Unit 2

DIGITAL MODULATION

Basic Concepts

A communication system has mainly 3 entities:

1. **Information(BaseBand)**
2. **Medium**
3. **Carrier**

Basic Concepts

Transmitting information

To transmit a signal over the air, there are three main steps:

1. A pure carrier is generated at the transmitter.
2. The carrier is modulated with the information to be transmitted. Any reliably detectable change in signal characteristics can carry information.
3. At the receiver the signal modifications or changes are detected and demodulated.

Basic Concepts

- **Modulation:**

  The process which places the signal information on to sinewave carriers.
- Definition

  The process by which some characteristics of the carrier,ie(amplitude/frequency/phase) is varied in accordance with the instantaneous value of the modulating signal.
- Modulation may be
- 1. Analog Modulation
- 2. Digital Modulation

Basic Concepts

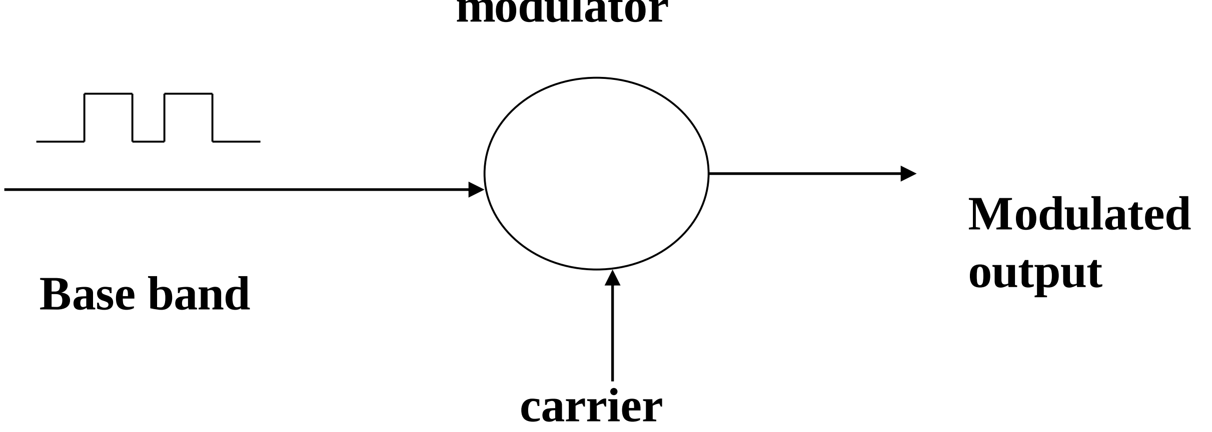
- Digital Modulation techniques translates the **digital baseband data into a form suitable for transmission through the medium.**

Basic Concepts

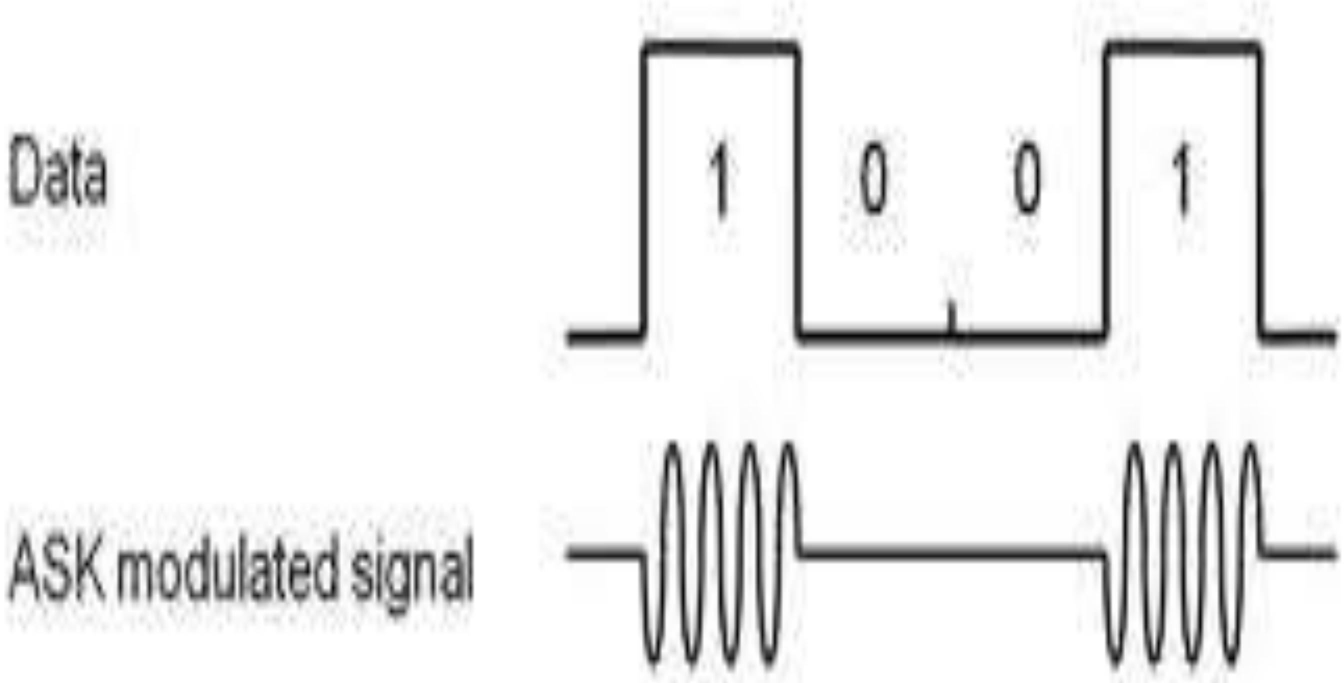
- The basic form of 3 different digital modulation methods used for transmitting digital signals methods are:
- Amplitude Shift Keying
- Frequency Shift Keying
- Phase Shift Keying

Digital Modulation methods

- Amplitude Shift Keying (ASK)
- The amplitude of the carrier is varied in accordance with the amplitude of the modulating signal.



ASK



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BASK

- In amplitude Shift keying, **logic levels are represented by different amplitudes of signals.**

- A binary amplitude-shift keying (BASK) signal can be defined by

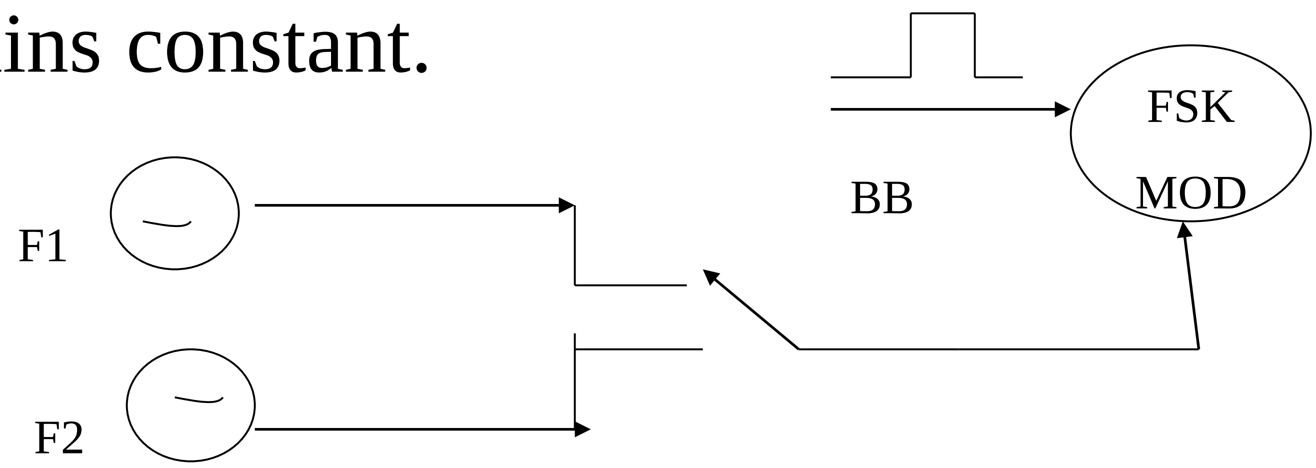
$$s(t) = A\, m(t) \cos 2pi *fc*t,\ 0 < t < T$$

where A is a constant, m(t) = 1 or 0, fc is the carrier frequency, and T is the bit duration.

