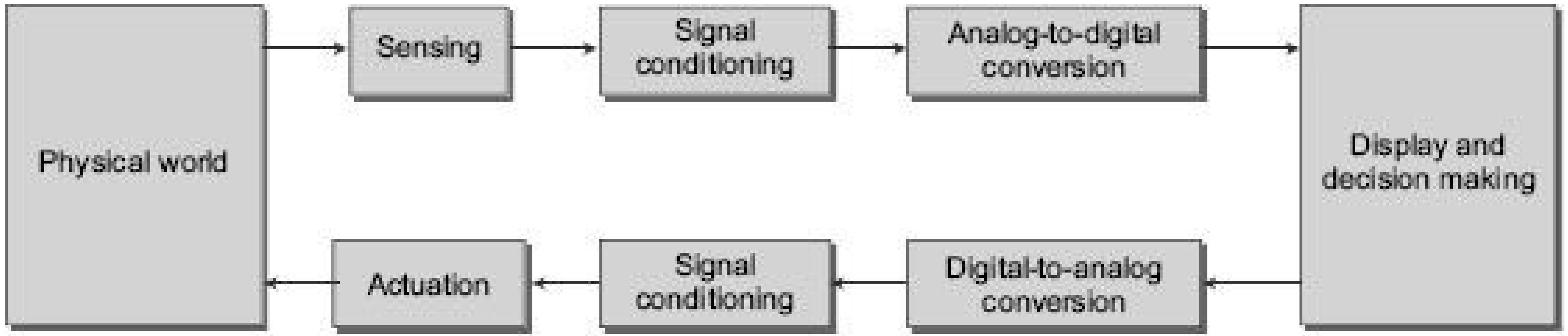
Data Acquisition Systems (DAQ)

Lecture Two

Related Tasks	Assessment Criteria	Assessment Methods
a) Use appropriate sensor to sense signal b) Perform signal sampling c) Perform signal quantization d) Perform data coding	Data acquisition methods are correctly described	• Assignments • Hands-on exercises • Written Exam

Basic Components of Data Acquisition Systems

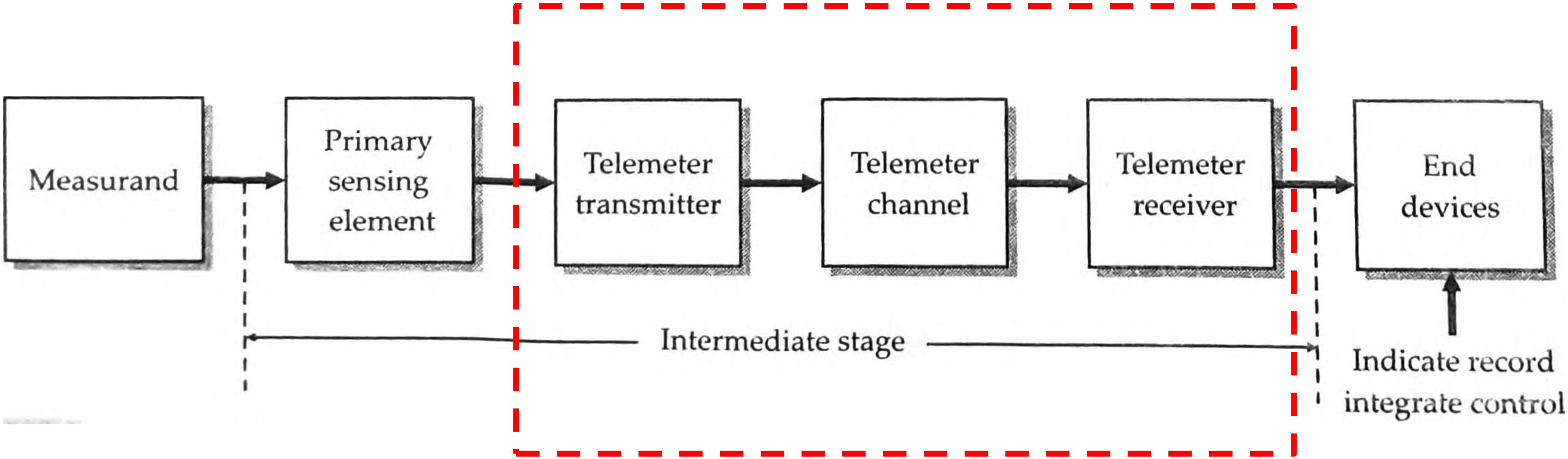
- The basic elements of a data acquisition system with exception of the channel, as shown in the functional diagram (Figure next slide) are as follows:
    1) Sensors and transducers
    2) Field wiring
    3) Signal conditioning
    4) Data acquisition hardware
    5) PC (operating system)
    6) Data acquisition software


Physical world
Sensing
Signal conditioning
Analog-to-digital conversion
Display and decision making
Digital-to-analog conversion
Signal conditioning
Actuation

Introduction

- In modem measurement systems, the various components comprising the system are usually located at a distance from each other.
- It, therefore, becomes necessary to transmit the data or an information between them through some form of communication channels.
- The terms *data transmission* and *telemetry* refer to the process by which information regarding the quantity being measured, may be using a transducer or sensor and signal conditioning equipment, is transferred to a remote location, perhaps to be processed, recorded and displayed.
- Telemetry is the technology which enables a user to collect data from several measurement points at inaccessible or inconvenient locations, transmit that data to a convenient location, and present the several individual measurements in a usable form.

General Telemetering System



Sensors and Transducers

The shift to smarter sensors

- Basically, a sensor is an input device that receives and responds to a signal or stimulus.
- The Latest Sensors are becoming more and more intelligent, providing higher accuracy, flexibility and easy integration into distributed systems.
- Intelligent sensors use standard bus or wireless network interfaces to communicate with one another or with microcontrollers (MCUs).
- The network interface makes data transmission easier while also expanding the system.

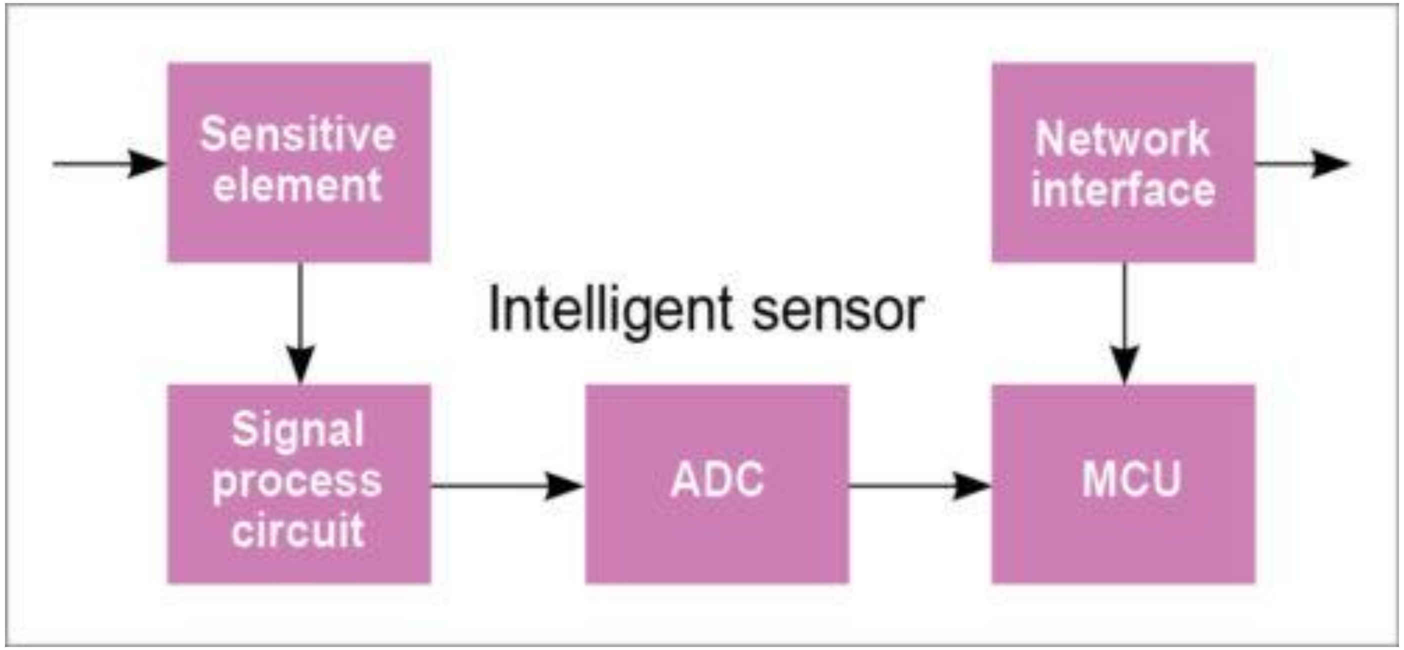


Figure: Intelligent sensor structure

- When choosing the best analog sensor to use, you must match the characteristics of the physical variable you are measuring with the characteristics of the sensor.
- The two most important sensor characteristics are

1) The sensor output
2) The sensor bandwidth

Example of sensors

- Sensors include (but not limited to) temperature sensors, proximity sensors, pressure sensors, RF sensors, pyroelectric IR sensors, water-quality sensors, chemical sensors, smoke sensors, gas sensors, liquid-level sensors, automobile sensors and medical sensors.

Classification of Sensors

- There are several classifications of sensors made by different authors and experts.
- In the first classification of the sensors, they are divided in to Active and Passive.
    - **Active Sensors**: require an external excitation signal or a power signal.
    - **Passive Sensors:** do not require any external power signal and directly generates output response.

- The other type of classification is based on the means of detection used in the sensor:

  Some of the means of detection are Electric, Biological, Chemical, and Radioactive.

- The next classification is based on conversion phenomenon i.e., the input and the output.

  Some of the common conversion phenomena are Photoelectric, Thermoelectric, Electrochemical, and Electromagnetic.

- The final classification of the sensors are Analog and Digital Sensors.

  Analog Sensors produce an analog output i.e., a continuous output signal (usually voltage but sometimes other quantities like Resistance etc.) with respect to the quantity being measured.

  Digital Sensors, in contrast to Analog Sensors, work with discrete or digital data.

- In general, signals acquired via sensors is in analogue form and must be amplified and/or conditioned to make them suitable for subsequent analysis or for integration within active control loops.
- Some level of analog signal conditioning or processing include:
    1) Amplify weak sensor signals,
    2) Filter the noise that is usually mixed in with the original signal, and to eliminate high-frequency components in the signal that would cause aliasing (i.e., appearance of dubious counterparts of the original signal when sampled below a certain sampling rate commonly, referred to as the Nyquist rate).

