Chapter 5: Introduction to Digital Communication

Addis Ababa Institute of Technology
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Undergraduate Program

School of Electrical and Computer Engineering

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- Introduction to Digital Communication
- Analog to digital conversion
- Baseband and Band pass Transmission of Digital Data
  - Line coding
  - Digital to analog conversion

Digital communications is:

- Simply the practice of exchanging information by using finite sets of signals.
- Is all about coding (at least at the physical layer).
- Is concerned with answering the question
  - *Which signal was transmitted?*

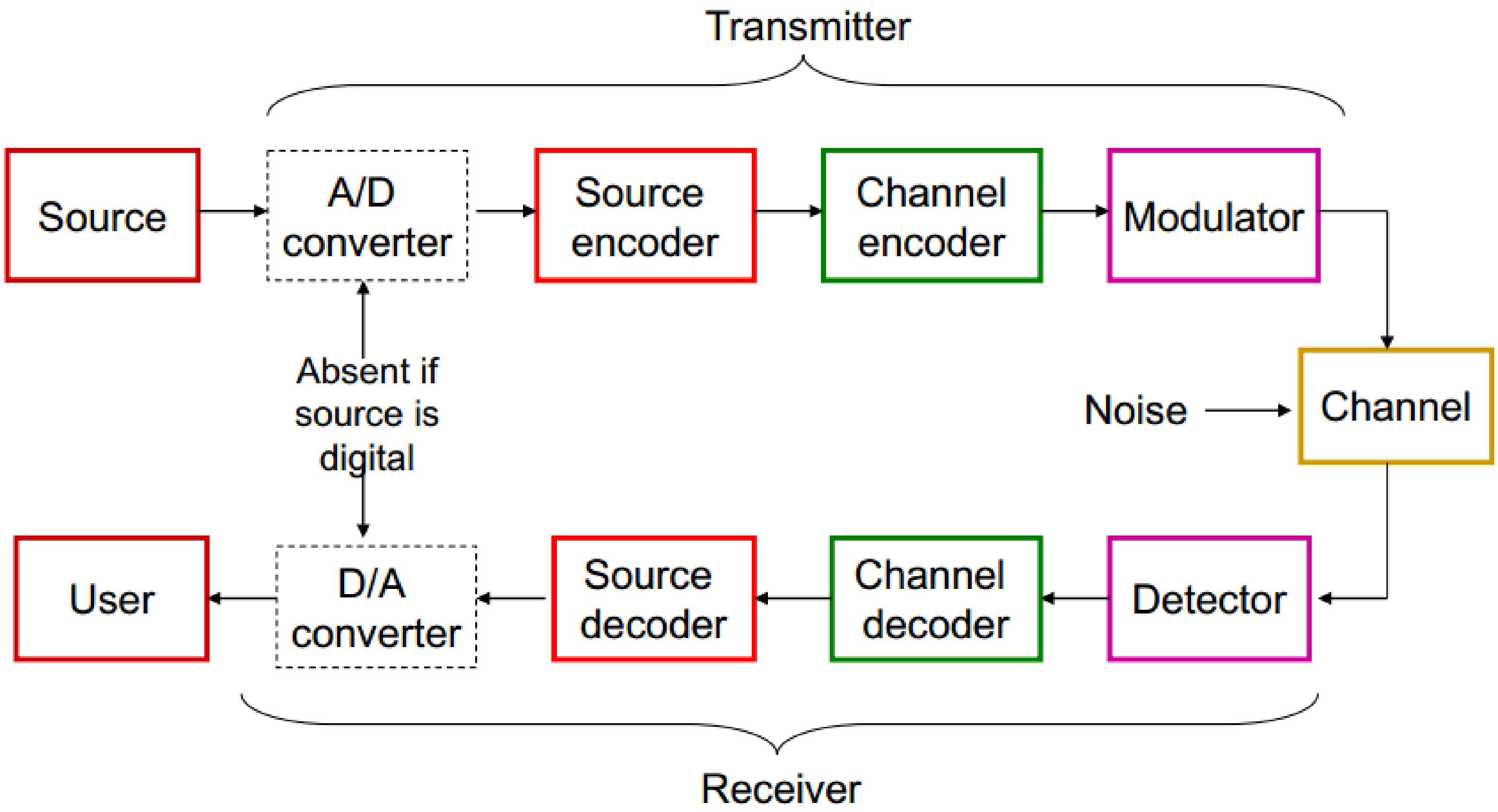
Analog ? Digital ?

Why Digital Communication ??

- ***Cheap Hardware***
  - flexible, complex hardware is becoming cheaper all the time.
- ***Demand for new services***
  - e.g. email, ecommerce, teleworking, network applications etc.
- ***Control of Quality***
  - i.e. powerful error control techniques can be employed to guarantee high quality communication.
- ***Compatibility and flexibility***
  - digital signals are easier and cheaper to store and process than analogue signals.
- ***Transmission***
  - digital signals are more spectrally efficient than analogue signals which means more information can be carried in a given bandwidth.
- ***Security***
  - powerful encryption techniques exist to ensure the confidentiality and integrity of the information, critical for ecommerce applications