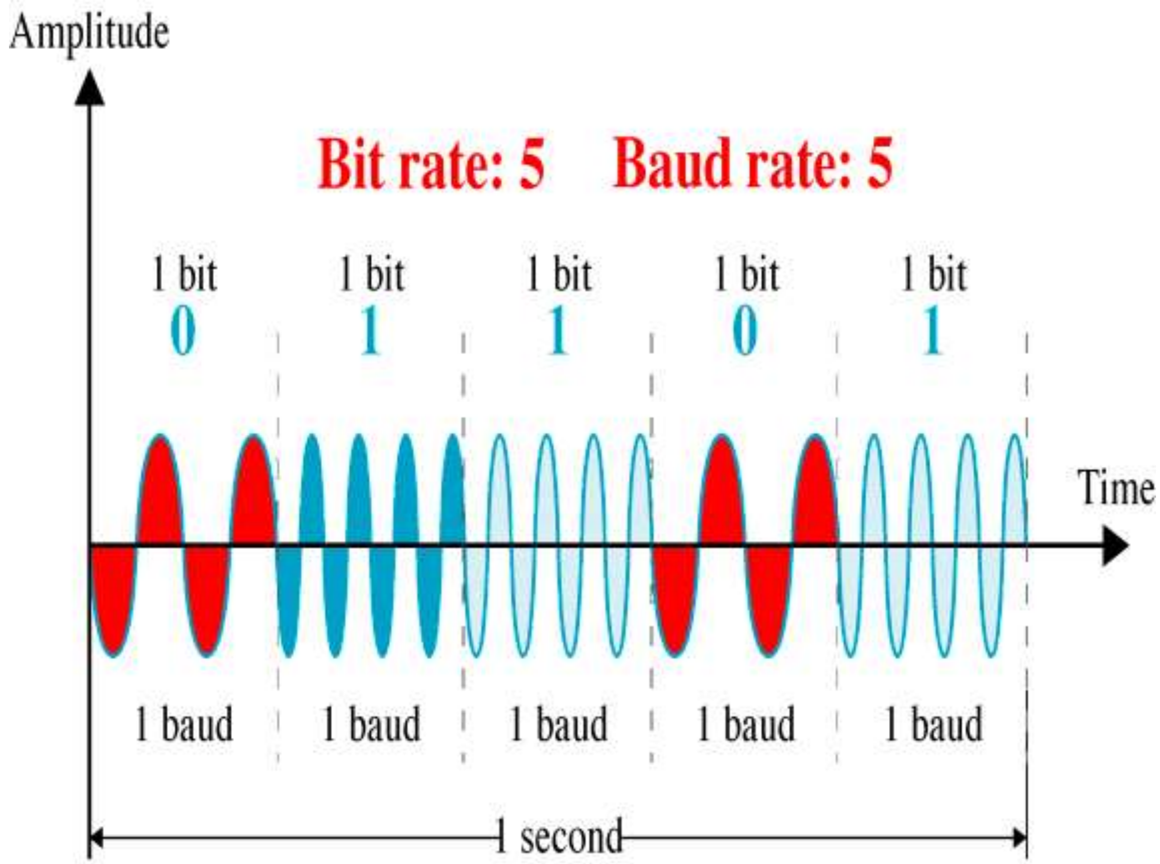
- For B Hz of bandwidth of the baseband signal, BW of BASK signal is 2B Hz

Digital Modulation methods

- Frequency Shift Keying
- Frequency of the carrier is varied in accordance with the amplitude of the modulating signal and the carrier amplitude remains constant.



FSK



Amplitude
Bit rate: 5
Baud rate: 5
1 bit
0
1 bit
1
1 bit
1
1 bit
0
1 bit
1
Time
1 baud
1 baud
1 baud
1 baud
1 baud
1 second

BFSK

- In BFSK, **binary 1 is represented by one frequency (called mark frequency) and binary 0 is represented by another frequency (called space frequency).**

- A binary frequency-shift keying (BFSK) signal is represented by

$$s(t) = E_c \cos 2pi * f_0 * t, \quad 0 \leq t \leq T$$
$$= E_c \cos 2pi * f_1 * t, \quad \text{Elsewhere}$$

where E_c is peak amplitude of signa*l*, f_0 *and* f_1 are the transmitted frequencies and T is the bit duration.

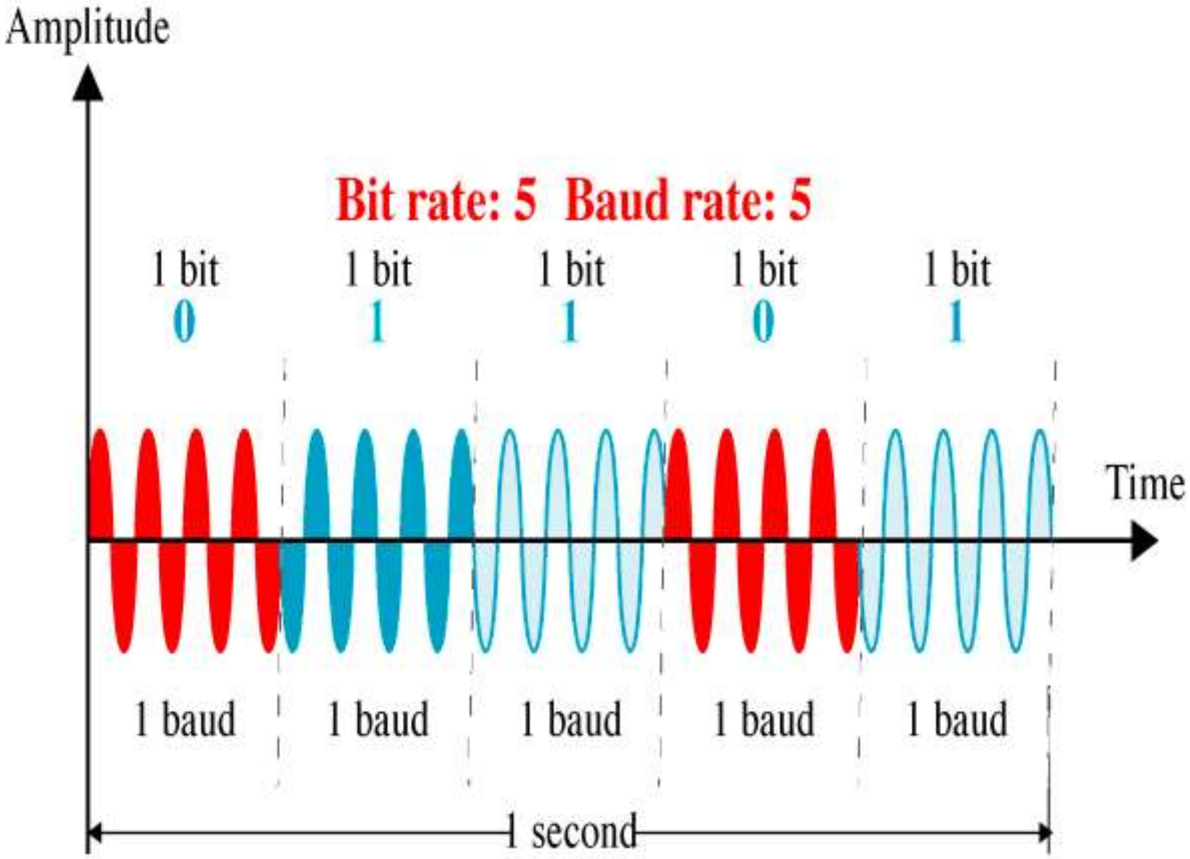
- Minimum BW required for FSK is $2(\Delta f + f_b)$.

Digital Modulation methods

PHASE SHIFT KEYING (PSK)

- The phase of the carrier is varied in accordance with the information.
- PSK is divided into two level and multilevel systems (M-ary schemes).

PSK



Amplitude
Bit rate: 5 Baud rate: 5
1 bit
0
1 bit
1
1 bit
1
1 bit
0
1 bit
1
Time
1 baud
1 baud
1 baud
1 baud
1 baud
1 second

BPSK

- A Binary Phase-Shift Keying (BPSK) signal is represented by

$$s(t) = A\, m(t) \cos 2pi * f_c * t, \quad 0 \le t \le T$$

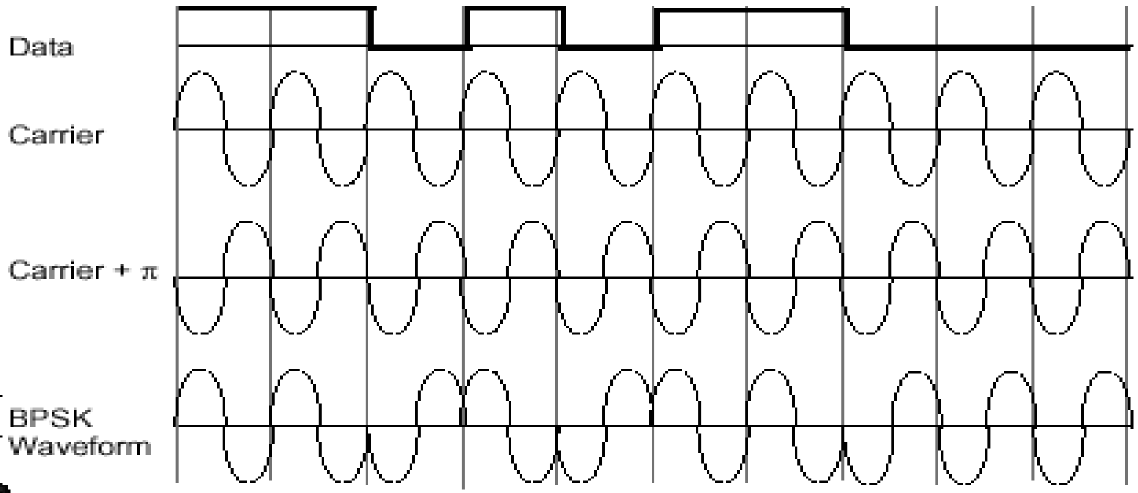
where A is constant, f_c is the carrier frequency, $m(t) = +1$ or -1 and T is the bit duration.