- Introduction of Amplifiers and Filters are discussed in the next lecture
- Details in this area are presented in the respective modules

- Amplification mainly serves for increasing resolution of the input signal.
- If, for example, a low-level signal of the order of a few mV is fed to a 12-bit ADC, there will be a loss of precision as the resolution of the ADC is of the order of 2 mV.
- However, if the signal is amplified to the order of 10 V (full scale voltage for ADC), we get the maximum precision.
- Amplifying a signal before sending it through a cable to the receiving end enables high SNR to the noises introduced in the path having noise interference.
- This ensures the improved precision of the measurement.
- If, however, the signal is amplified after the noise interference causes low SNR which implies the noise causes a considerable error in the input signal.

Analog Input Subsystem

- Analog signals are continuous in time and in amplitude (within predefined limits).
- Sampling takes a "snapshot" of the signal at discrete times, while quantization divides the voltage (or current) value into discrete amplitudes.
- Sampling frequency has to be at least twice the frequency of the event that requires capture.
- This rule is called the Nyquist Criterion.
- If one fails to follow this rule then a phenomenon called aliasing occurs.
- Aliasing is when a frequency higher than half of our sampling frequency gets "folded" back onto a frequency that is less than half of our sampling frequency.
- This creates a ghost signal that can really mess up our results.

Analogue to Digital Conversion (ADC)

- Analog to digital conversion of a continuous input signal normally occurs in two steps: *sampling* and *quantization*.
- The sampler takes a time-varying analog input signal and converts it to a fixed voltage, current, electrical charge, or other output level.
- The quantizer takes the constant sampled level and compares it to the closest level from a discrete range of values called quantization levels.
- Sampling and Quantization Link

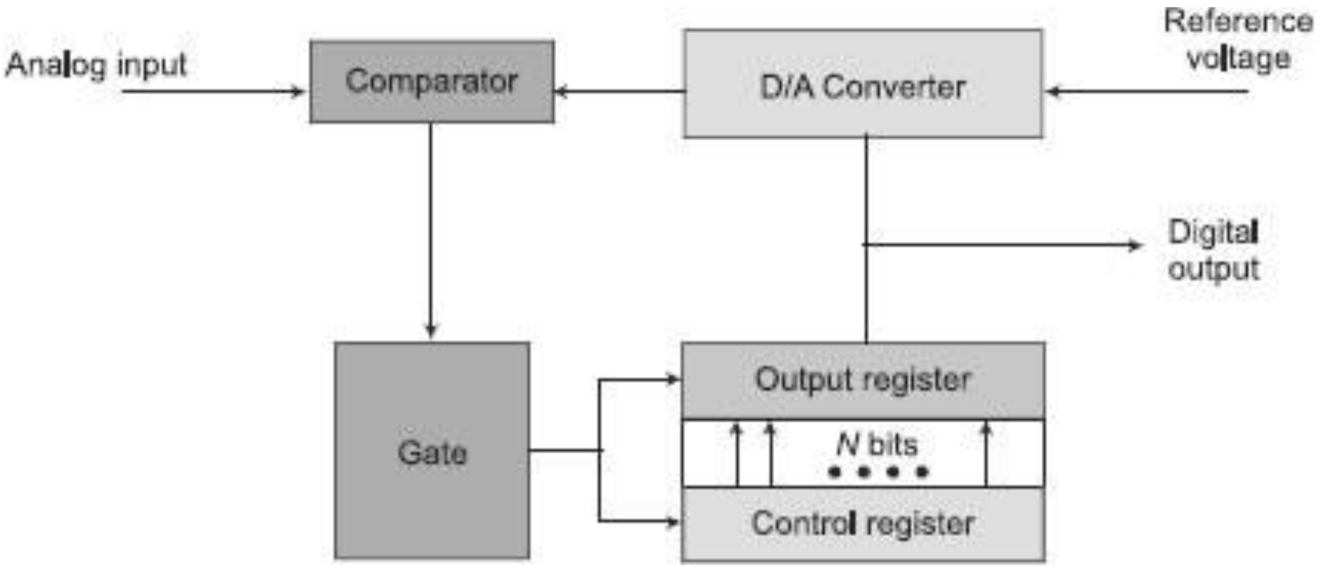
- The performance of analog and digital converters is typically quantified by two primary parameters, speed (in samples per second) and resolution (in bits).
- Higher resolution ADCs typically require a large signal-to-noise ratio and good linearity.
- ADCs with high sampling rates are frequently desired, but generally have lower resolution.
- There are two basic techniques for performing analog-to-digital conversion: an open-loop technique and a feedback technique.

- **Different Types of A/D Converters**
- While all analog-to-digital converters are classified by their resolution or number of bits, how the A/D circuitry achieves this resolution varies from device to device.
- There are four primary types of A/D converters used for industrial and laboratory applications:

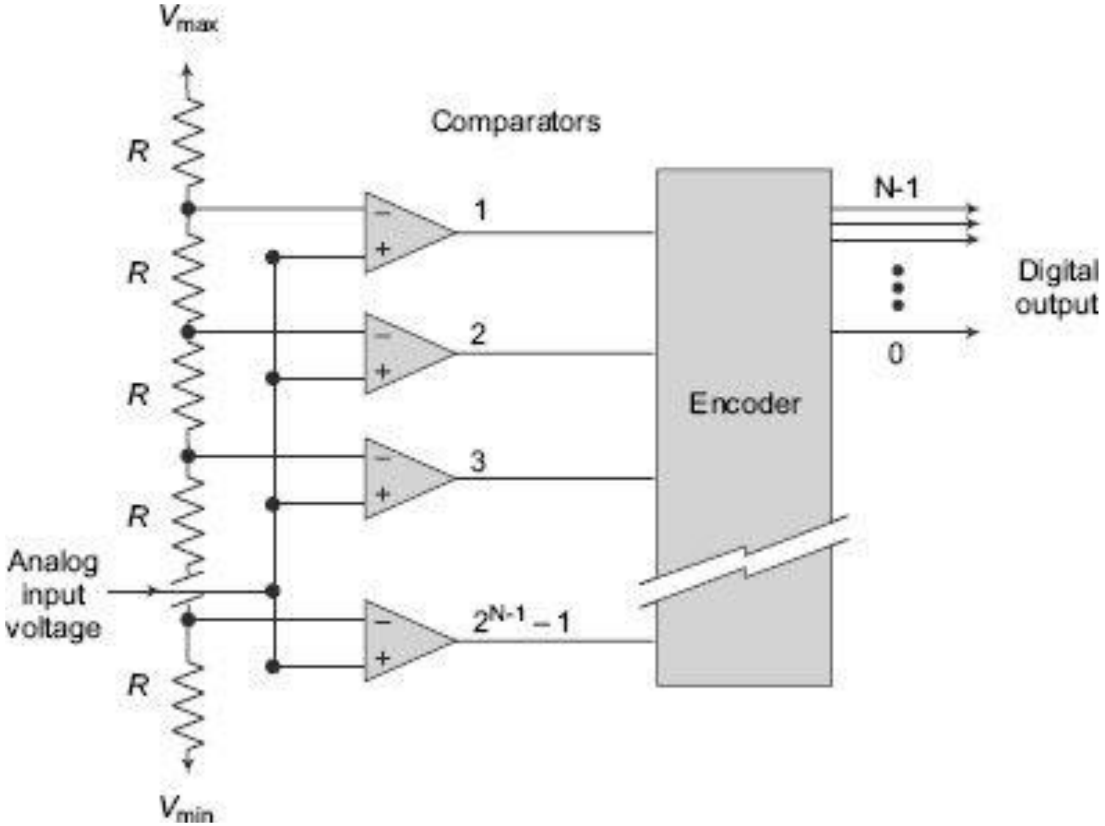
1) Successive approximation
2) Flash/Parallel
3) Integrating
4) Ramp/Counting

- Industrial and lab data acquisition tasks typically require 12 to 16 bits. 12 are the most common.
- As a rule, increasing resolution results in higher costs and slower conversion speed.

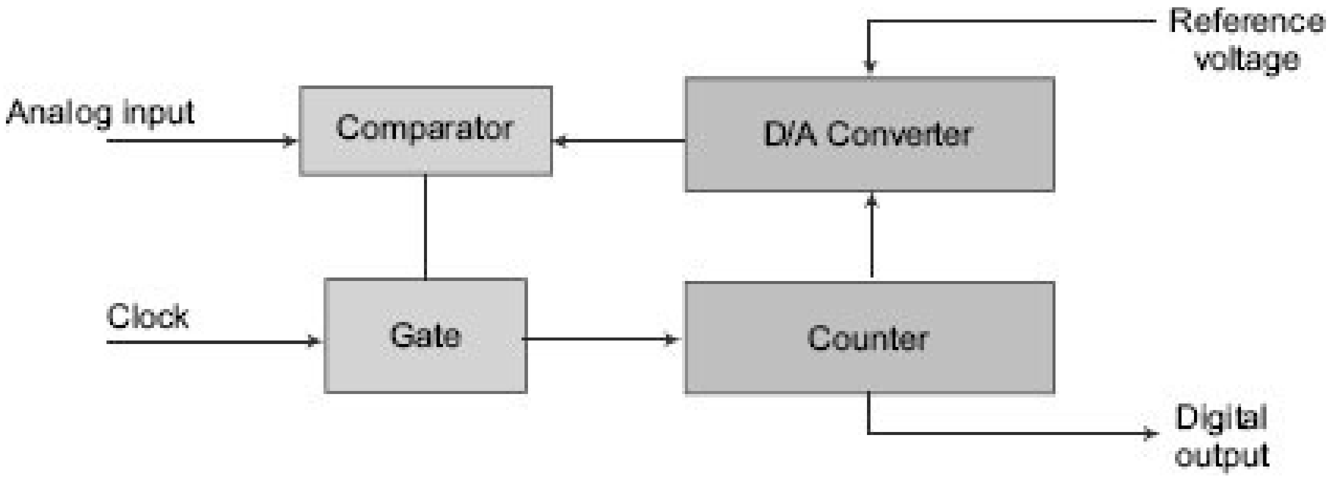
DESIGN	SPEED	RESOLUTION	NOISE IMMUNITY	COST
Successive approximation	Medium	10–16 bits	Poor	Low
Integrating	Slow	12–18 bits	Good	Low
Ramp/Counting	Slow	14–24 bits	Good	Medium
Flash/Parallel	Fast	4–8 bits	None	High



Successive Approximation



Flash/Parallel



Ramp/Counter

Elements of Telemetering System

- There are three system elements in the intermediate stage which are peculiar to a telemetering system, they are :
    i. telemeter transmitter,
    ii. telemeter channel,
    iii. telemeter receiver.