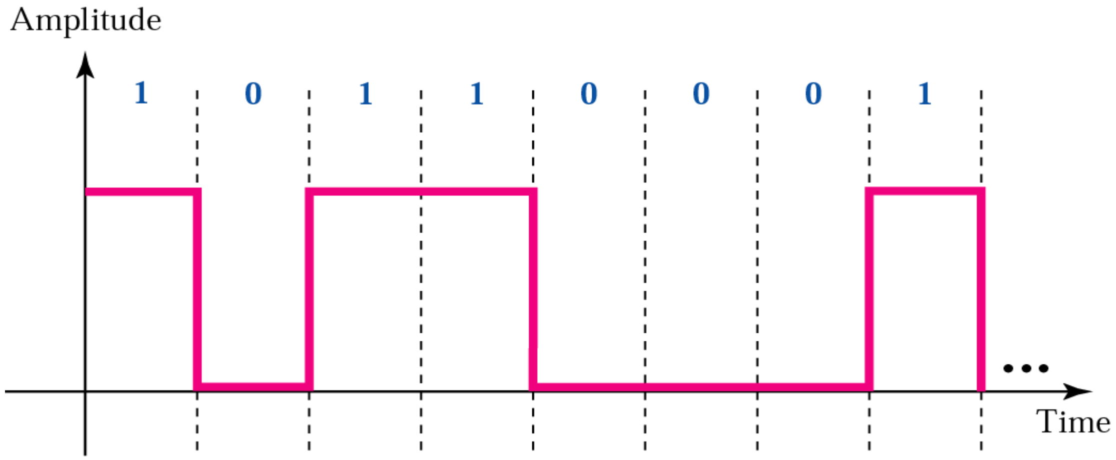


Digital Communication System

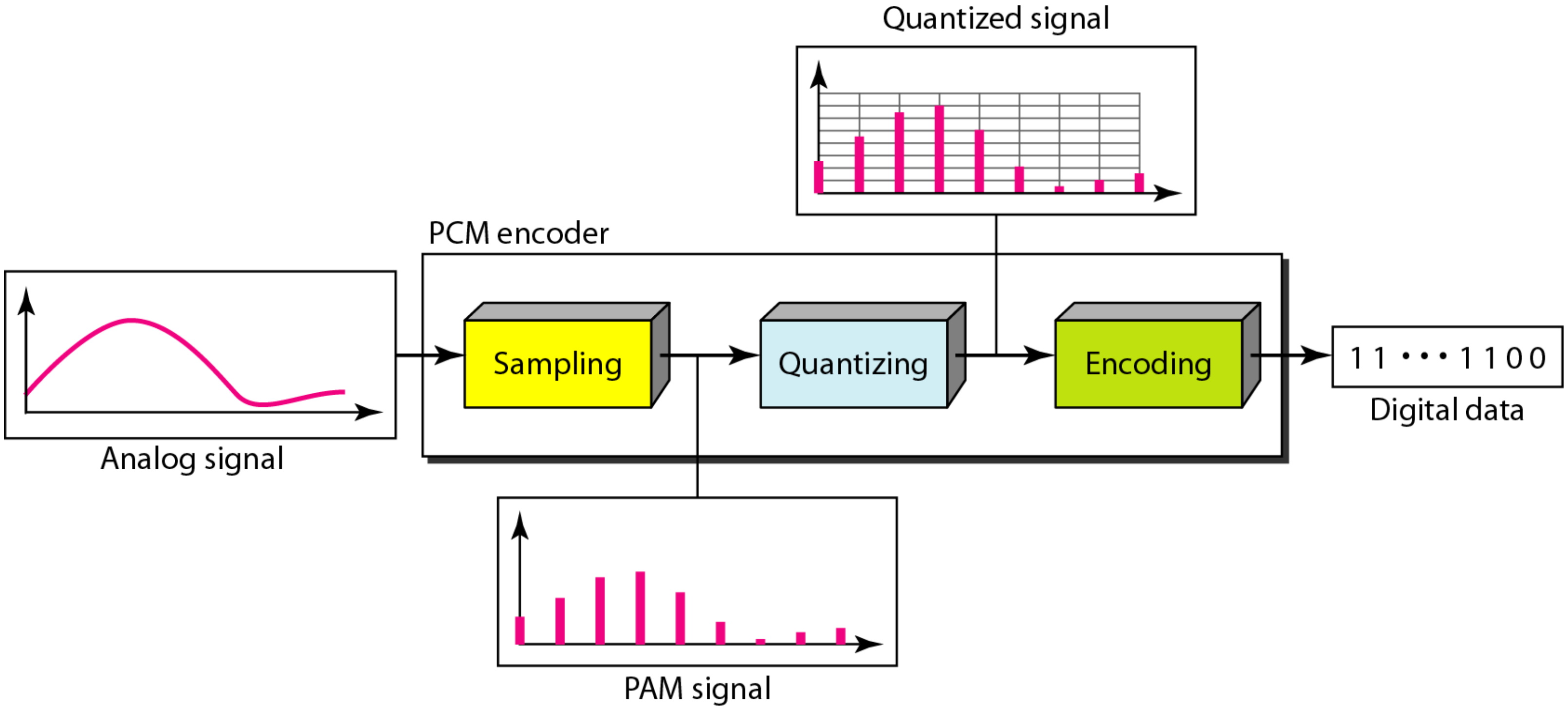


Digital terms

- Pulse
- Pulse duration
- Pulse amplitude
- Signal strength



Analog to digital conversion



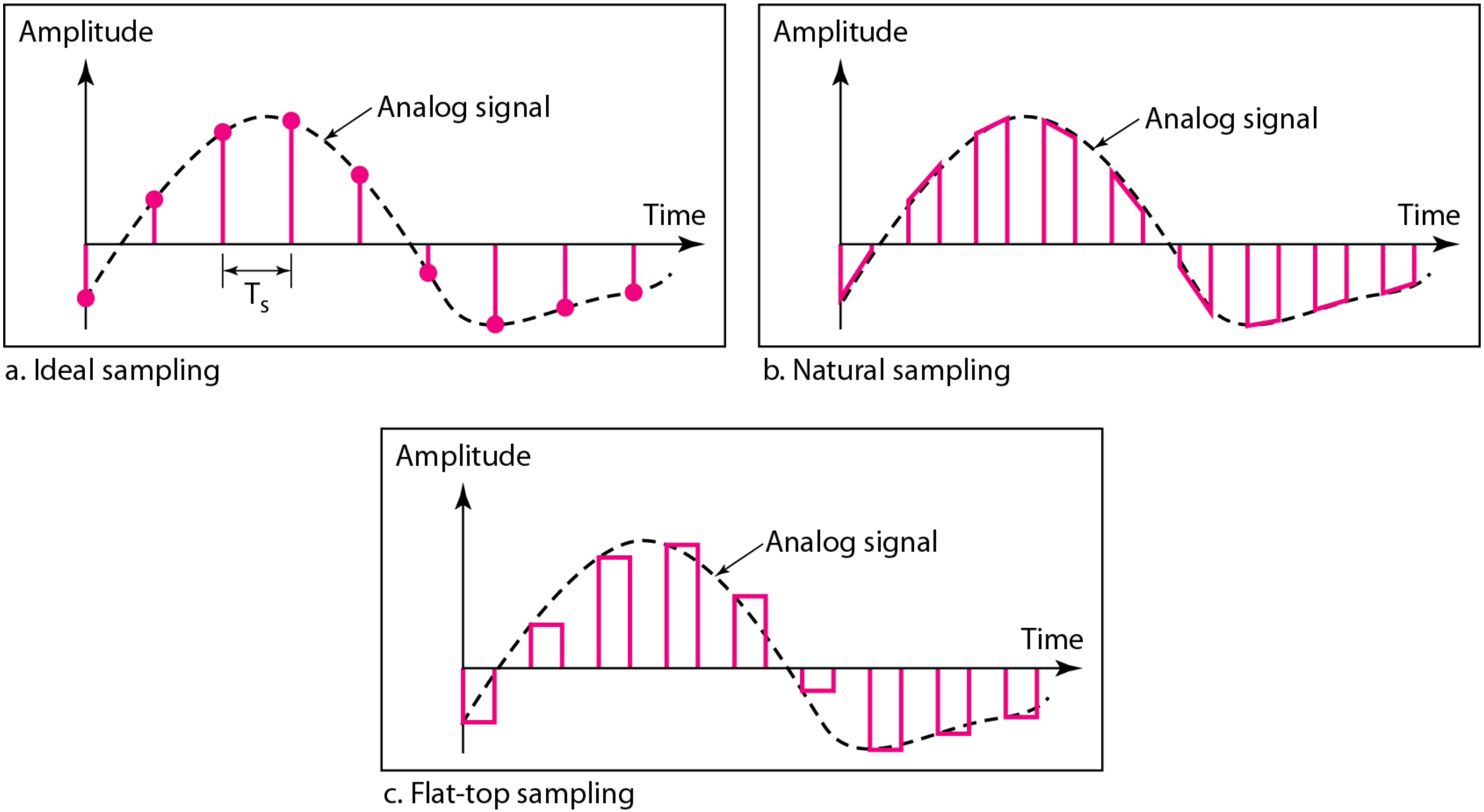
Analog to digital conversion ….

- ***Sampling***: an analog signal is converted into a corresponding sequence of samples that are usually spaced uniformly in time.
  - Proper selection of the sampling rate, so that the sequence of samples can uniquely define the original analog signal!!!
  - anti-aliasing filter might be used.
- ***Quantization:*** Approximating (rounding –off) the sampled values to a finite number of discrete amplitude levels.
- ***Encoding***: The discrete signals coded in binary form.

Sampling

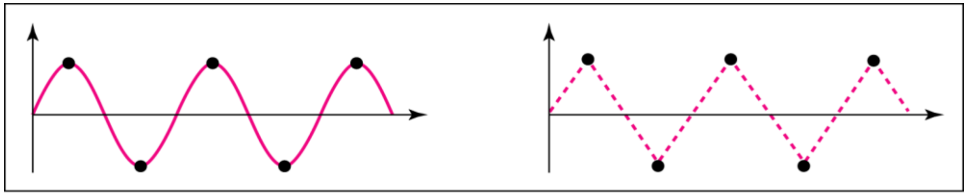
- Analog signal is sampled every T_S sec.
  - T_s is referred to as the sampling interval.
  - $f_s = 1/T_s$ is called the sampling rate or sampling frequency.
- According to the *Nyquist theorem*, the sampling rate must be *at least 2 times the highest frequency* contained in the signal.
- There are 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