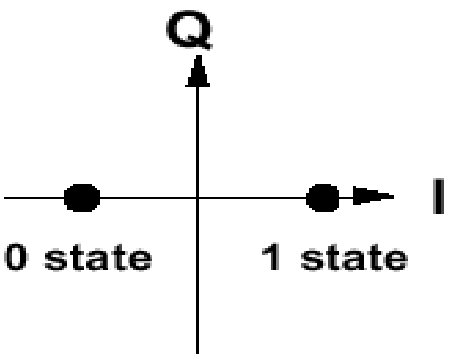
- Minimum BW required for BPSK is f_b

Digital Modulation methods

Phase Shift Keying

- **Binary Phase Shift Keying (BPSK)**
  - Use alternative sine wave phase to encode bits
  - Simple to implement, inefficient use of bandwidth
  - Very robust, used extensively in satellite communications



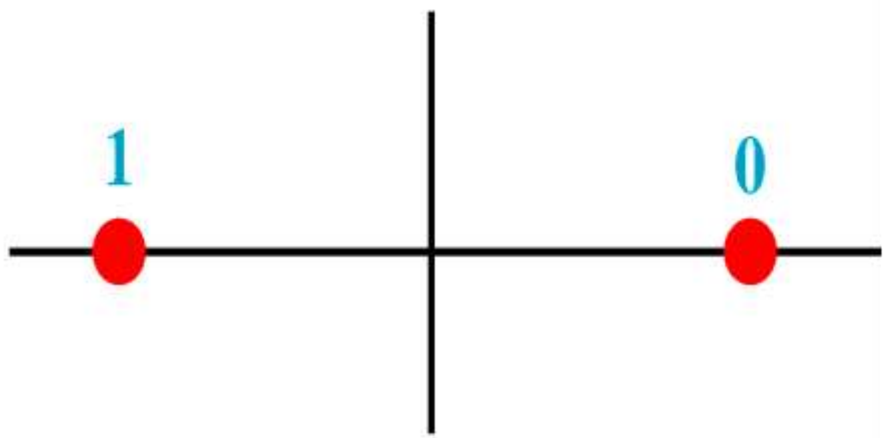


Phases separated by 180˚
(π radians)

PSK

Bit	Phase
0	0
1	180

Bits



Constellation diagram

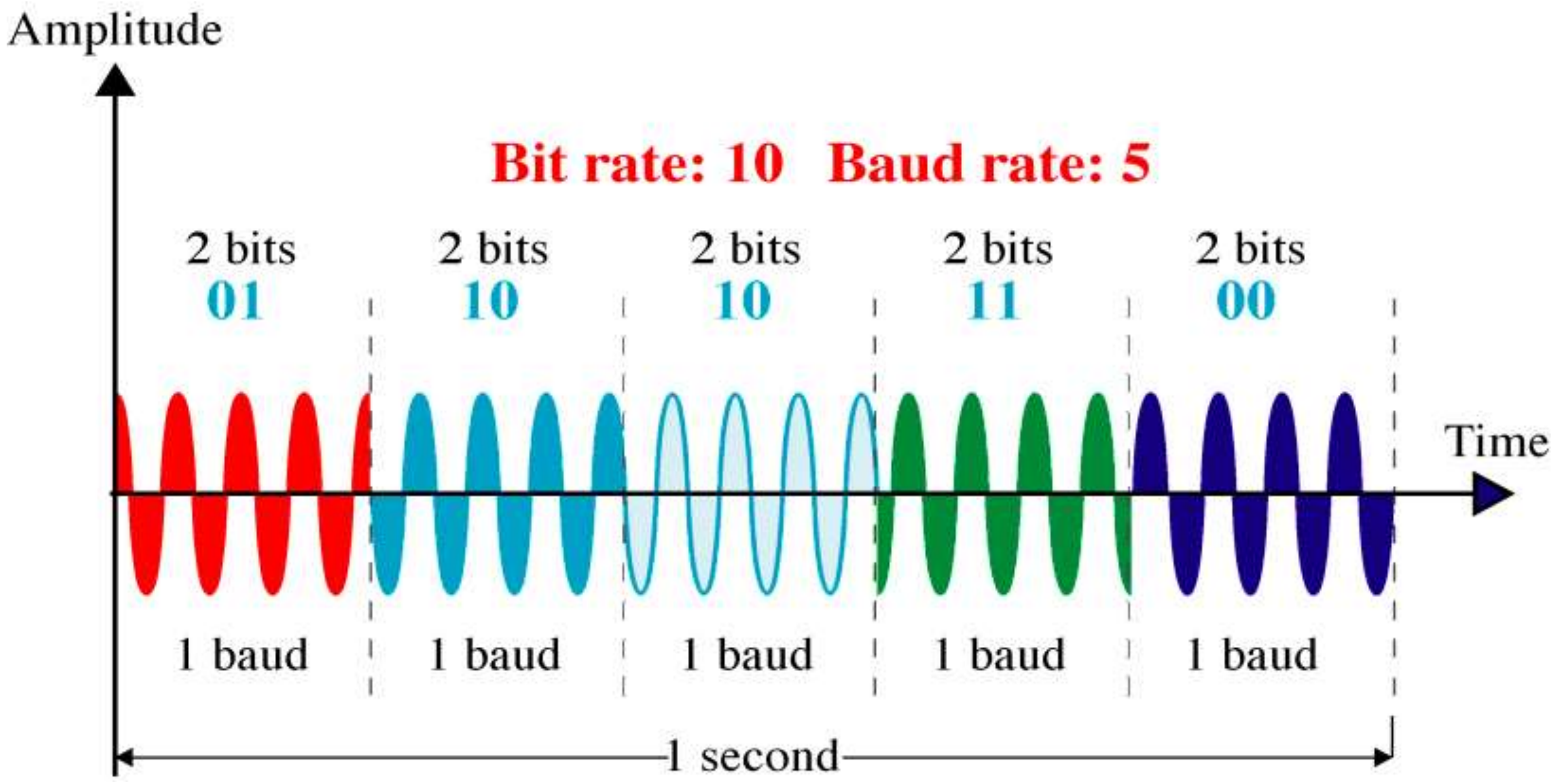
Digital Modulation methods

QPSK

- The modulated output signal is shifted by four phases in accordance with the input binary data.
- QPSK requires two input bits for each phase shift.ie each symbol carries 2 bits. Therefore

  symbol rate in QPSK=Bitrate/2
- In QPSK, four different phasors are represented by 45deg, 135deg, -45deg & +135 deg.
- When the input dibit changes from 00 to 11 or 01 to 10 output phase changes by 180 degrees

QPSK

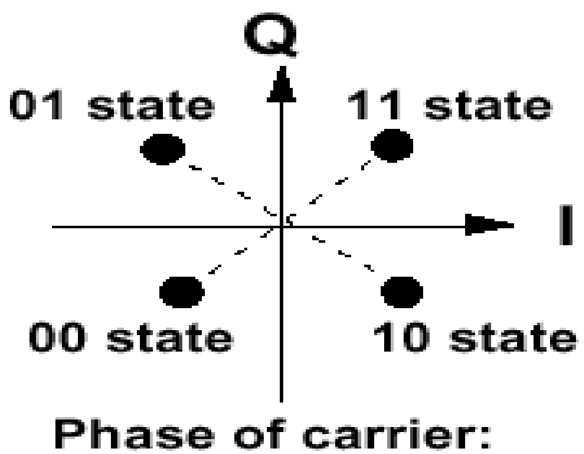


Amplitude
Bit rate: 10
Baud rate: 5
2 bits
01
2 bits
10
2 bits
10
2 bits
11
2 bits
00
Time
1 baud
1 baud
1 baud
1 baud
1 baud
1 second

Digital Modulation methods

Phase Shift Keying

- **Quadrature Phase Shift Keying (QPSK)**
  - Multilevel modulation technique: 2 bits per symbol
  - More spectrally efficient, more complex receiver



+Π/4   +3Π/4   -Π/4   -3Π/4

2x bandwidth efficiency of BPSK

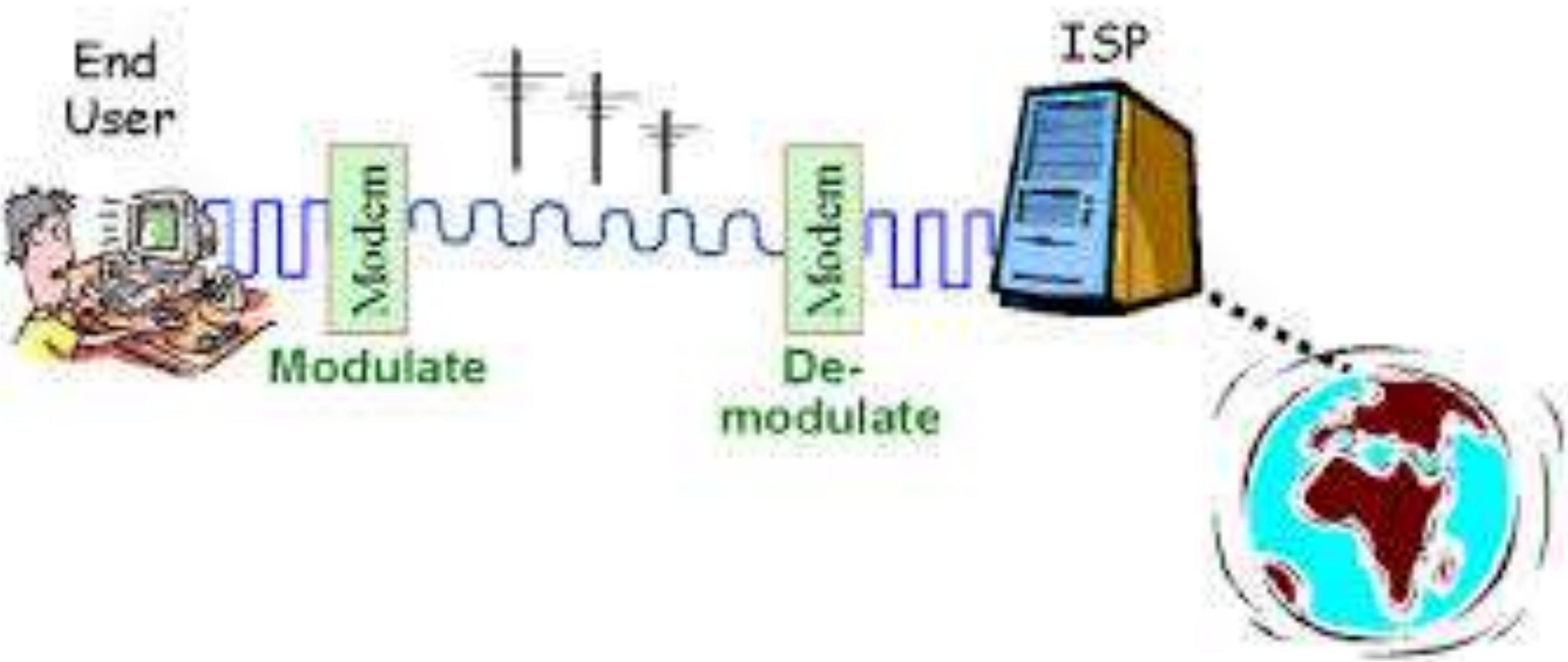
Binary Input		QPSK o/p phase
I	Q	
0	0	-135°
0	1	-45°
1	0	+135°
1	1	+45°

