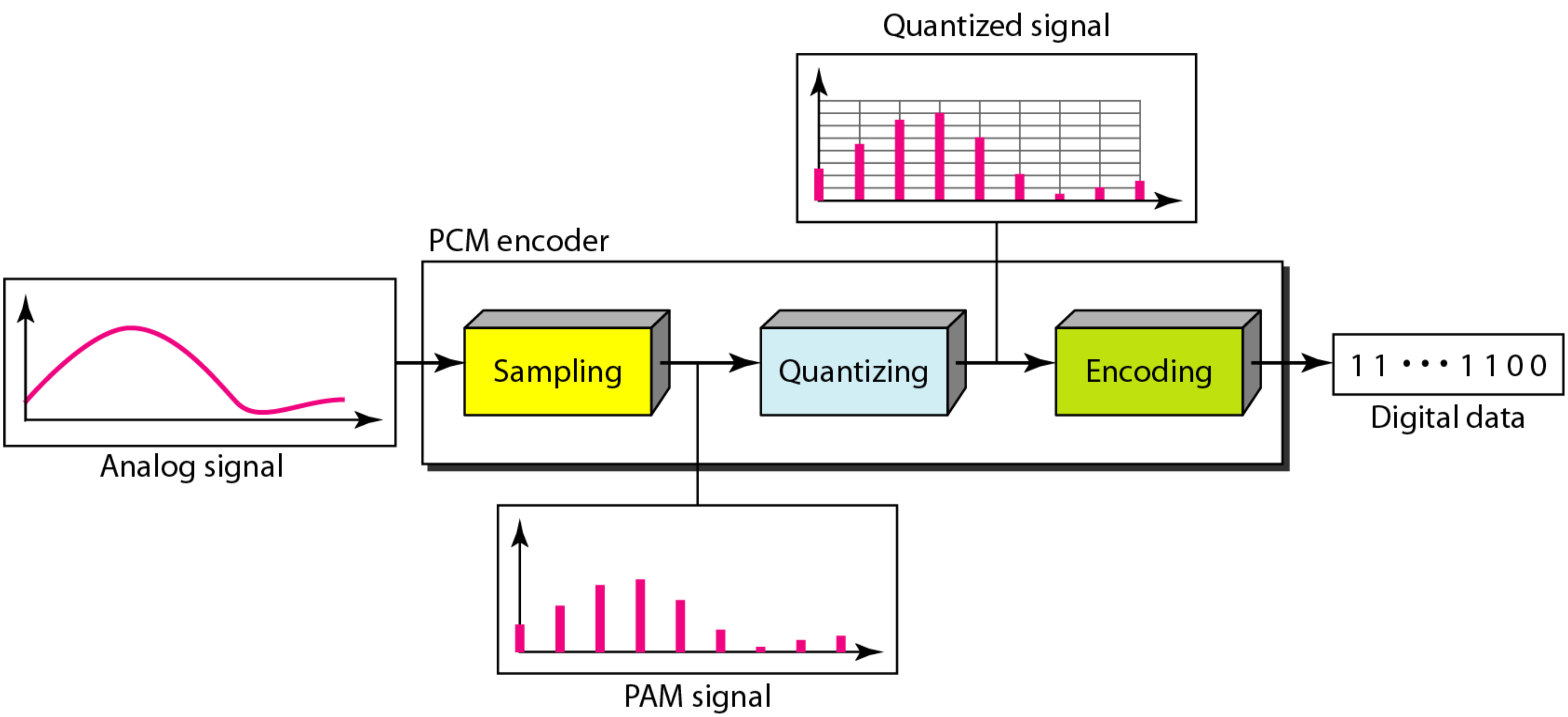
PCM

- PCM consists of three steps to digitize an analog signal:

1. Sampling
2. Quantization
3. Binary encoding

- Before we sample, we have to filter the signal to limit the maximum frequency of the signal as it affects the sampling rate.
- Filtering should ensure that we do not distort the signal, ie remove high frequency components that affect the signal shape.