Three different sampling methods for PCM

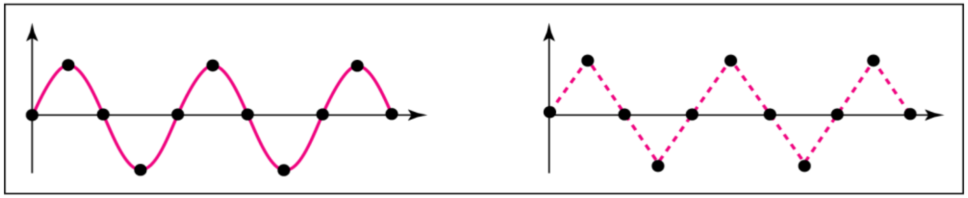


Cont…

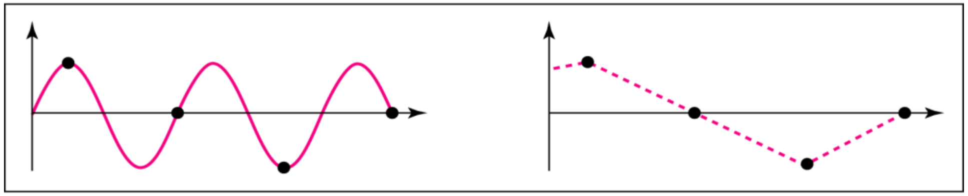
- *For an intuitive example of the Nyquist theorem, let us sample a simple sine wave at three sampling rates: $f_s = 4f$ (2 times the Nyquist rate), $f_s = 2f$ (Nyquist rate), and $f_s = f$ (one-half the Nyquist rate). Figure shows the sampling and the subsequent recovery of the signal*



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



b. Oversampling: $f_s = 4f$



c. Undersampling: $f_s = f$

Example 1:

- Standard sampling rate for telephone network is adopted to be 8000 Hz.

Example 2:

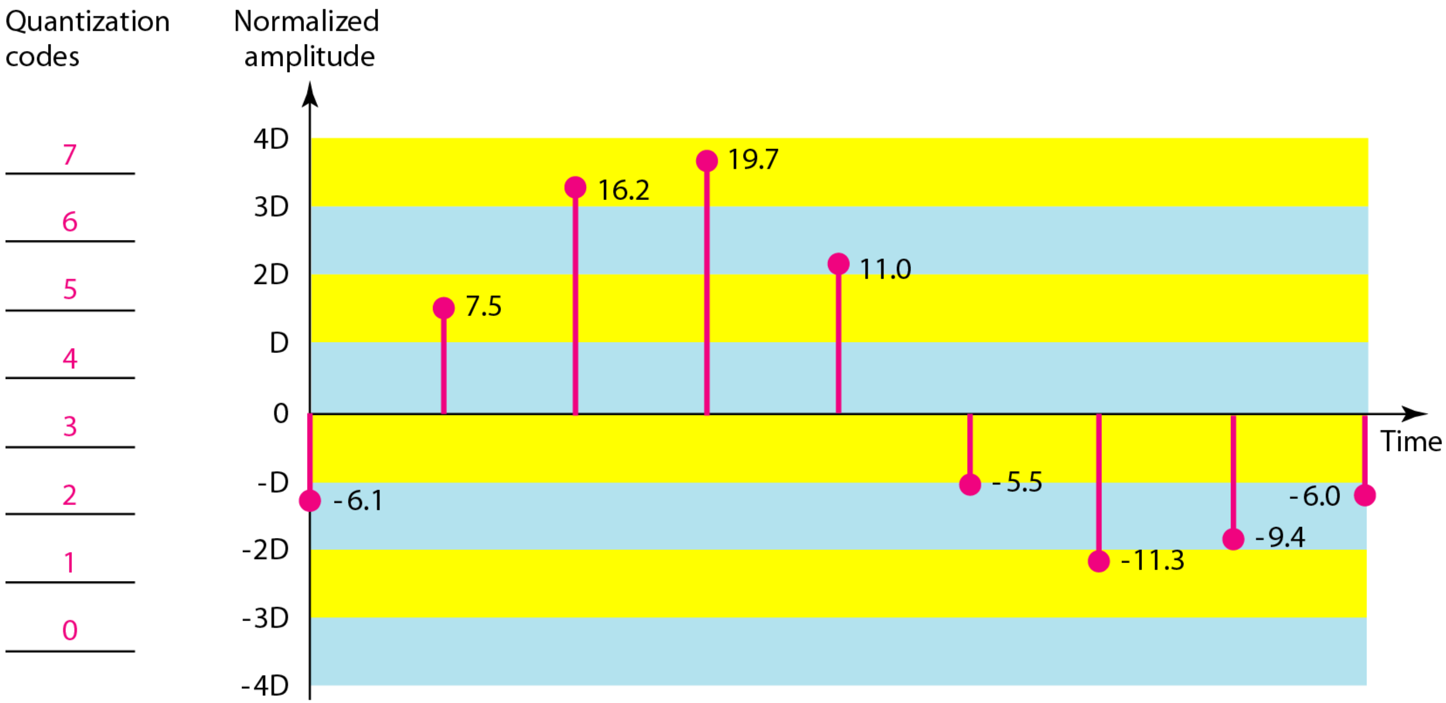
- A complex low-pass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?

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 quantization levels (zones), each of height Δ.

$$\Delta = (\max - \min)/L \ldots\ldots \text{ uniform quantization}$$

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

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

Encoding : 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.