Modem

- **Modem is abbreviation for Modulator – Demodulator.** Modems are used for data transfer from one computer network to another computer network through telephone lines.
- **Modulator** converts information from **digital mode to analog mode** at the transmitting end and demodulator converts the same from **analog to digital at receiving end.**

Modem modulation methods:

FSK,PSK, QAM



Modem

- Modes of modem operation

1. Simplex
2. Half Duplex
3. Full Duplex

- Modem transmission speed

  Low speed – Up to 600 bps

  Medium speed – 600 to 2400 bps

  High speed – 2400 to about 10800 bps

PULSE MODULATION

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 amplif**ied. For this reason, the tendency today is to change an analog signal to digital data.

Advantages of Digital Over Analog For Communications

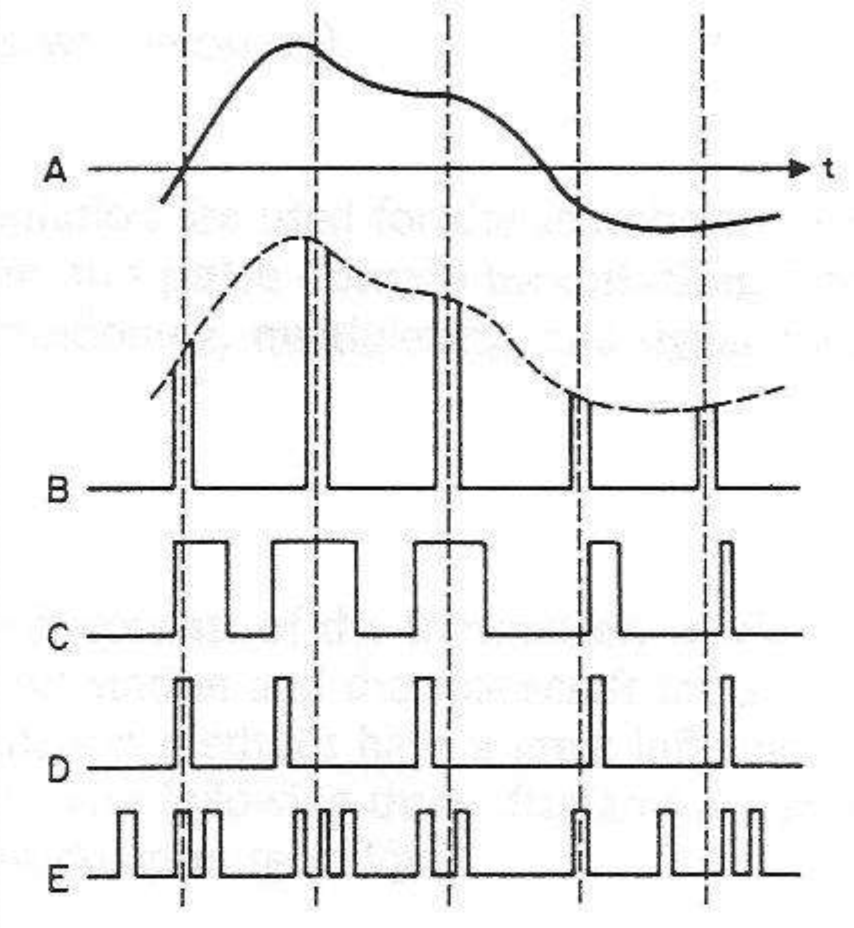
1. **Digital is more robust than analog to noise and interference.**
2. **Digital is more viable to using regenerative repeaters.**
3. **Digital hardware more flexible by using microprocessors and VLSI .**
4. **Can be coded to yield extremely low error rates with error correction.**
5. **Easier to multiplex several digital signals than analog signals .**
6. **Digital is more efficient in trading off SNR for bandwidth .**
7. **Digital signals are easily encrypted for security purposes.**
8. **Digital signal storage is easier, cheaper and more efficient.**
9. **Reproduction of digital data is more reliable without deterioration .**
10. **Cost is coming down in digital systems faster than in analog systems and DSP algorithms are growing in power and flexibility .**

What is Pulse Modulation Technique?

- **The continuous time signals are sampled & sampled values are used to change the periodic pulse train.**

- E.g. In pulse-amplitude modulation (PAM) the amplitude of a train of pulses is varied in proportion to the modulating (message) signal. The pulses are usually spaced at equal time interval.

Pulse Modulation Techniques



Analog signal

Pulse Amplitude Modulation

Pulse Width Modulation

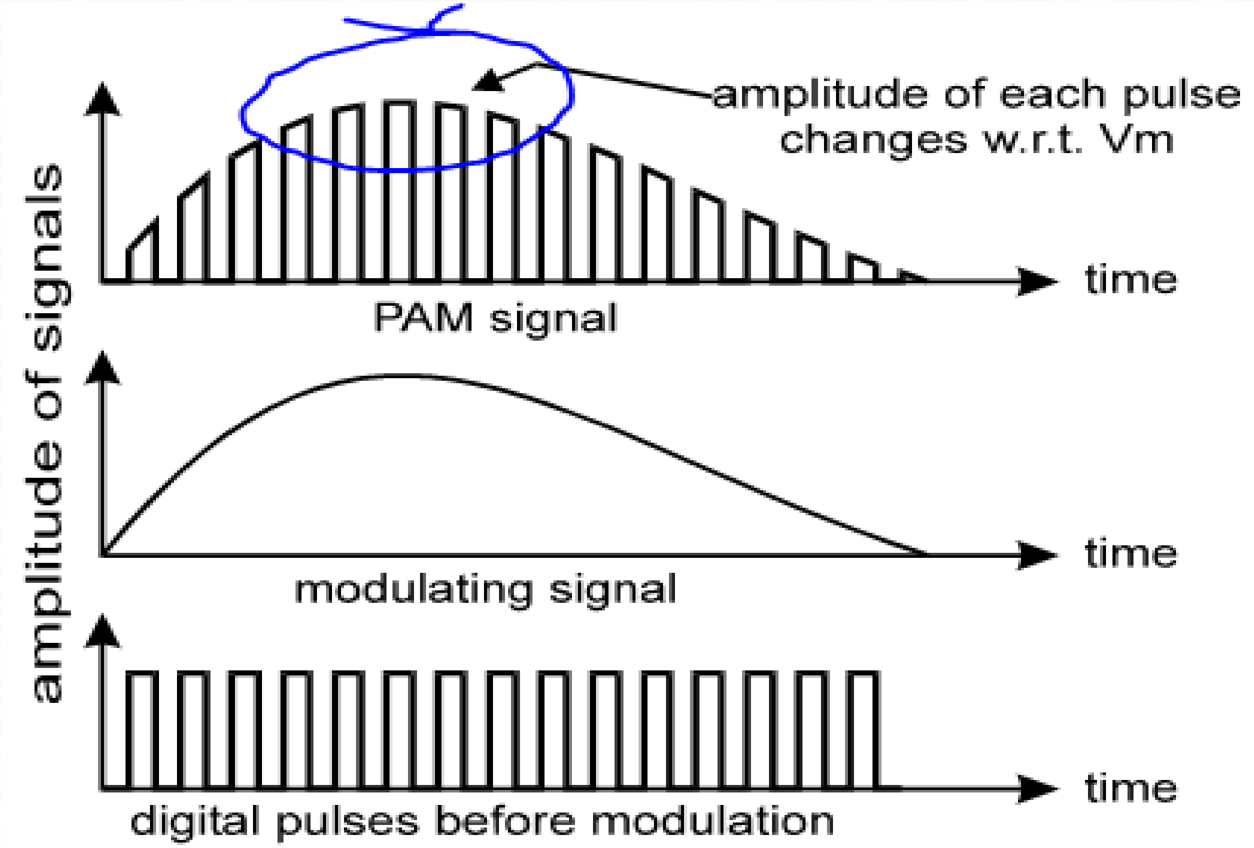
Pulse Position Modulation

Pulse Code Modulation

PAM

- Pulse Amplitude Modulation is achieved by multiplying the carrier with the modulating signal.
- Periodic time of the pulse train T_s is known as the sampling period.
- Sampling frequency $f_s = 1/ T_s$