Elements of Telemetering System

- The function of the *telemeter transmitter* is to convert the output of a primary sensing element into an electrical signal and to transmit it over a telemetering channel.
- This signal is in electrical format and is received by a receiver placed at a remote location.
- This signal is converted into a usable form by the *receiver* and is indicated or recorded by an end device, which is graduated in terms of the measurand.
- The end device may be a control element which may be used for the control of the input quantity (measurand), through a feedback loop to produce desired output.

Types of Telemetry Systems

- Two types of telemetering systems are used :
    1) A Land line telemetry,
    2) A R.F. (radio frequency) telemetry.

Land Line Telemetering System

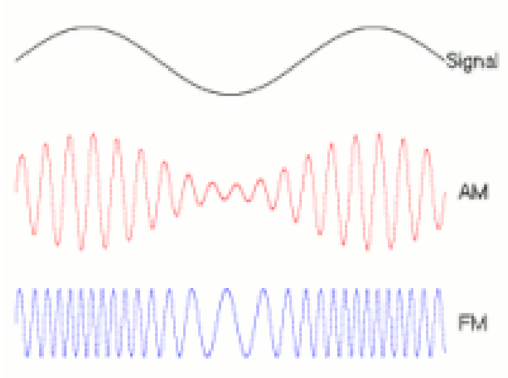
- A land line telemetering system requires a telemeter channel which is a physical link between the telemeter transmitter and receiver.
- The land line telemetering is, in fact, a direct transmission of information through cables and transmission lines.

- The direct transmission via cables employs current, voltage, frequency, position or impulses to convey the information.
- Current, voltage and position type systems can be used only for short distances while for long distance telemetry, pulse and frequency types of systems are used.
- The information may be in the form of analog or digital signals.
- While current, voltage, position, frequency and pulse types of signals can be used for analog telemetry, only pulse signals can be used for digital telemetry.

Radio Frequency (R.F.) Telemetry

- The telemetry, earlier on, has been defined, as a technology that enables the user to collect data from several measurement points at inaccessible or inconvenient locations.
- This is very true of applications which require Radio Frequency (R.F.) telemetry, as in such applications, there is no physical link between the transmitting and receiving stations.
- R.F. telemetry is usually more suitable if the data is to be transmitted over distances greater than 1 km.

Modulation Methods



Analogue modulation methods

- In analog modulation, the modulation is applied continuously in response to the analogue information signal.

Common analog modulation techniques include:

- Amplitude modulation (AM)
- Angle modulation,
  1) Frequency modulation (FM)
  2) Phase modulation (PM)

Analogue Modulation Methods

- The modulation methods used for transmission in R.F. (radio frequency) are also applicable to land line transmission.

The expression for a carrier wave is :

$$e_c = A_c \sin(2\pi f_c t + \theta) \qquad \ldots$$

where A_c = amplitude of carrier,

f_c = frequency of carrier ; Hz,

θ = relative phase shift of carrier, rad.

- A signal can be described by its amplitude, frequency and phase shift.
- Accordingly, three methods of modulation are used and they are :
    1) Amplitude Modulation (AM),
    2) Angle modulation (Frequency Modulation (FM) and Phase Modulation(PM)).

Amplitude Modulation

- In amplitude modulation, the amplitude of a carrier signal is varied by a modulating voltage signal whose frequency is much lower than that of the carrier.

Frequency Modulation

- Frequency modulation is a system in which the amplitude of the modulated carrier is kept constant, while its frequency is varied by the modulating signal.

Phase Modulation

- Phase modulation is a system in which the amplitude of the modulated carrier is kept constant, while its phase is varied by the modulating signal.

- However, it should be noted that, a digital signal is superior to an analog signal because it is more robust to noise and can easily be recovered, corrected and amplified.

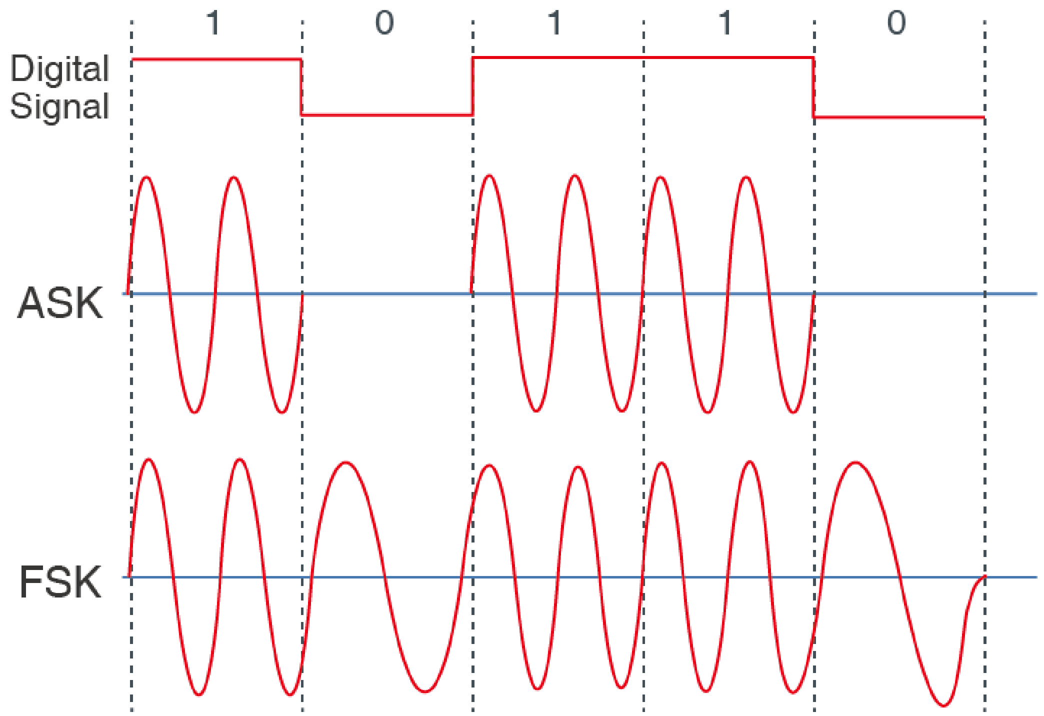
Digital Modulation

ASK (Amplitude Shift Keying)

- A digital modulation method that sends transmission data by varying the presence/absence of analog signals.

FSK (Frequency Shift Keying)

- This technique utilizes the difference in the amplitude of analog signals to modulate digital signals by switching between low frequency and high frequency in order to represent 0 and 1.



OFDM (Orthogonal Frequency Division Multiplexing)

- A multicarrier digital modulation method that transmits large amounts of data over multiple closely spaced data streams.

DSSS (Direct Sequence Spread Spectrum)

- A type of spread spectrum method utilizing a direct spread technique.
- Data signals are spread over a wide frequency band at low power.

PULSE MODULATION (UNCODED)

- In pulse-modulation systems, information is conveyed by modulating some parameter of the transmitted pulses such as the amplitude, duration, time of occurrence, or shape of pulse.
- This type of modulation is based on the "sampling principle," which states that a continuous message waveform that has a spectrum of finite width could be recovered from a set of discrete instantaneous samples whose rate is higher than twice the highest signal frequency.
- In *pulse-amplitude modulation* (PAM), the series of periodically recurring pulses is modulated in amplitude by the corresponding instantaneous samples of the message function.

- Pulse-duration, pulse-position, and pulse-frequency modulation are particular forms of pulse-time modulation.
- In *pulse-duration modulation* (PDM) also called *pulse-length* or *pulse-width modulation* (PWM), the time of occurrence of either the leading or trailing edge of each pulse (or both) is varied from its unmodulated position by the samples of the modulating wave.
- In *pulse-position* (or *phase*) *modulation* (PPM), the samples of the modulating wave are used to vary the position in time of a pulse, relative to its unmodulated time of occurrence.
- Pulse-position modulation is essentially the same as PDM, except that the variable edge is now replaced by a short pulse.
- In *pulse-frequency modulation* (PFM), the samples of the message function are used to modulate the frequency of the series of carrier pulses.