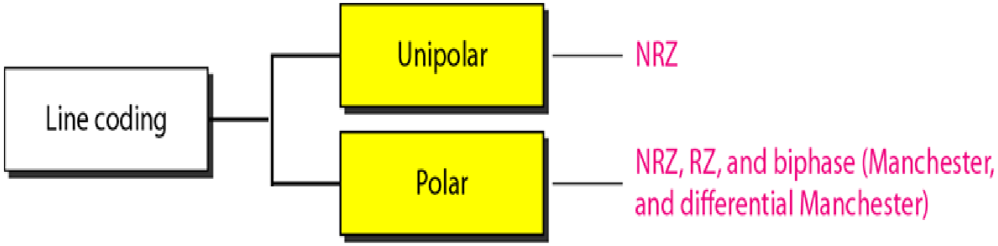
Quantization Error

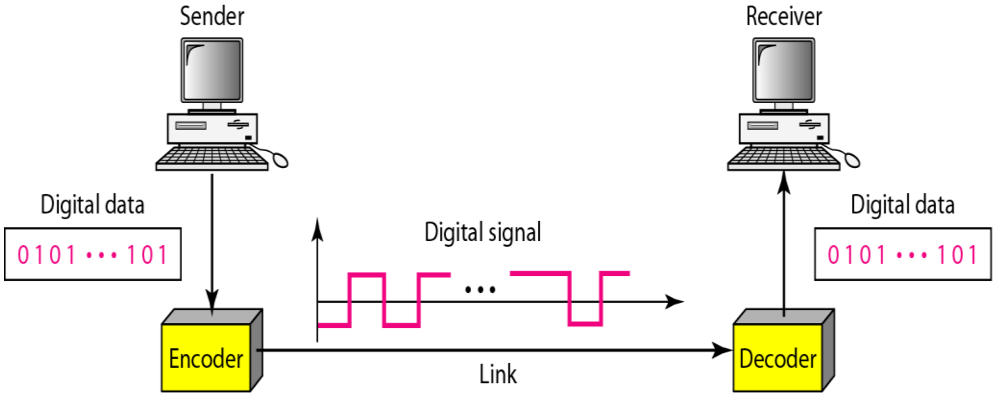
- When a signal is quantized, we introduce an error , max of $\Delta/2$ - the coded signal is an approximation of the actual amplitude value.
  - Irreversible Noise of $\Delta^2/12$
- The difference between actual and coded value (midpoint) is referred to as the quantization error.
  - The more zones, the smaller Δ which results in smaller errors.
  - BUT, the more zones the more bits required to encode the samples -> ***higher bit rate***
- The bit rate of a PCM signal can be calculated form the number of bits per sample x the sampling rate

$$Bit\ rate\ = n_b\, x\ f_s$$

Line Coding: Baseband Transmission of Digital Data

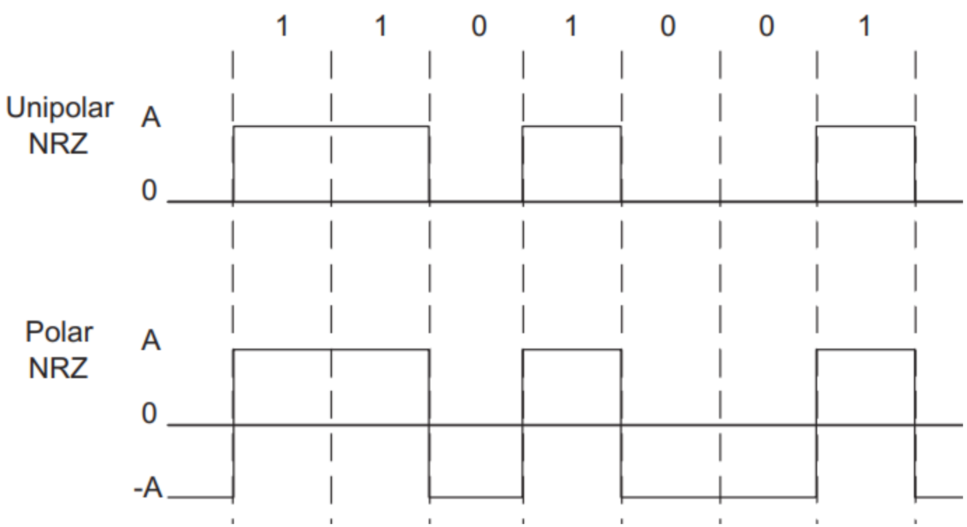
- Electrical representation of that binary sequence
  - Converting a string of 1's and 0's (digital data) into a sequence of signals that denote the 1's and 0's.
- For example a high voltage level (+V) could represent a "1" and a low voltage level (0 or -V) could represent a "0".





Non-Return To Zero

- All signal levels are on one side of the time axis - either above or below
- 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.



- 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.