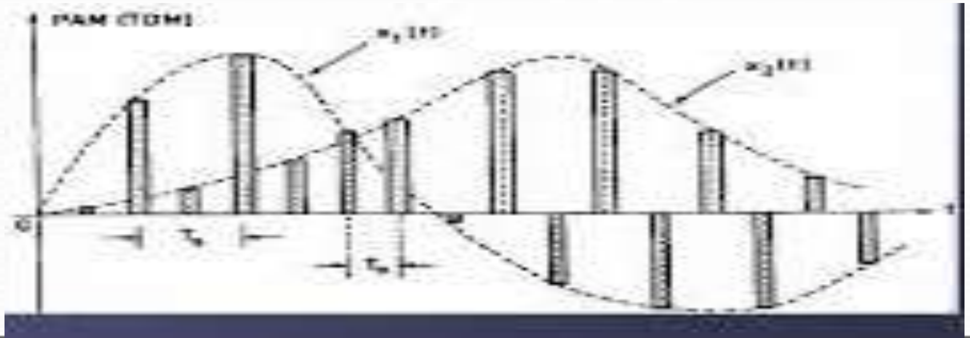
PAM


amplitude of each pulse changes w.r.t. Vm
amplitude of signals
time
PAM signal
time
modulating signal
time
digital pulses before modulation

Advantages of Pulse Modulation

- **It permits the simultaneous transmission of several signals on time-sharing basis. (Time Division Multiplexing).**

- Since pulse modulated signals occupies only part of channel time, several pulse modulated signals can be transmitted on same channel by interweaving.
- **Cab be done by reducing the pulse widths.**

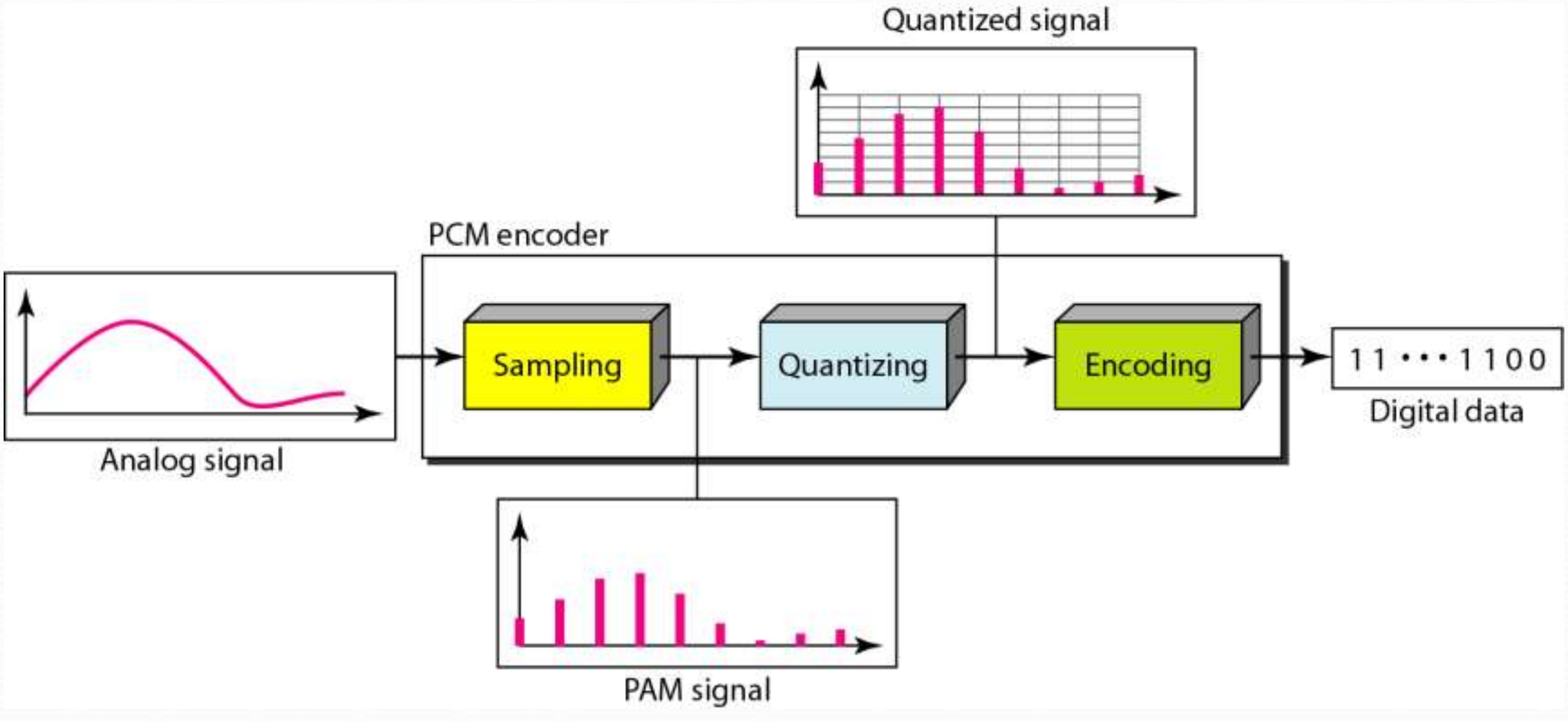


Pulse Code Modulation

- 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.

PCM Encoder Block Diagram



PCM- Sampling Process (1)

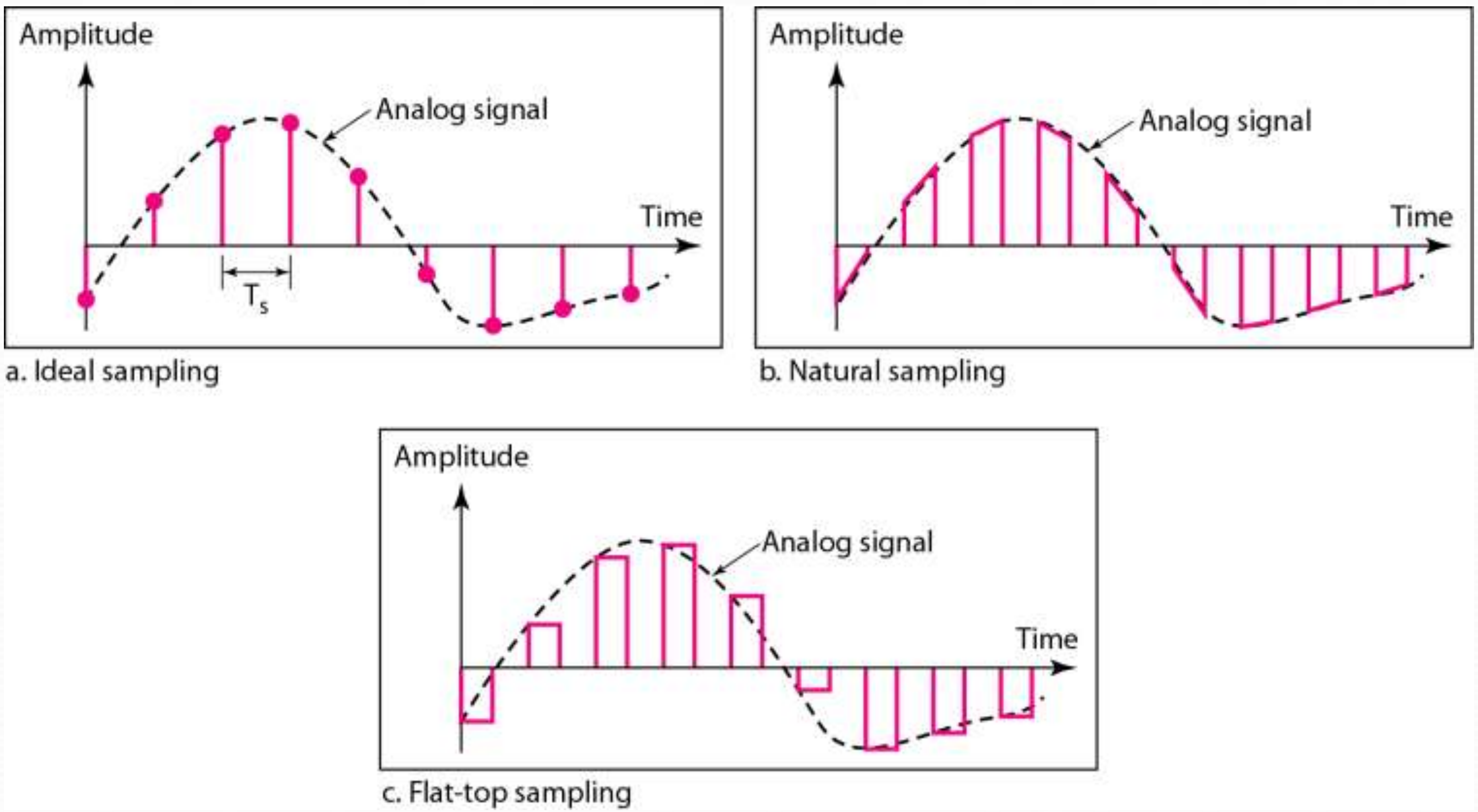
- Analog signal is sampled every T_S secs.
- T_s is referred to as the sampling interval.
- $f_s = 1/T_s$ is called the sampling rate or sampling frequency.