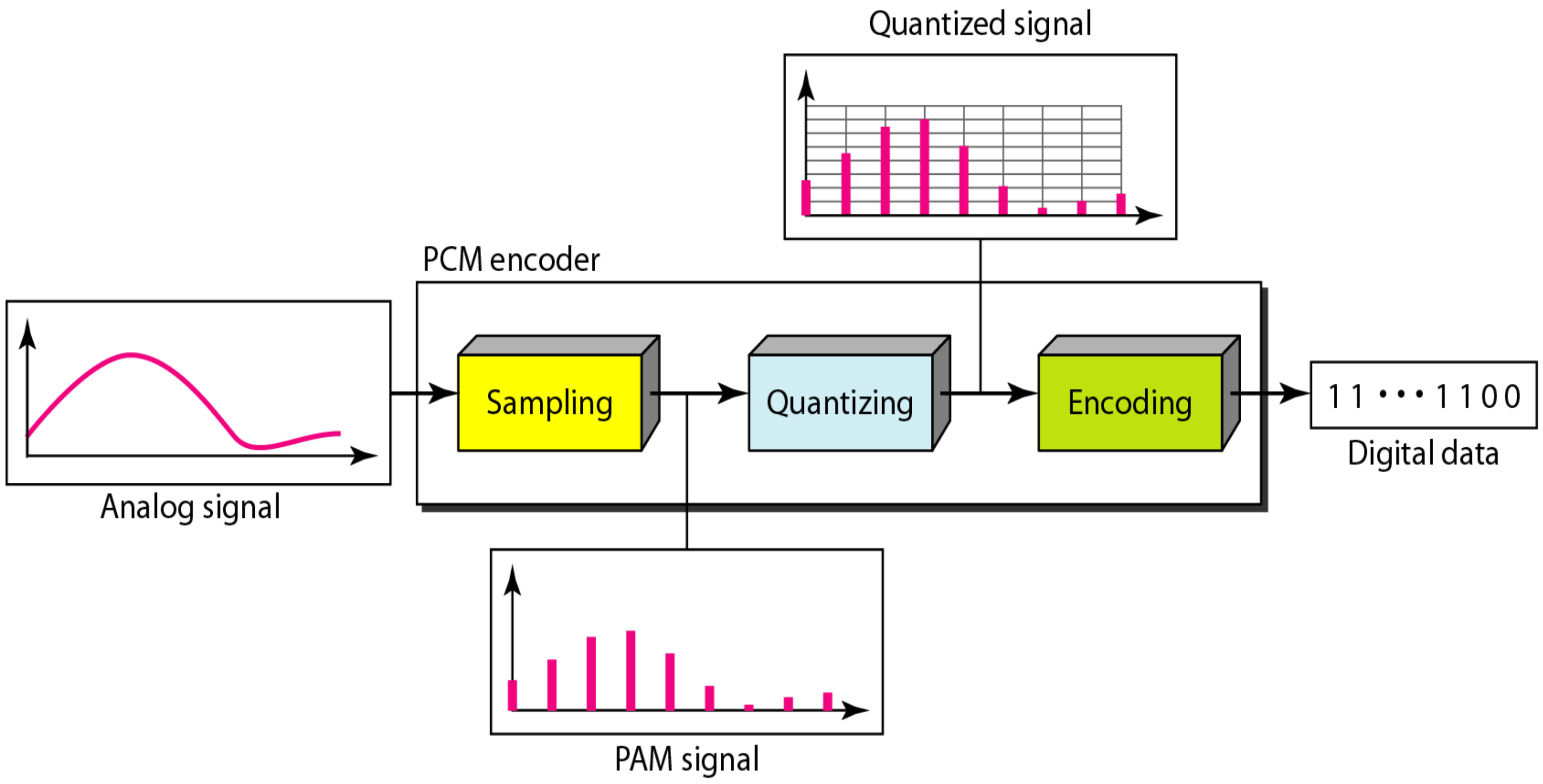
- Another example of a code-modulation system is *delta modulation*. This quantized the gap between the observed and previous steps instead of value of the input signal waveform.
- As in PCM, the range of signal amplitudes is quantized, and binary pulses are produced at the sending end at regular intervals.
- However, in delta-modulation systems, instead of the absolute quantized signal amplitude being transmitted at each sampling, the transmitted pulses carry the information corresponding to the derivative of the amplitude of the modulating signal.

Pulse Code Modulation (PCM)

- In PCM, the modulating signal waveform is sampled at regular intervals as in conventional pulse modulation.
- In PCM, the samples are first quantized into discrete steps; i.e., within a specified range of expected sample values, only certain discrete levels are allowed, and these are transmitted over the system by means of a code pattern of a series of pulses.

- 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, i.e. remove high frequency components that affect the signal shape.

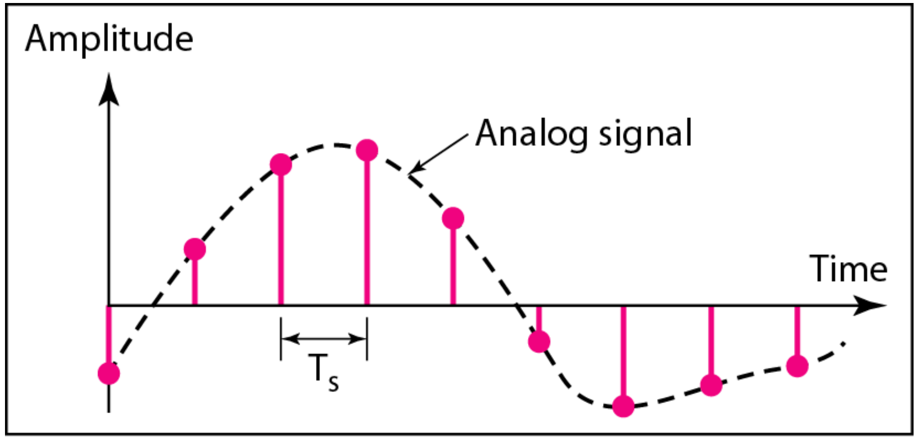
Figure Components of PCM encoder



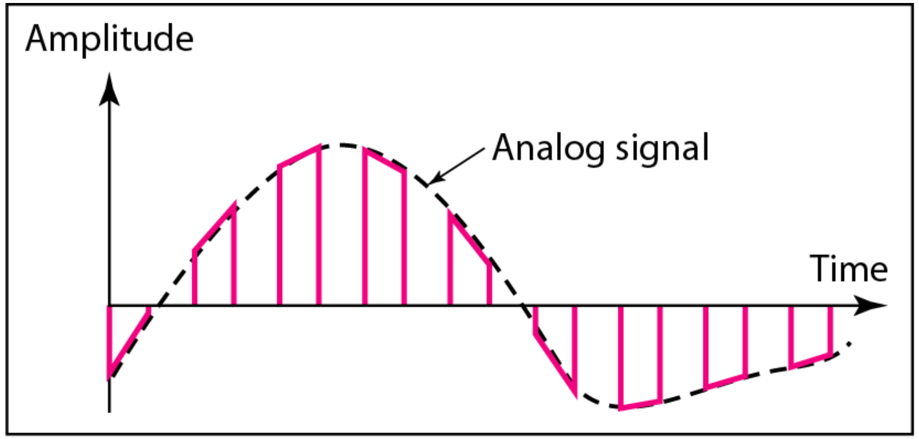
Sampling

- 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

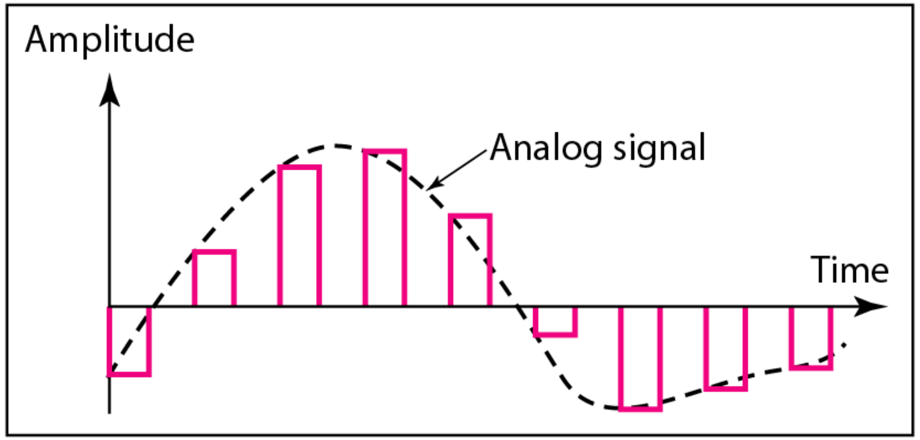
Figure Three different sampling methods for PCM