Cont.…

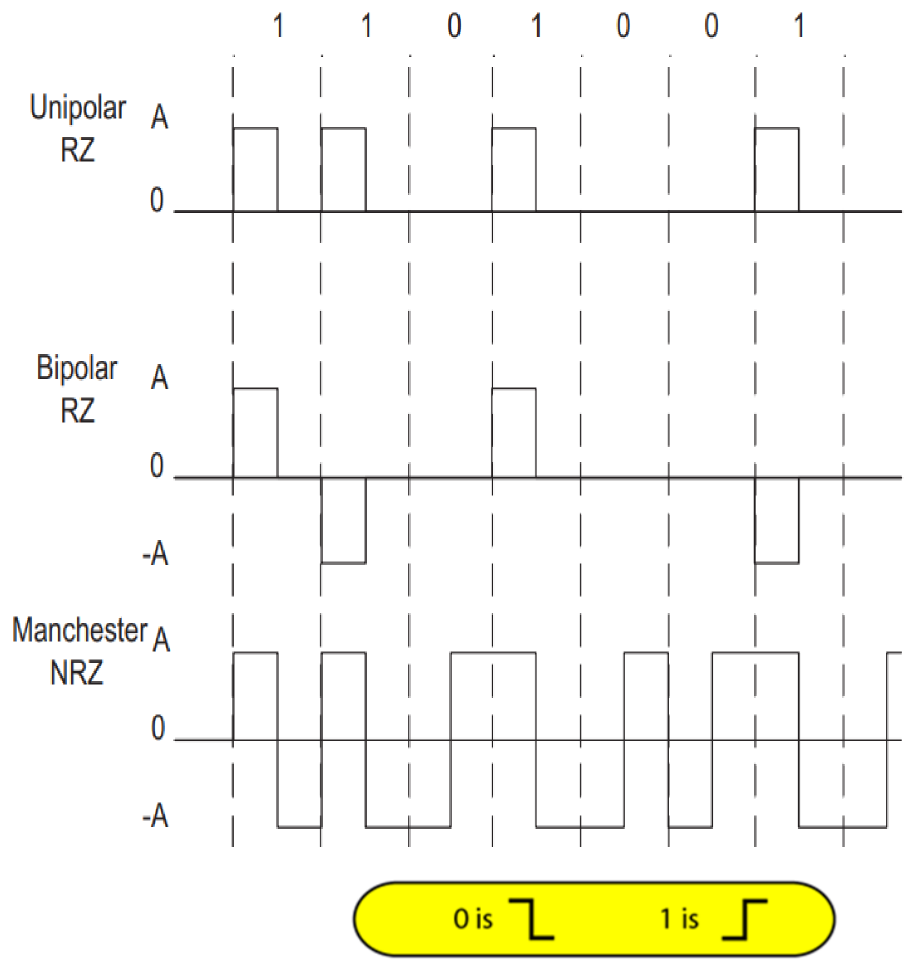
Note:

- 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.

- 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.

Cont.…

- *Bipolar (Pseudoternary) Signaling:* Binary 1's are represented by alternating +ve or - ve values. The binary 0 is represented by a zero level.
  - This is also called alternate mark inversion (AMI) signaling
- *Manchester Signaling:* Each binary 1 is represented by a positive half–bit period pulse followed by a negative half–bit period pulse. Similarly, a binary 0 is represented by a negative half–bit period pulse followed by a positive half–bit period pulse.
  - This type of signaling is also called split–phase encoding



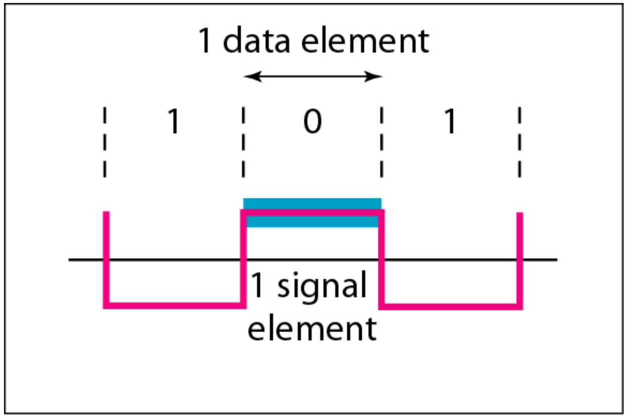
Mapping Data symbols onto Signal levels

- A data symbol (or element) can consist of a number of data bits:
  - 1 , 0 or
  - 11, 10, 01, ……
- A data symbol can be coded into a single signal element or multiple signal elements
  - 1 -> +V, 0 -> -V
  - 1 -> +V and -V, 0 -> -V and +V
- The ratio ‘r’ is the number of data elements carried by a signal element.

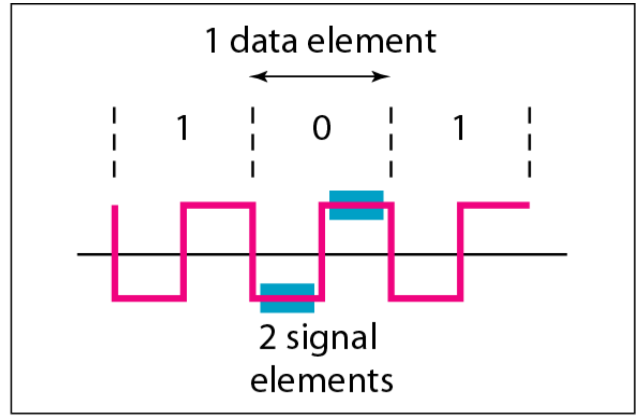
Relationship between data rate and signal rate

- The data rate defines the number of bits sent per sec - bps. It is often referred to the bit rate.
- The ***signal rate*** is the number of signal elements sent in a second and is measured in ***bauds***. It is also referred to as the modulation rate.
- Goal is to increase the data rate at the same time as reducing the baud rate.