- There are 3 sampling methods:
  - Ideal - an impulse at each sampling instant
  - Natural - a pulse of short width with varying amplitude
  - Flattop - a pulse of short width 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.

PCM-Sampling Methods

Three different sampling methods for PCM

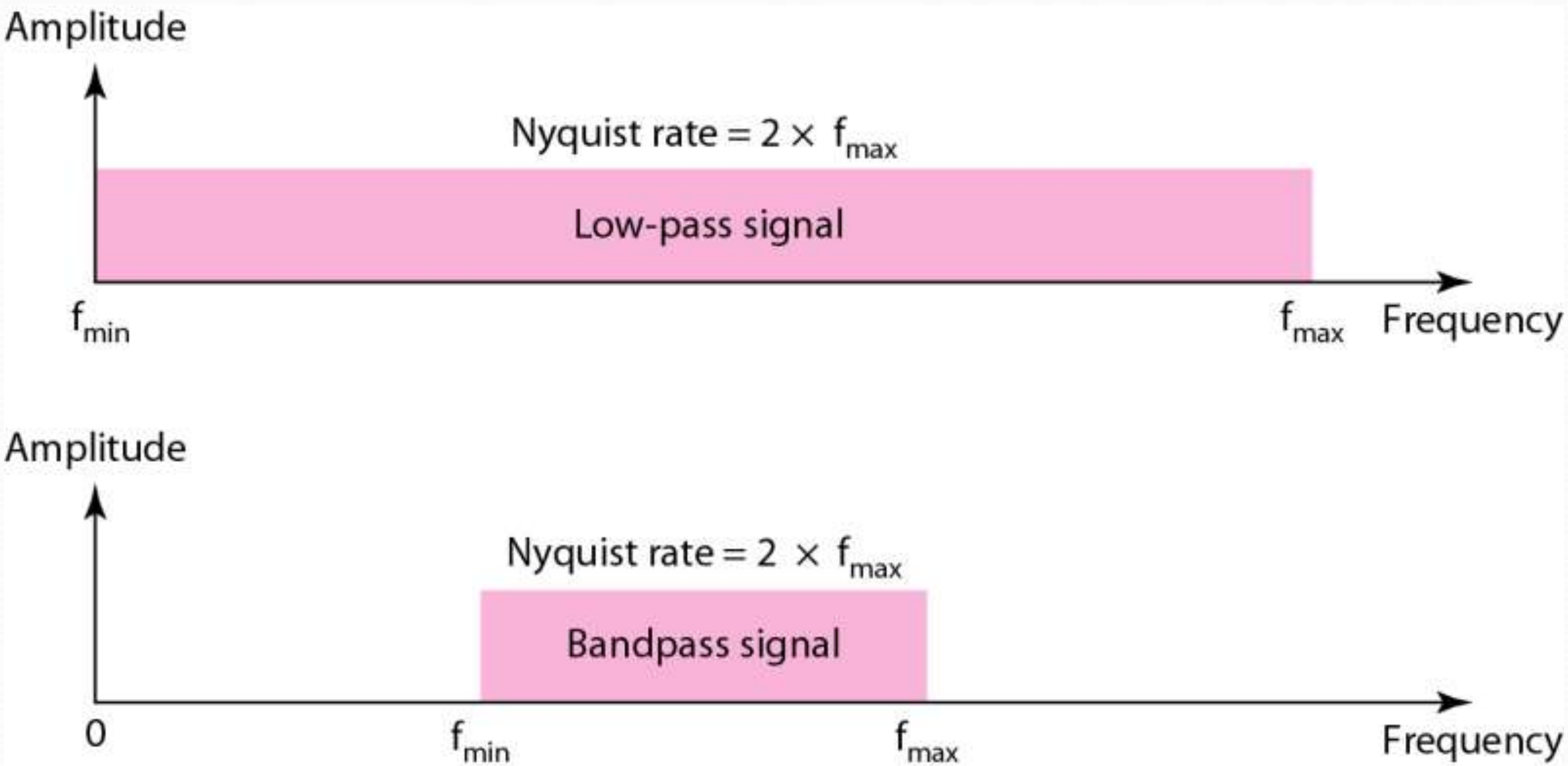


PCM

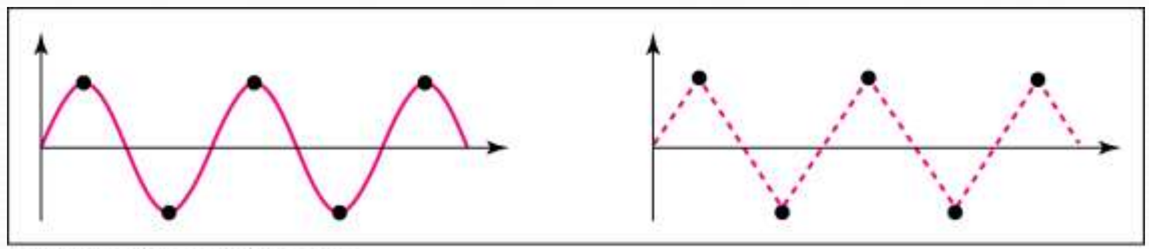
- **According to the Nyquist theorem, the sampling rate must be at least 2 times the highest frequency contained in the signal**

PCM

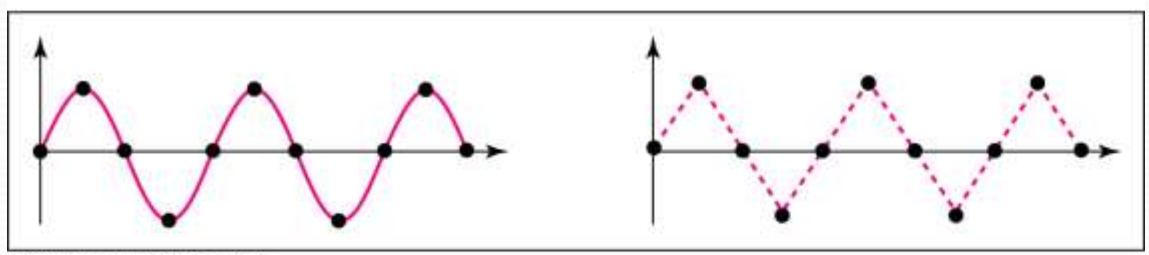
Nyquist sampling rate for low-pass and bandpass signals



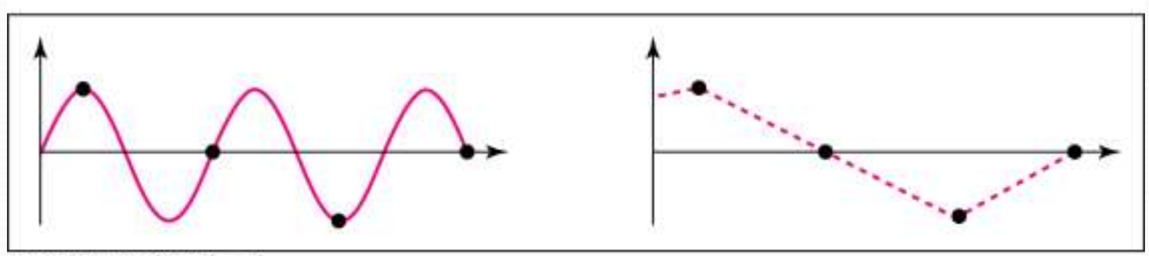
PCM-Effect of Sampling Rate



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



b. Oversampling: $f_s = 4f$



c. Undersampling: $f_s = f$

PCM

Example

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

PCM- Quantization Process (2)

- 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.
- **Quantization is the process 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.

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

PCM

- 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.**

- Assume we have a voltage signal with amplitudes V_{min}=-20V and V_{max}=+20V
- We want to use L=8 quantization levels
- Zone width $\Delta = (20 - -20)/8 = 5$
- 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

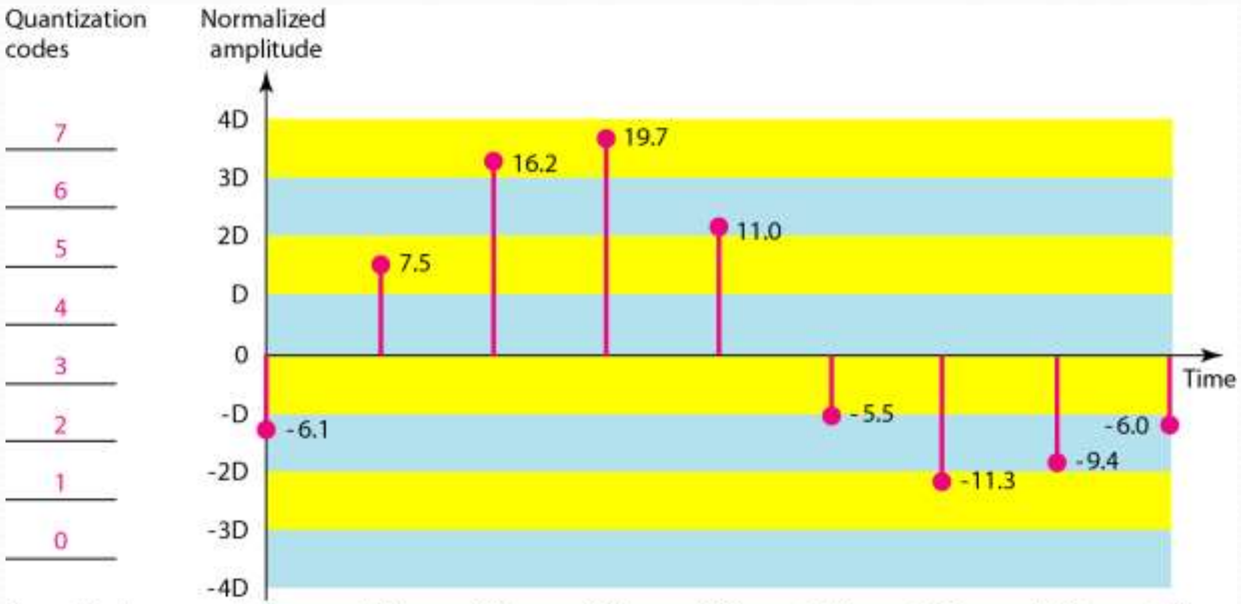
PCM- Encoding (3)

- Each zone is then assigned a binary code.
- The number of bits required to encode the zones, or the number of bits per sample is obtained as follows:

$$n_b = \log_2 L$$

- Given our example, $n_b = 3$
- The 8 zone (or level) codes are therefore: 000, 001, 010, 011, 100, 101, 110, and 111
- Assigning codes to zones:
  - 000 will refer to zone -20 to -15
  - 001 to zone -15 to -10, etc.