a. Ideal sampling



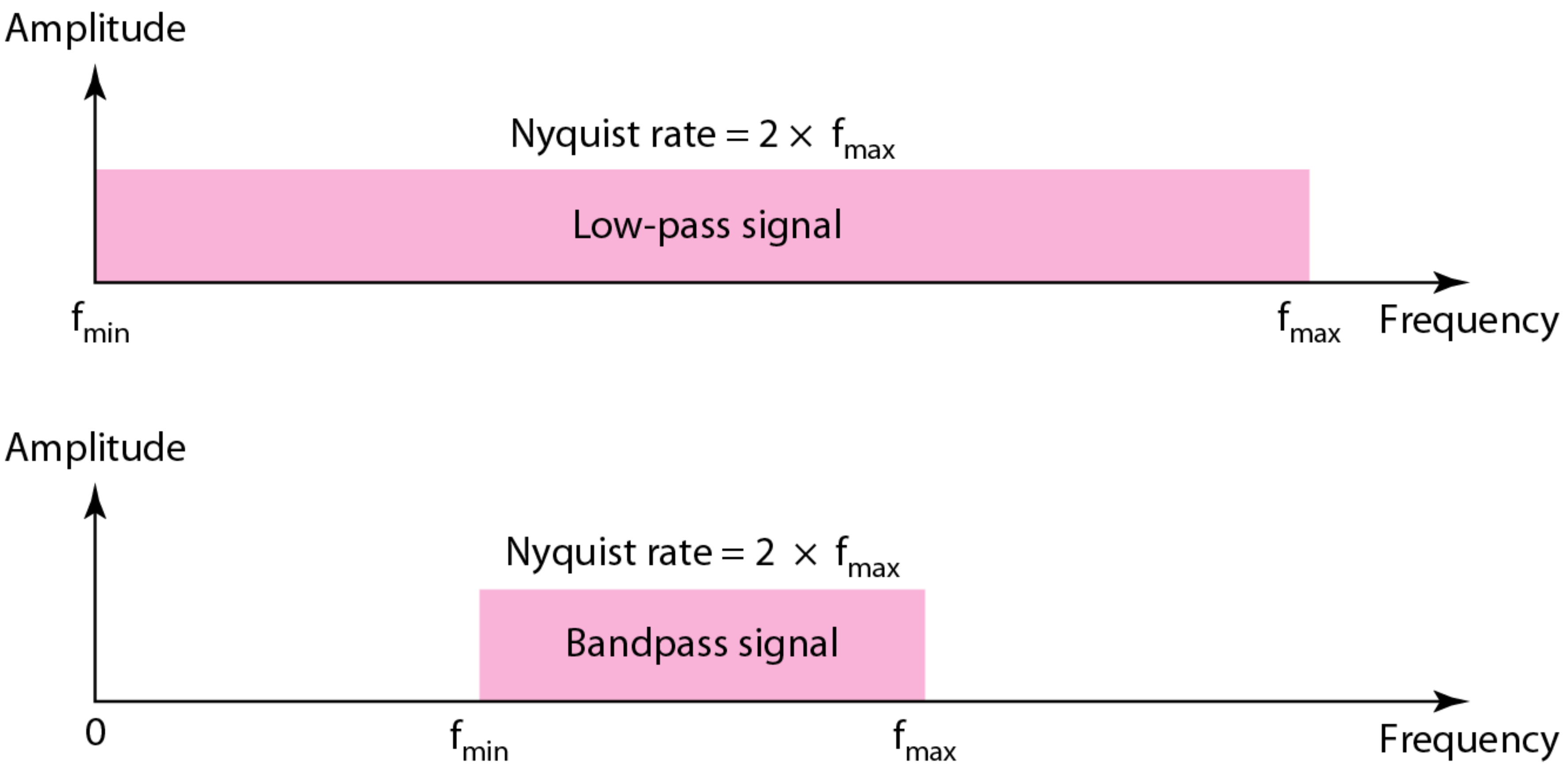
b. Natural sampling



c. Flat-top sampling

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

Figure Nyquist sampling rate for low-pass and band-pass signals



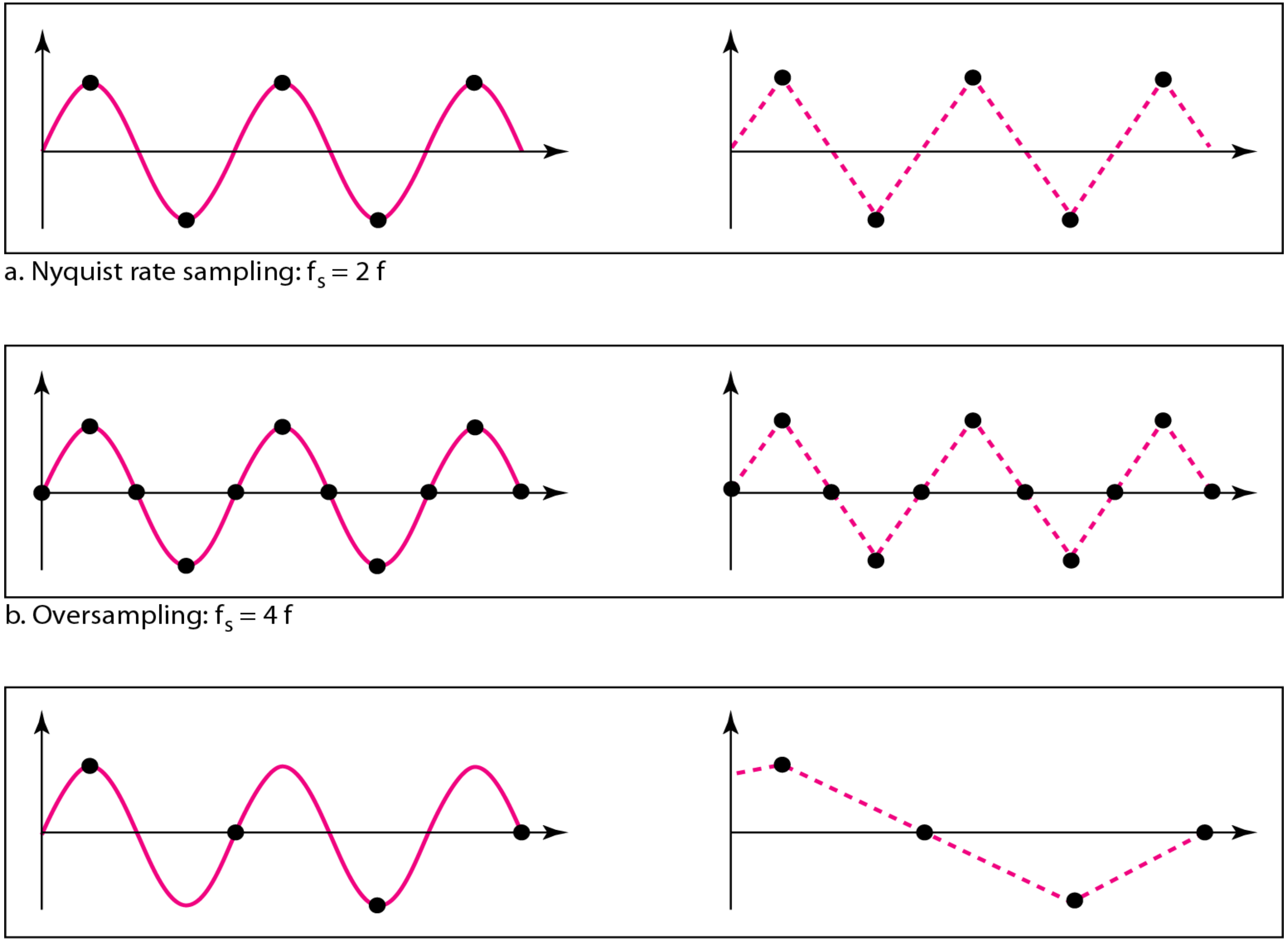
Example

- 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).

- 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 Recovery of a sampled sine wave for different sampling rates



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

b. Oversampling: $f_s = 4f$

c. Undersampling: $f_s = f$

Quantization

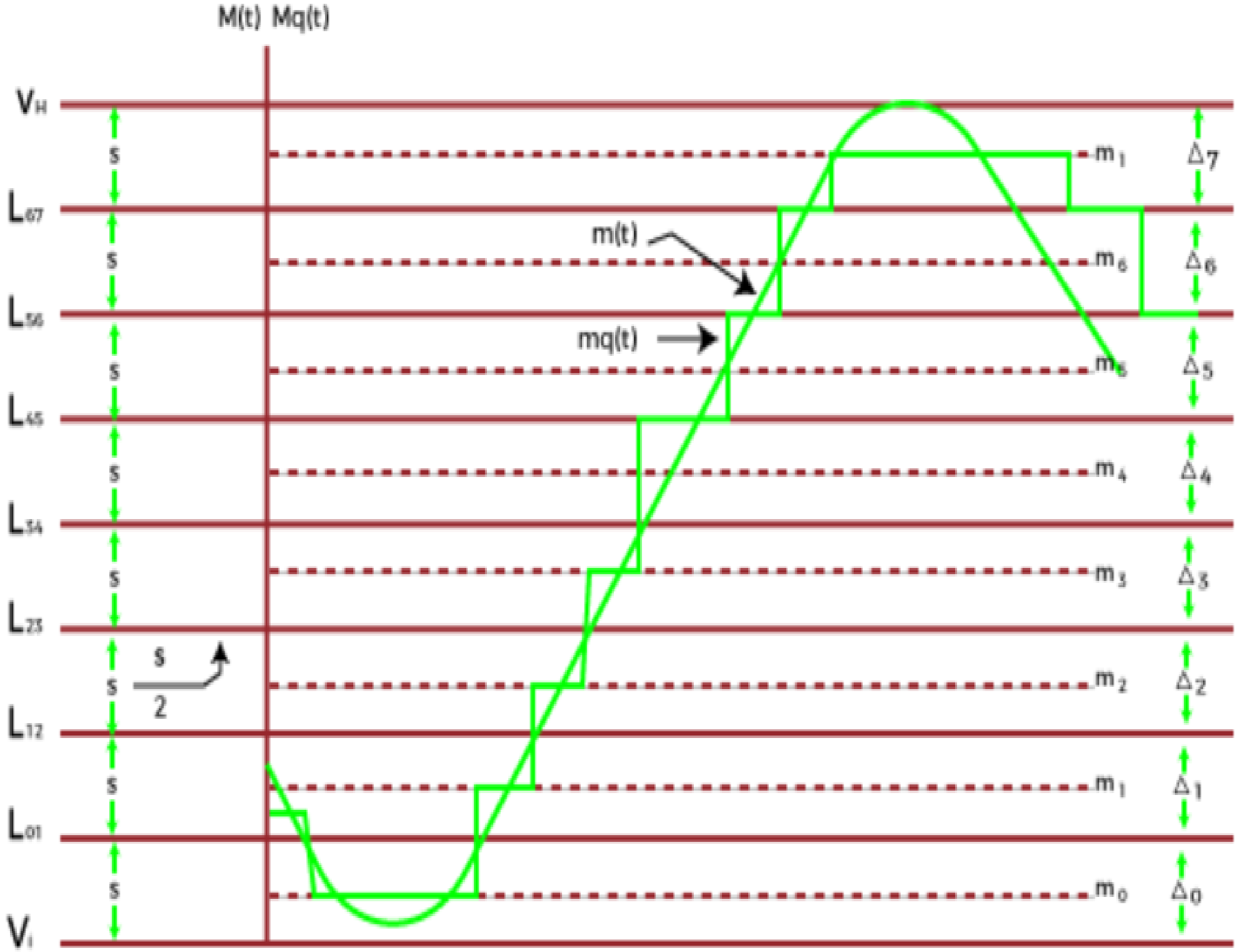
- Sampling results in a series of pulses of varying amplitude values ranging between two limits: a min and a max.
- The amplitude values are infinite between the two limits.
- We need to map the *infinite* amplitude values onto a finite set of known values.
- This is achieved by dividing the distance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

- The resolution or size of the steps is Δ, which depends on the resolution of the A/D conversion process.
- In particular, if a 12-bit A/D converter has a range of ±10 V, then its resolution or quantization level is given by