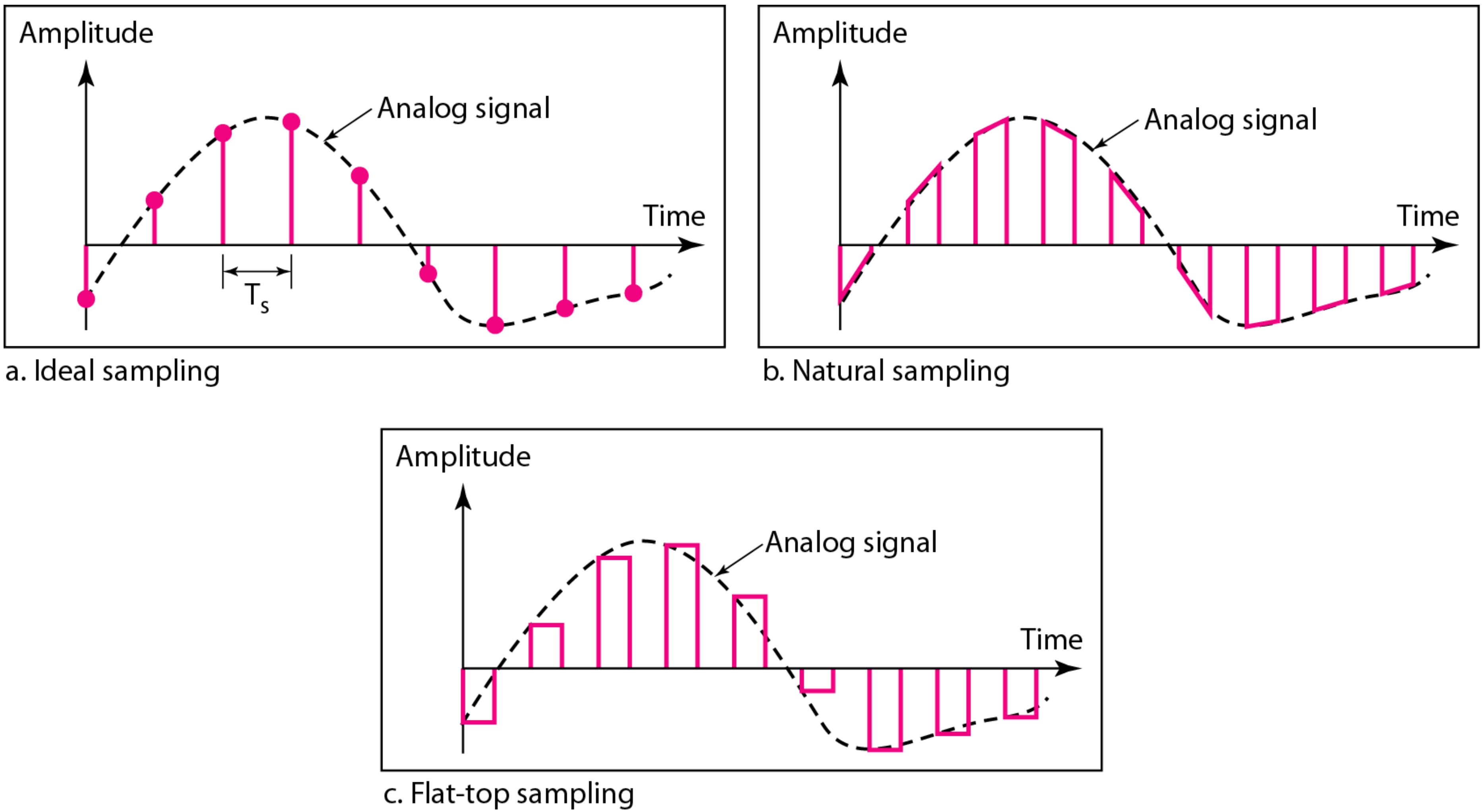
Figure *Components of PCM encoder*



Sampling

- Analog signal is sampled every T_S secs.
- T_s is referred to as the sampling interval.
- $f_s = 1/T_s$ is called the sampling rate or sampling frequency.
- There are 3 sampling methods:
  - Ideal - an impulse at each sampling instant
  - Natural - a pulse of short width with varying amplitude
  - Flattop - sample and hold, like natural but with single amplitude value
- The process is referred to as pulse amplitude modulation PAM and the outcome is a signal with analog (non integer) values