PCM



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 Error

- When a signal is quantized-the coded signal is an approximation of the actual amplitude value.
- The difference between actual and coded value (midpoint) is referred to as the quantization error.
- The more zones, the smaller Δ which results in smaller errors.
- BUT, the more zones the more bits required to encode the samples -> higher bit rate.

Bit rate and bandwidth requirements of PCM

- The bit rate of a PCM signal can be calculated form the number of bits per sample x the sampling rate.

$$\textbf{Bit rate} = n_b \times f_s$$

- The bandwidth required to transmit this signal depends on the type of line encoding used.

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

- A digitized signal will always need more bandwidth than the original analog signal. Price we pay for robustness and other features of digital transmission.

Problem

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

Solution

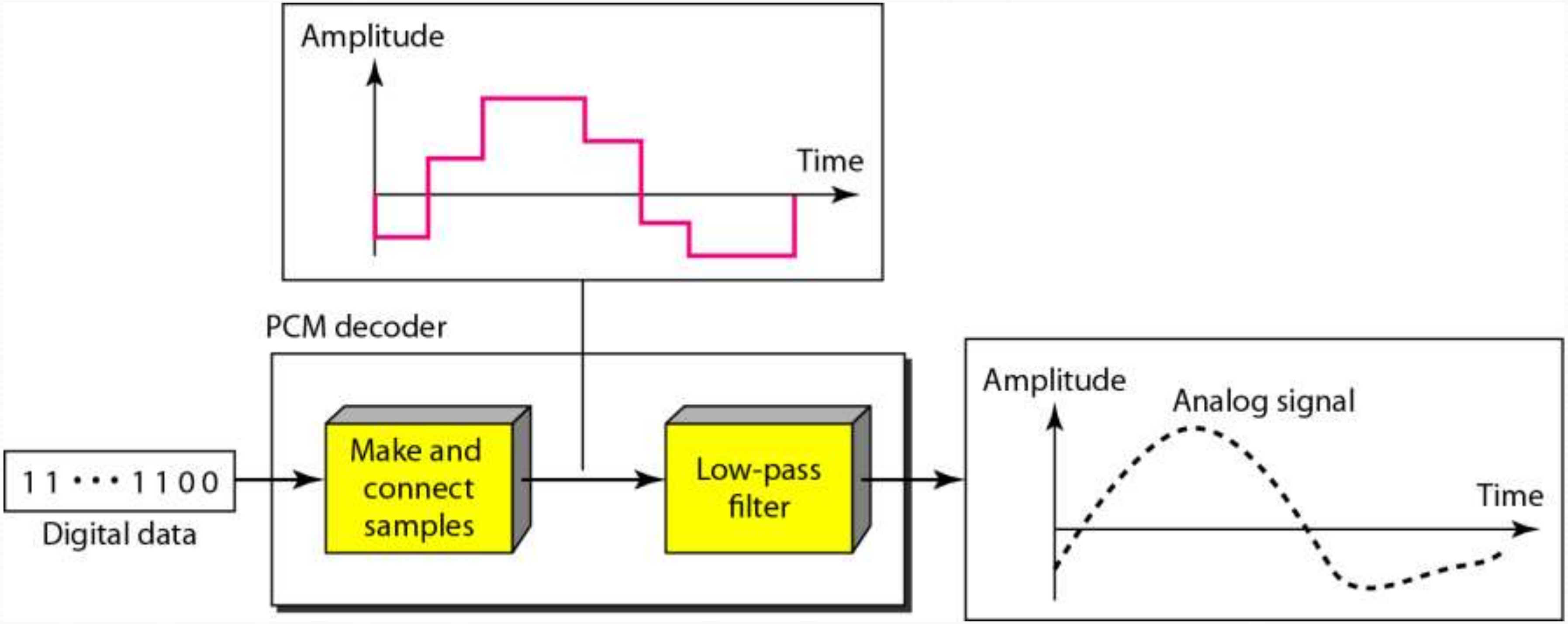
The human voice normally contains frequencies from 0 to 4000 Hz. So the sampling rate and bit rate are calculated as follows:

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

PCM decoder

- To recover an analog signal from a digitized signal we follow the following steps:
  1. We use a **hold circuit** that holds the amplitude value of a pulse till the next pulse arrives.
  2. We pass this signal through a **low pass filter** with a cutoff frequency that is equal to the highest frequency in the pre-sampled signal.
- The higher the value of L, the less distorted signal can be recovered.

Block Diagram-PCM decoder



ADVANTAGES OF PCM

- **Low Noise Susceptibility:** PCM signals can be transmitted farther than analog signals without signal degradation, information loss, and distortion.
- **Repeatability:** The signal is completely regenerated by each repeater, making it noise-free at the start of each repeated transmission.
- **Encoded Signal:** A PCM signal can be modulated in such a way that only a specific decoder can make sense of the underlying data. This is useful when the data being sent requires a level of security.
- **Storage:** Compact disk is the classic application of PCM.

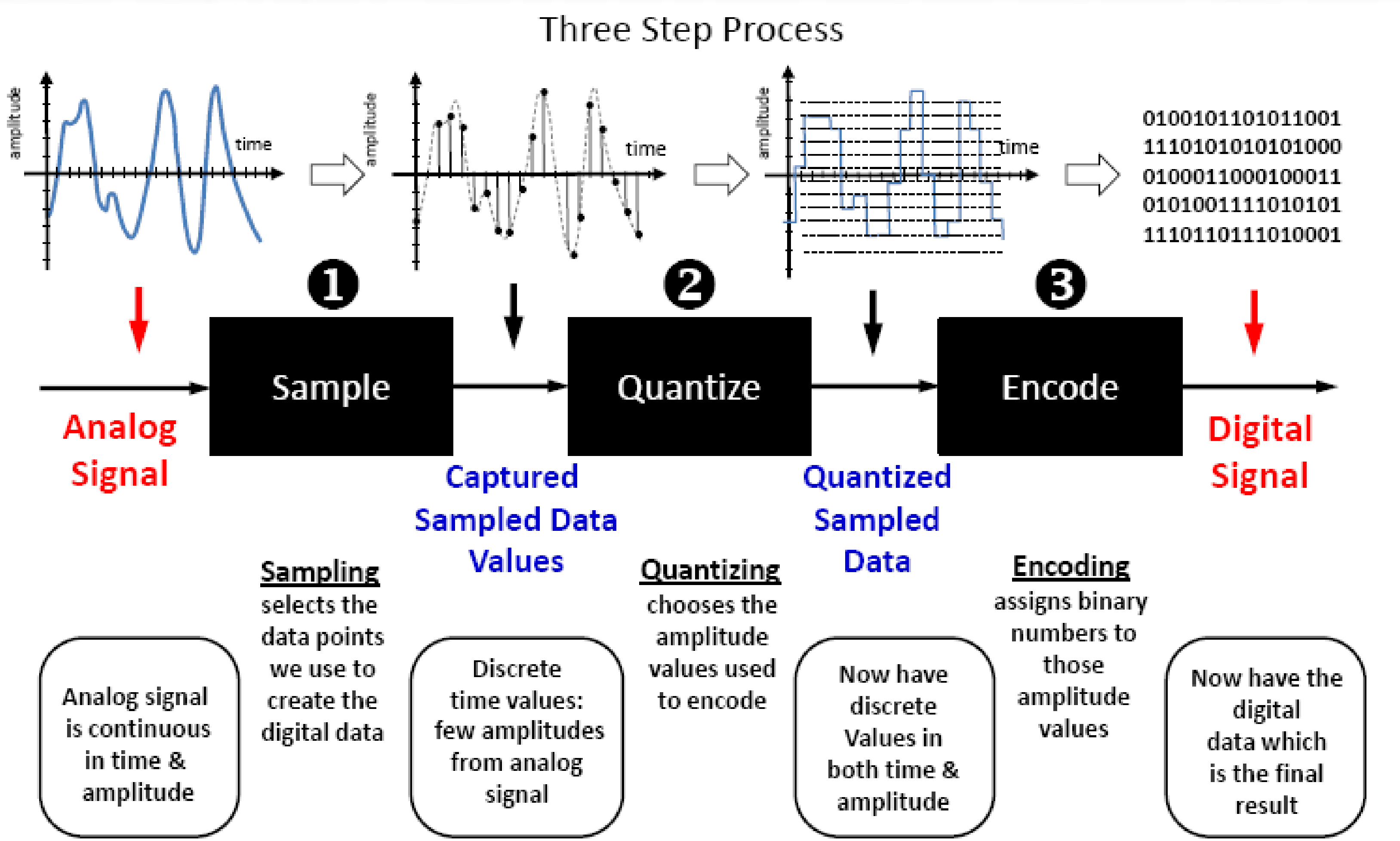
APPLICATIONS OF PCM

- PCM was originally intended for use in telephone systems.