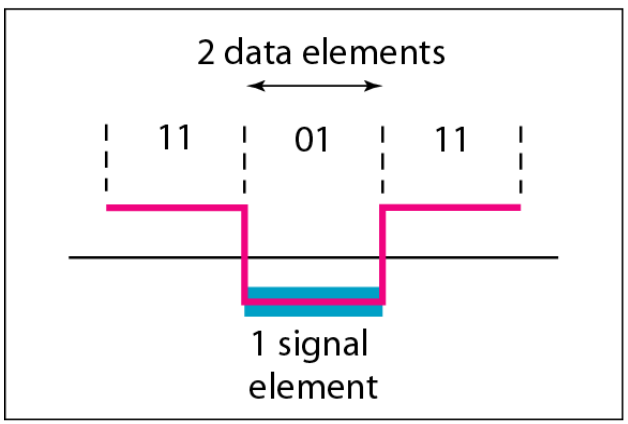
Signal element versus data element



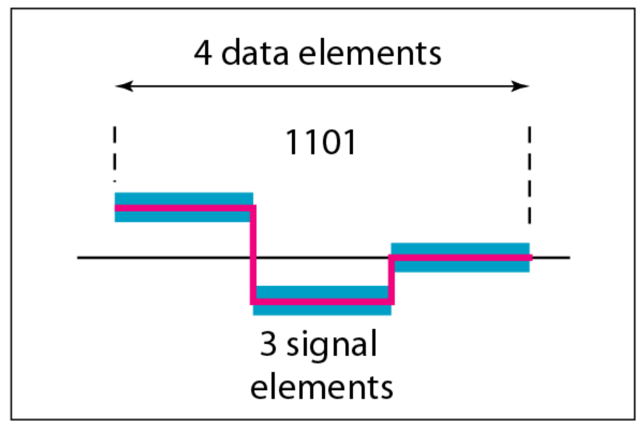
a. One data element per one signal element (r = 1)



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



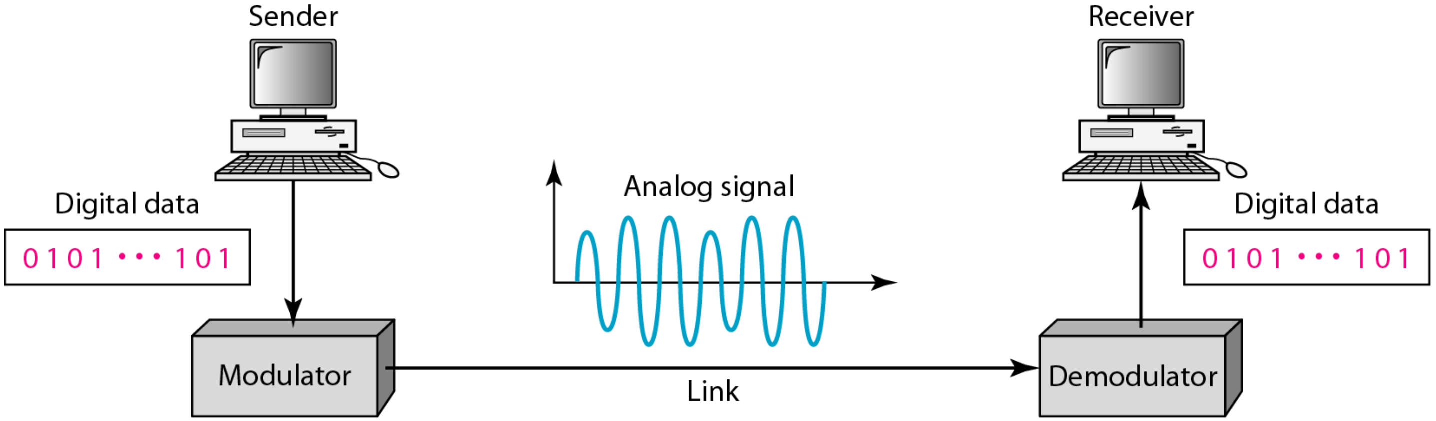
c. Two data elements per one signal element (r = 2)