$$\text{Resolution} = \frac{\text{Range}}{2^n}$$

where *n*=number of bits

$$\Delta = \frac{20}{2^{12}} = 0.005V$$

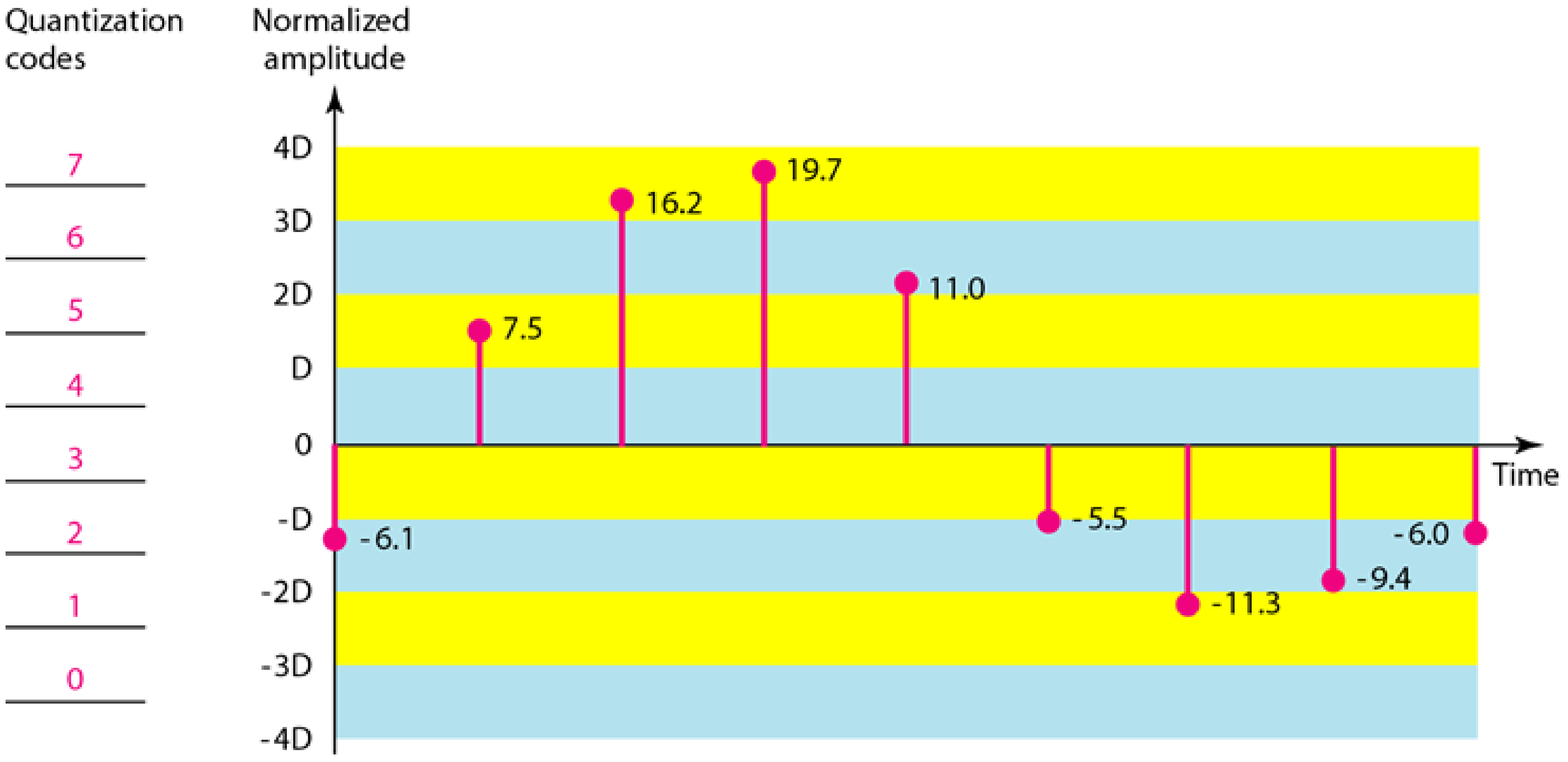
Assigning Codes to Zones

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

Figure Quantization and encoding of a sampled signal



Quantization Error

- When a signal is quantized, we introduce an error - 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

Quantization Error and SN_QR

- Signals with lower amplitude values will suffer more from quantization error as the error range: $\Delta/2$, is fixed for all signal levels.
- Non linear quantization is used to alleviate this problem.
- Goal is to keep SN_QR fixed for all sample values.
- Two approaches:
  - The quantization levels follow a logarithmic curve.

    Smaller Δ's at lower amplitudes and larger Δ's at higher amplitudes.
  - **Companding**: The sample values are compressed at the sender into logarithmic zones, and then expanded at the receiver.

    The zones are fixed in height.

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

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

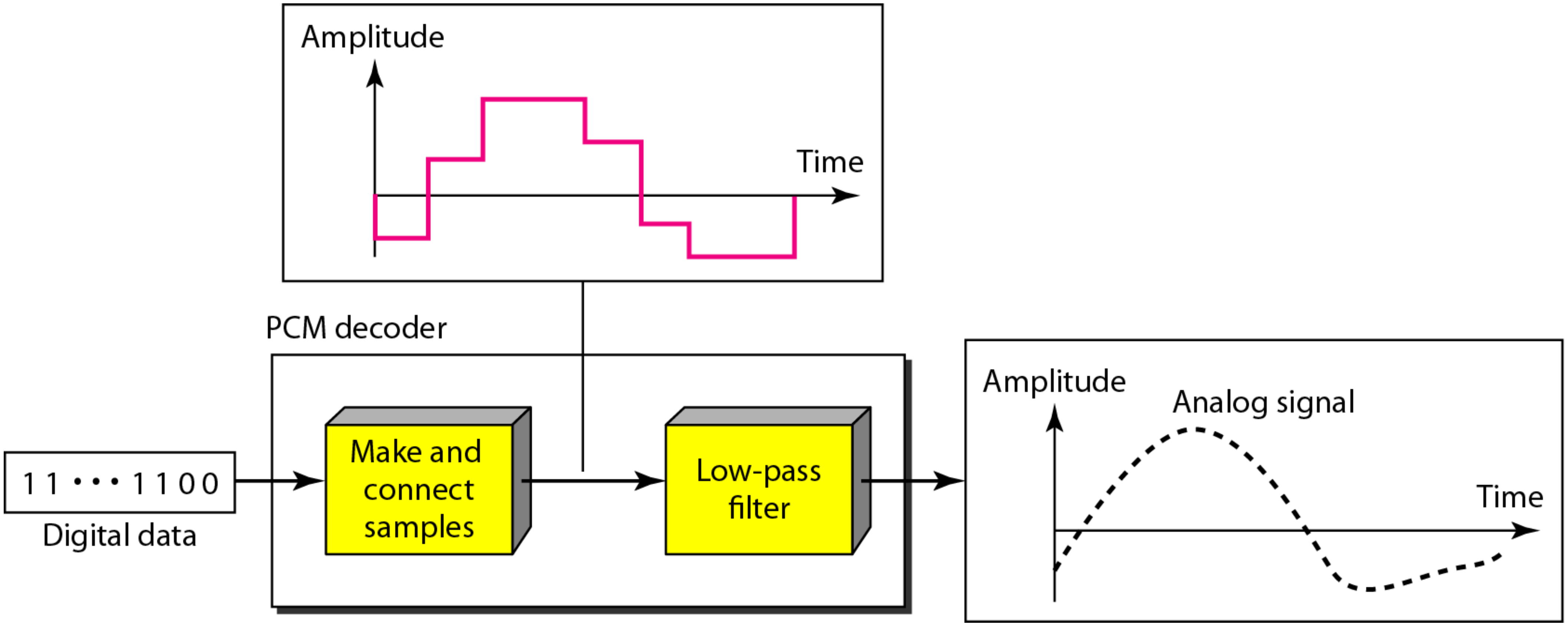
- The bandwidth required to transmit this signal depends on the type of line encoding used. Refer to previous section for discussion and formulas.
- A digitized signal will always need more bandwidth than the original analog signal.

  Price we pay for robustness and other features of digital transmission.

PCM Decoder

- To recover an analog signal from a digitized signal we follow the following steps:
  - ☆ We use a hold circuit that holds the amplitude value of a pulse till the next pulse arrives.
  - ☆ 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 a signal is recovered.

Figure Components of a PCM decoder



Delta Modulation (UNCODED)

- 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.