- But in the 21st century, it is also the standard way for digitalizing analog data such as in digital audio in computers, digital video and CD formats, telemetry, digital telephony and other digital audio applications.

- Also used in space communication .

Analog to Digital Conversion Process (ADC)

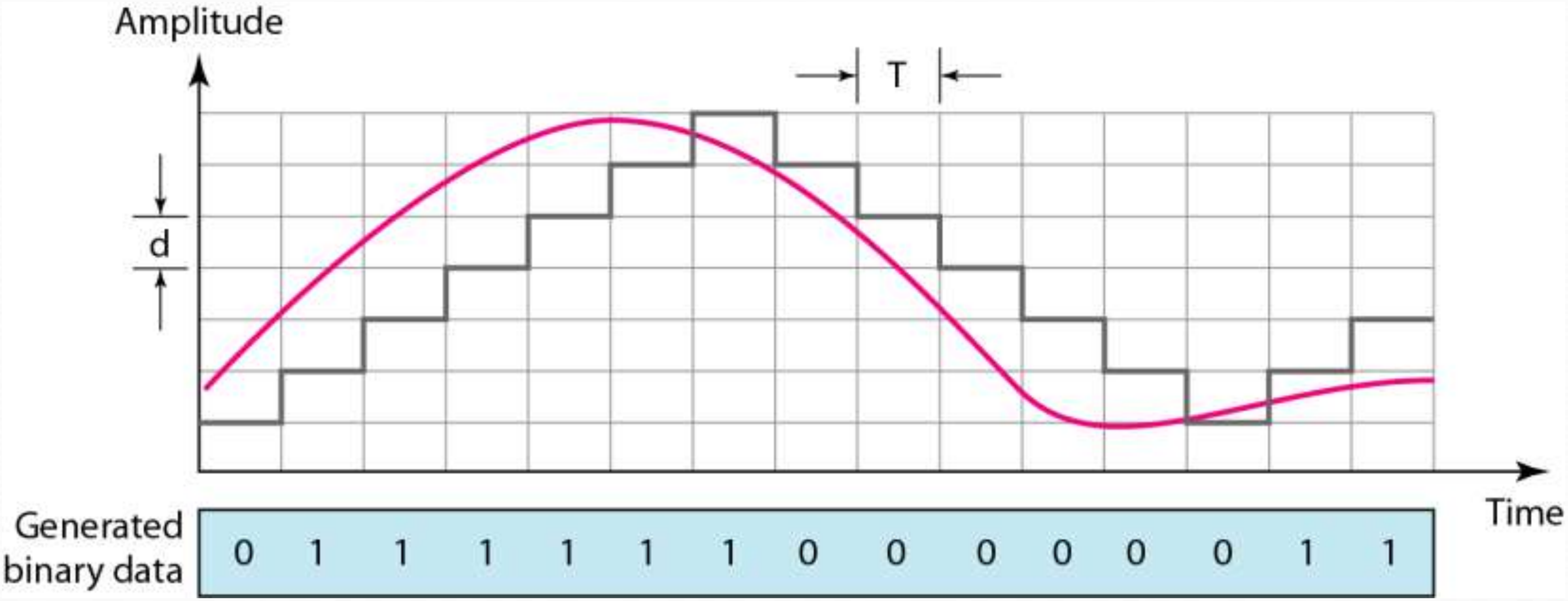


Note: “Discrete time” corresponds to the timing of the sampling.

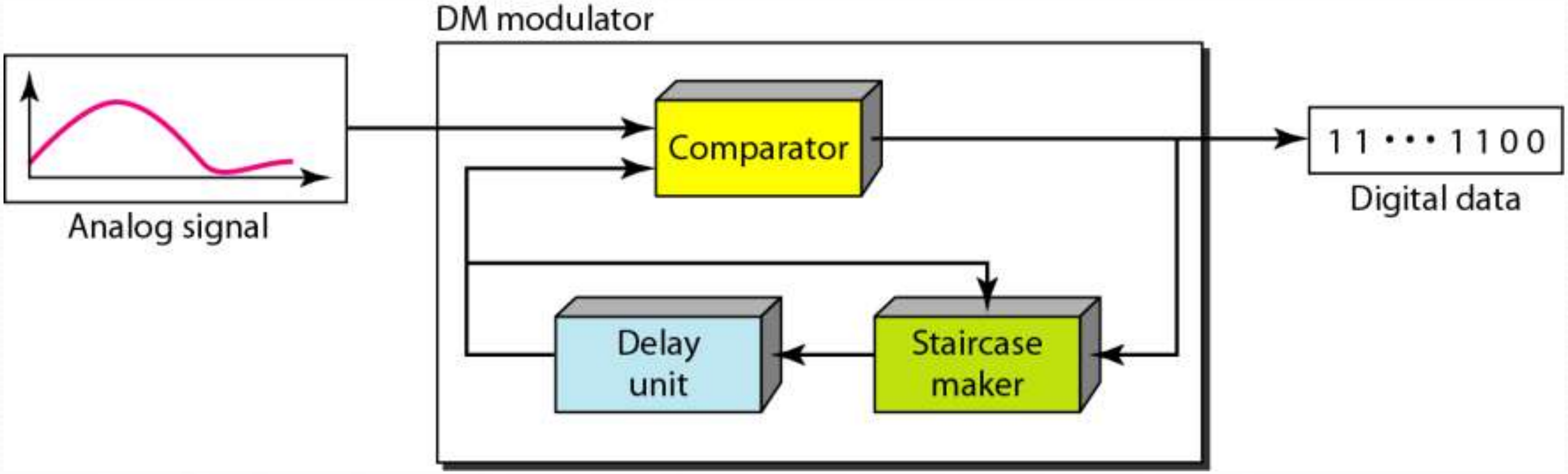
Delta Modulation

- This scheme sends only the difference between pulses, if the pulse at time t_{n+1} is higher in amplitude value than the pulse at time t_n, then a single bit, say a "1", is used to indicate the positive value.
- If the pulse is lower in value, resulting in a negative value, a "0" is used.
- This scheme works well for small changes in signal values between samples.
- If changes in amplitude are large, this will result in large errors.

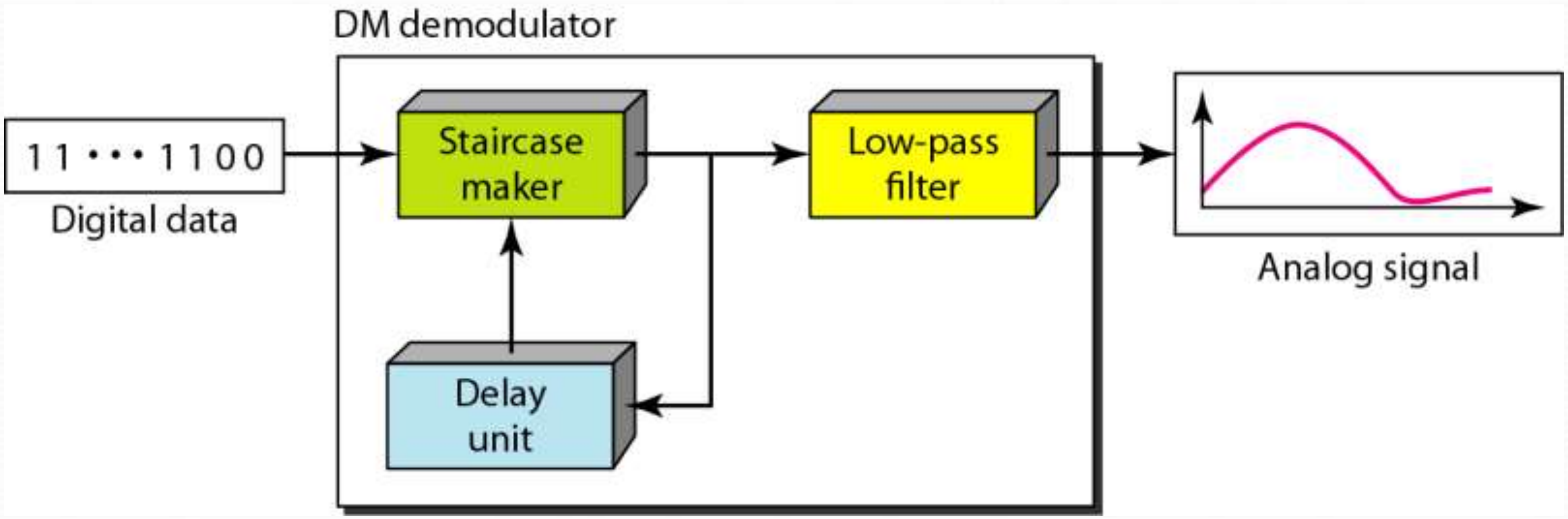
The process of delta modulation



Delta modulation components



Delta demodulation components



Delta PCM (DPCM)

- Instead of using one bit to indicate positive and negative differences, we can use more bits -> quantization of the difference.

- Each bit code is used to represent the value of the difference.

- The more bits the more levels -> the higher the accuracy.