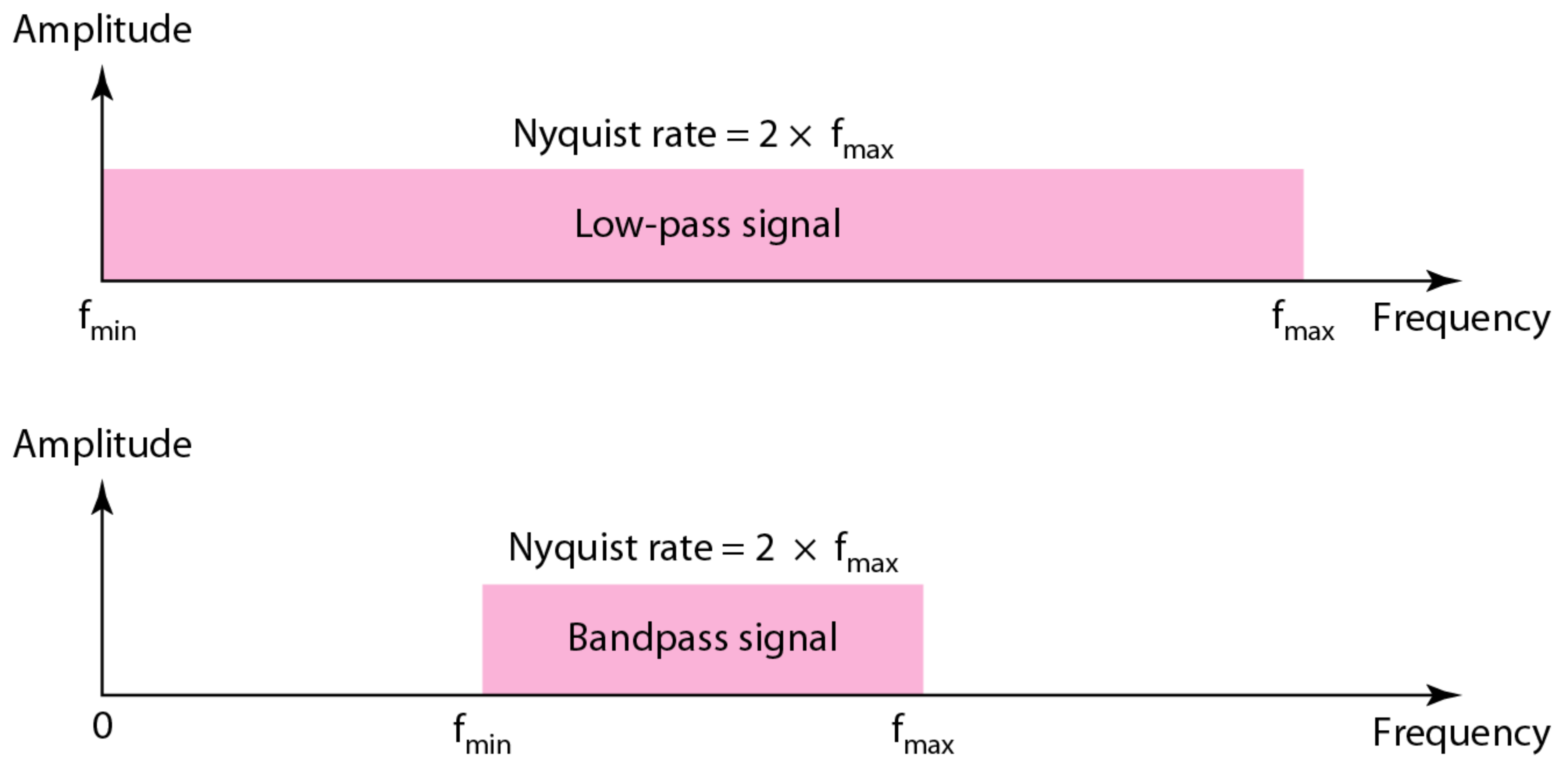
Figure *Three different sampling methods for PCM*



Note

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

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



Quantization

- Sampling results in a series of pulses of varying amplitude values ranging between two limits: a min and a max.
- The amplitude values are infinite between the two limits.
- We need to map the *infinite* amplitude values onto a finite set of known values.
- This is achieved by dividing the distance between min and max into L zones, each of height Δ.