Figure The process of delta modulation

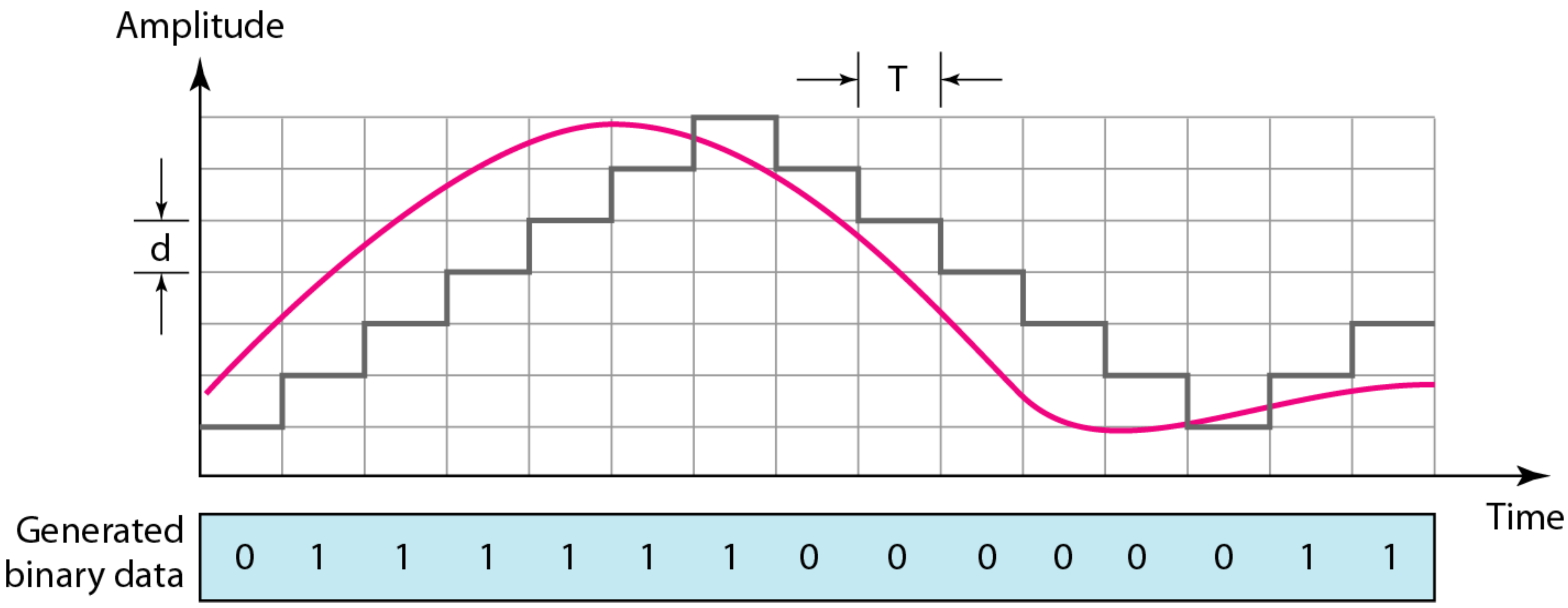


Figure Delta modulation components

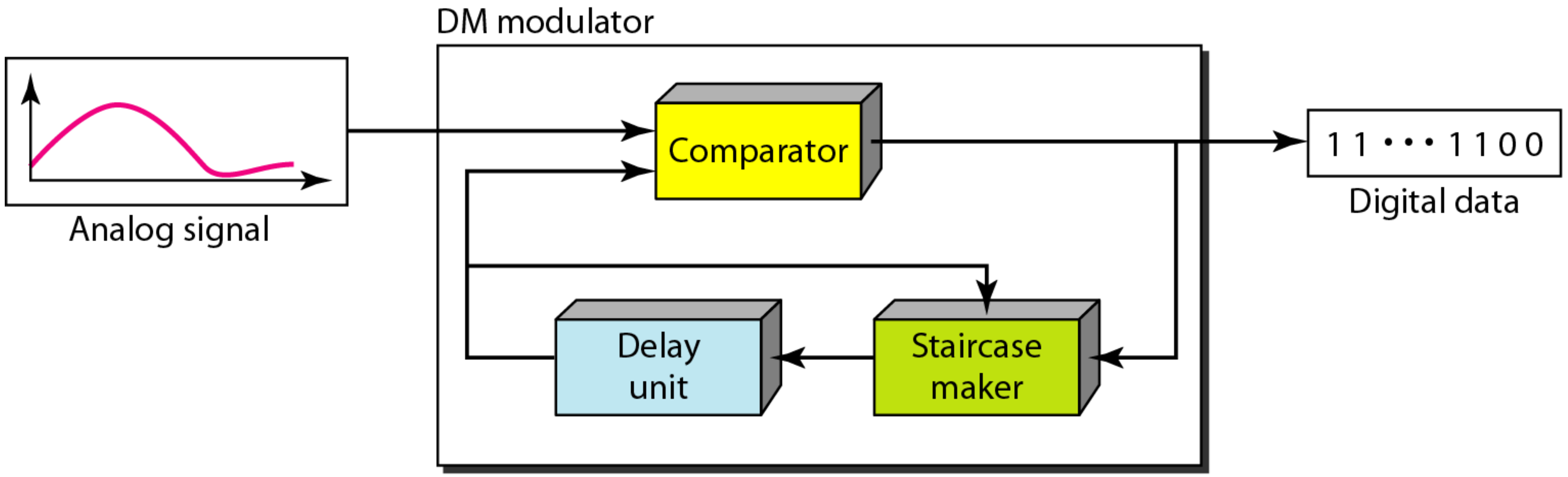
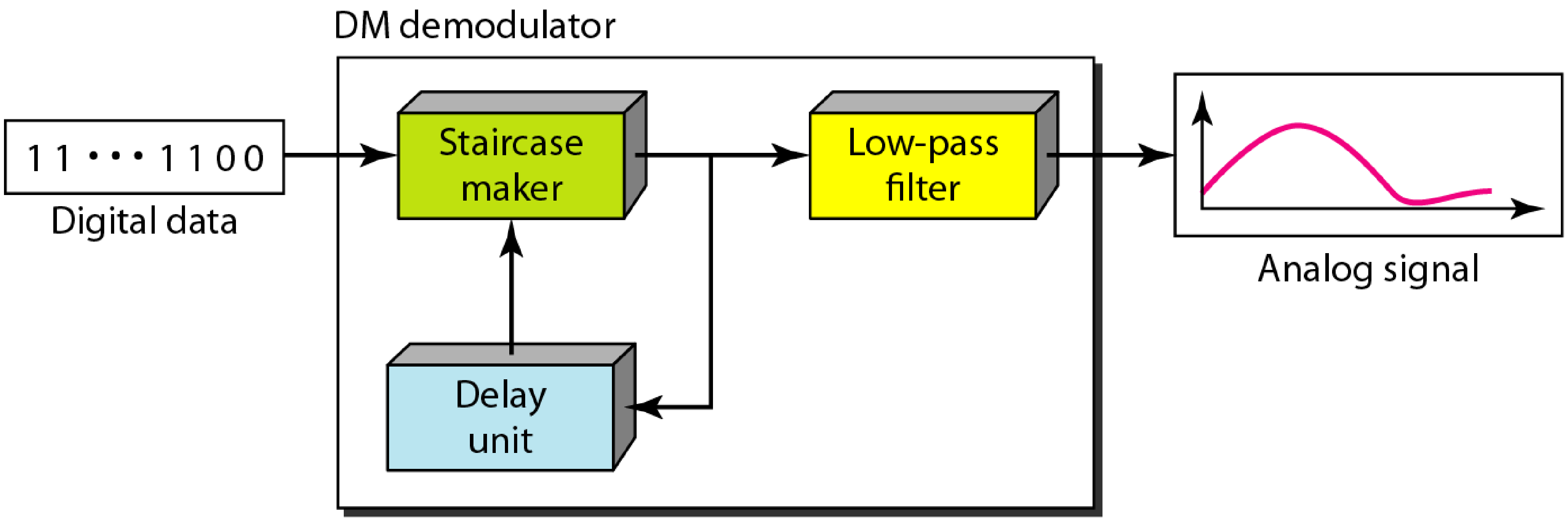


Figure Delta demodulation components



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.

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 Data transmission and modes