$$\Delta = (max - min)/L$$

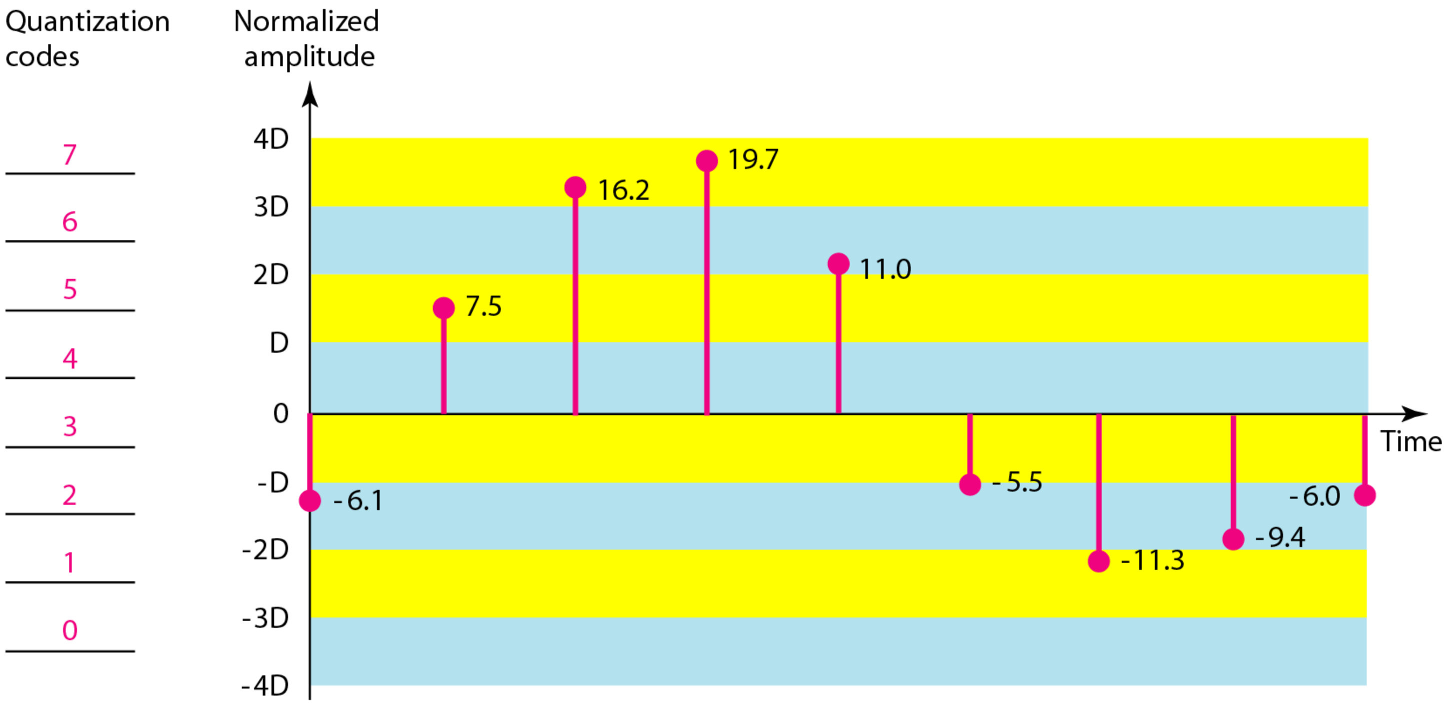
Quantization Levels

- The midpoint of each zone is assigned a value from 0 to L-1 (resulting in L values)
- Each sample falling in a zone is then approximated to the value of the midpoint.

Quantization Zones

- Assume we have a voltage signal with amplitutes V_{min}=-20V and V_{max}=+20V.
- We want to use L=8 quantization levels.
- Zone width Δ = (20 - -20)/8 = 5
- The 8 zones are: -20 to -15, -15 to -10, -10 to -5, -5 to 0, 0 to +5, +5 to +10, +10 to +15, +15 to +20
- The midpoints are: -17.5, -12.5, -7.5, -2.5, 2.5, 7.5, 12.5, 17.5

Figure *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