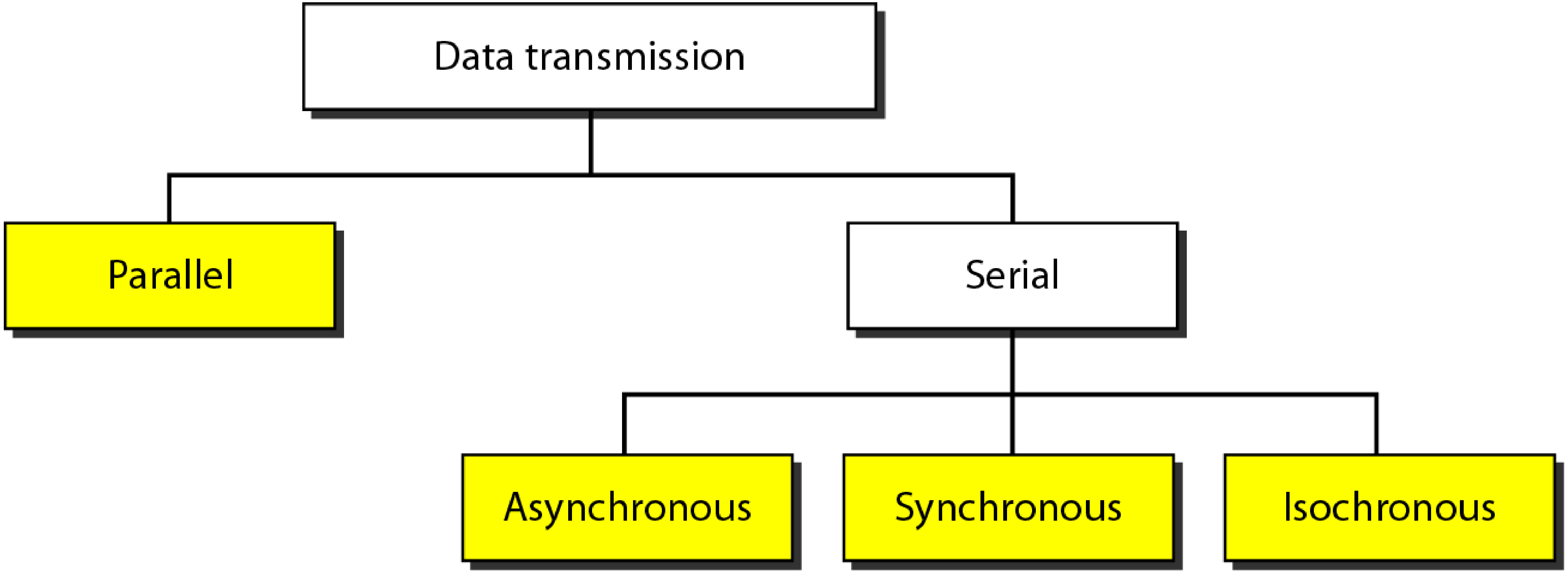


Figure Parallel transmission

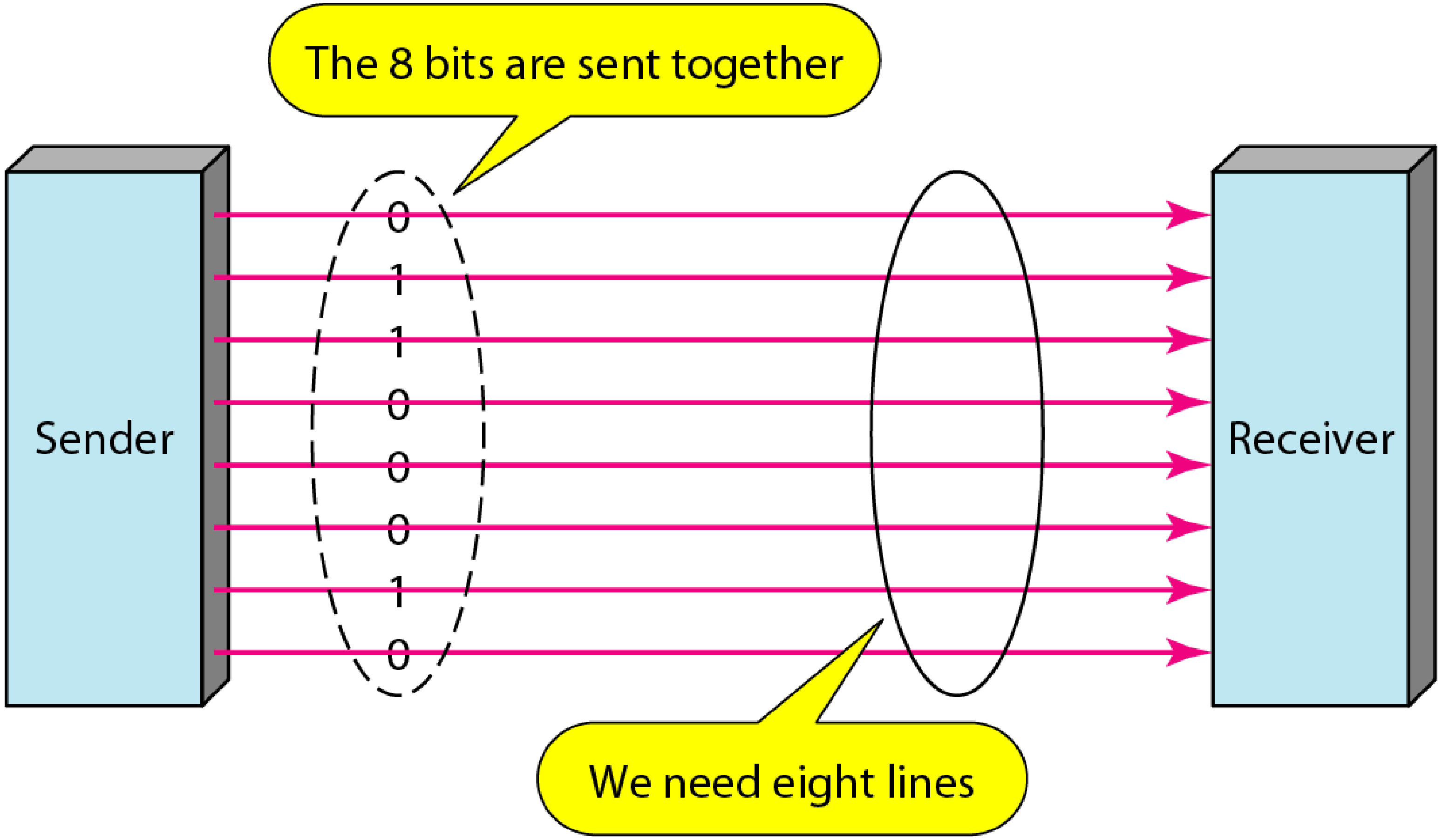
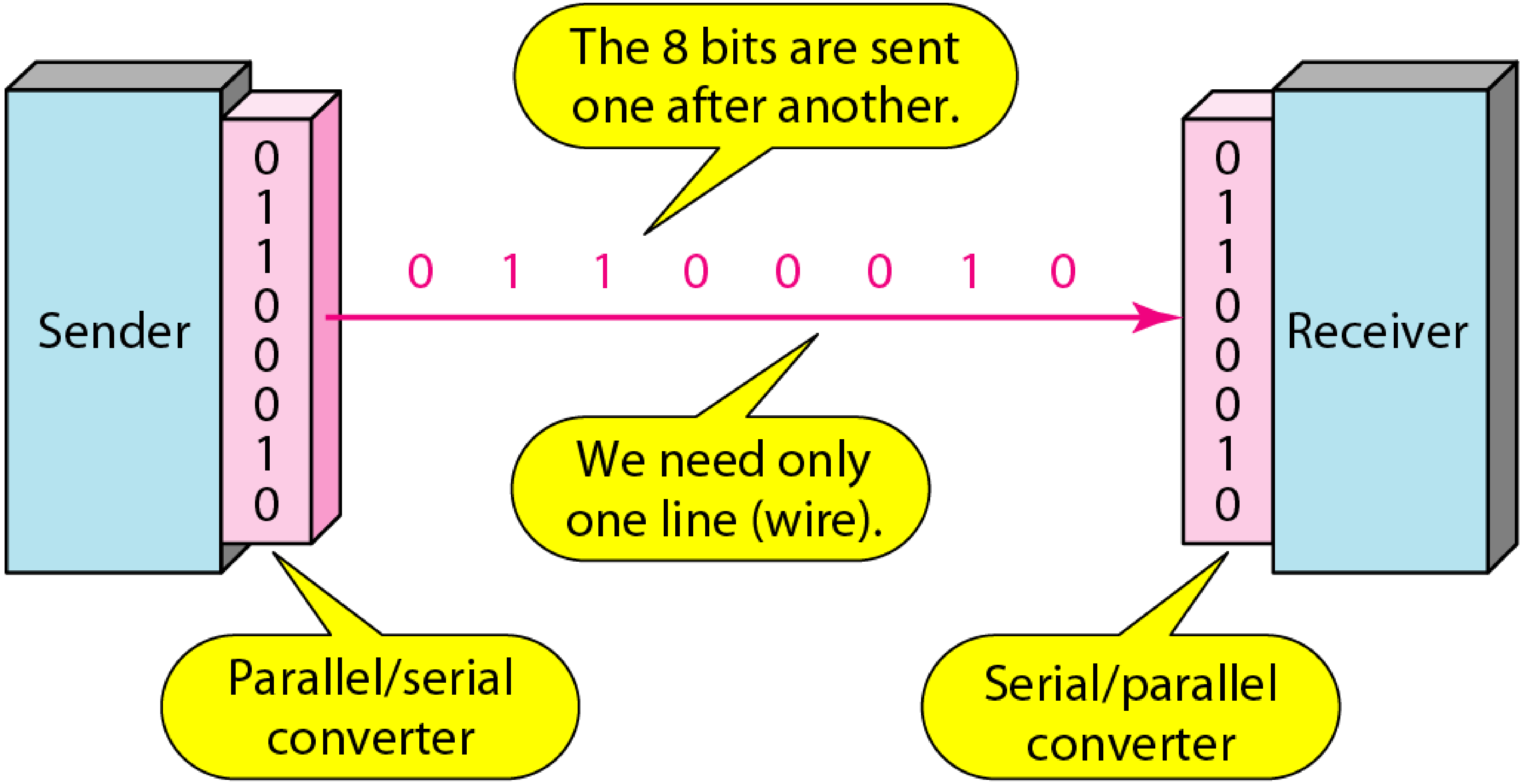


Figure Serial transmission

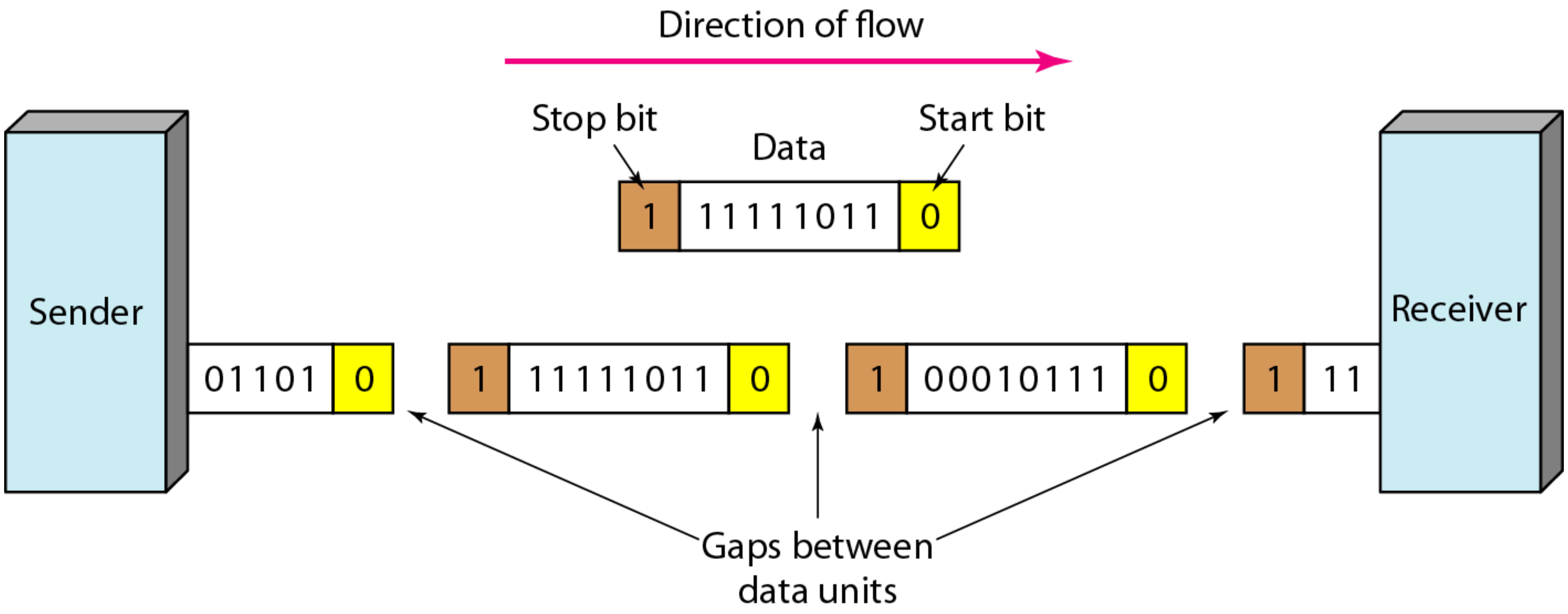


- In asynchronous transmission, we send 1 start bit (0) at the beginning and 1 or more stop bits (1s) at the end of each byte.
- There may be a gap between each byte.

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

- 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 bits are usually sent as bytes and many bytes are grouped in a frame.
- A frame is identified with a start and an end byte.

Figure Asynchronous transmission



$$\text{Efficiency} = \frac{\text{Actual Data bits}}{\text{Total Bits}} \times 100\%$$

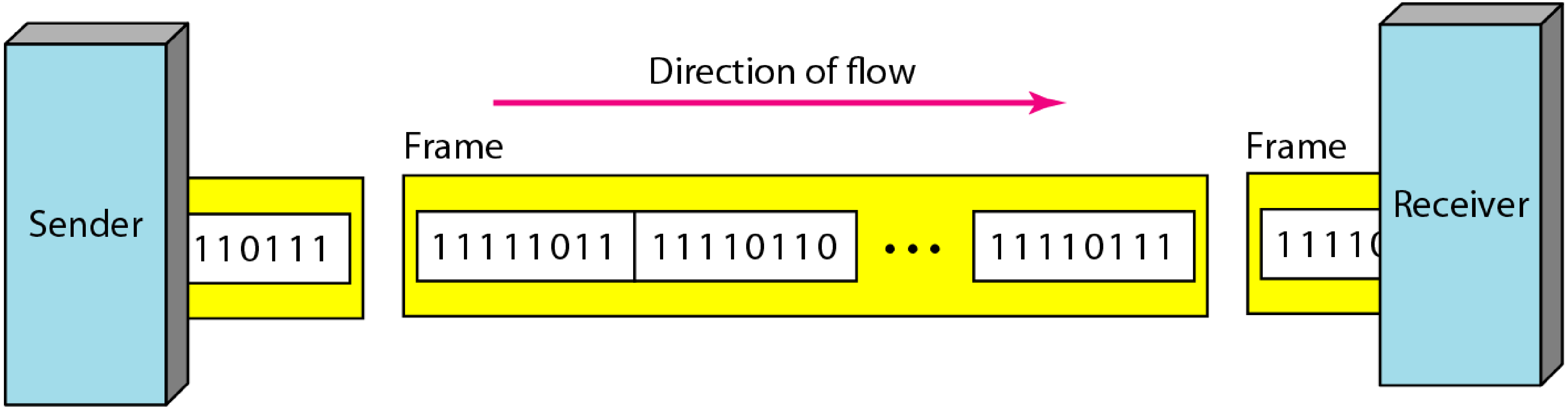
Here,

Actual data bits refers to the amount of data bits to be sent.

Total bits refers to the sum of Actual data bits and Overhead data bits.

Overhead data bits are start bit(1 bit) , stop bit(1 bit) & parity bit(1 bit) .

Figure Synchronous transmission



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 frames, 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.

- Isochronous transmission is similar to synchronous transmission but the time interval between blocks is almost zero.

Advantages

- Transmission speed is much higher.
- There is no need to pause between each character.
- Start bit at the beginning of each character and Stop bit at the end is not required.