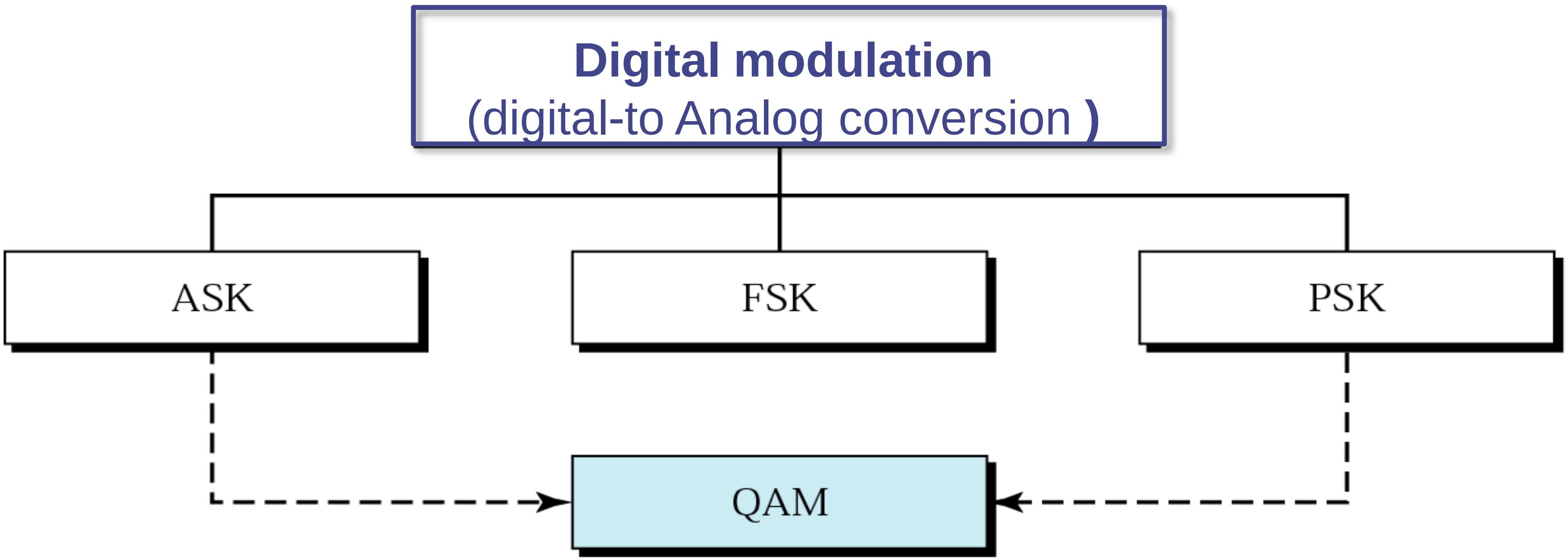


d. Four data elements per three signal elements $\left(r = \frac{4}{3}\right)$

Band pass Transmission of Digital Data

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

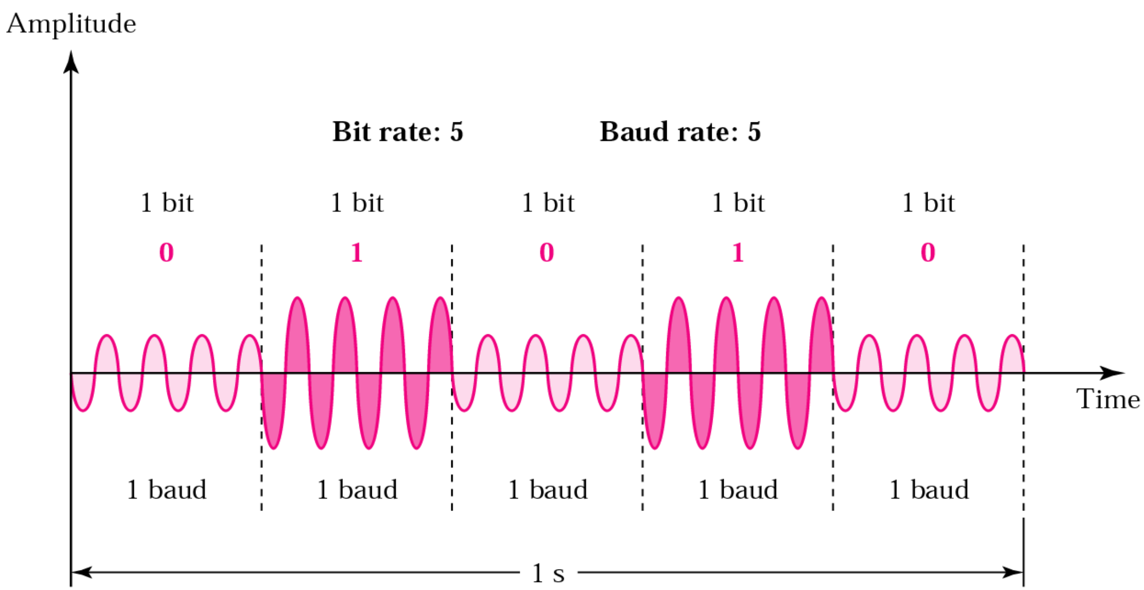



Digital modulation
(digital-to Analog conversion)
ASK
FSK
PSK
QAM

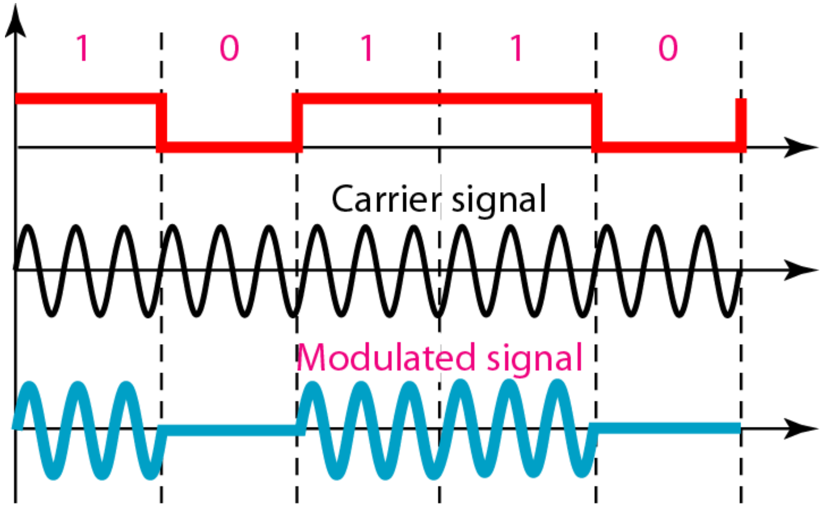
Amplitude Shift Keying (ASK)

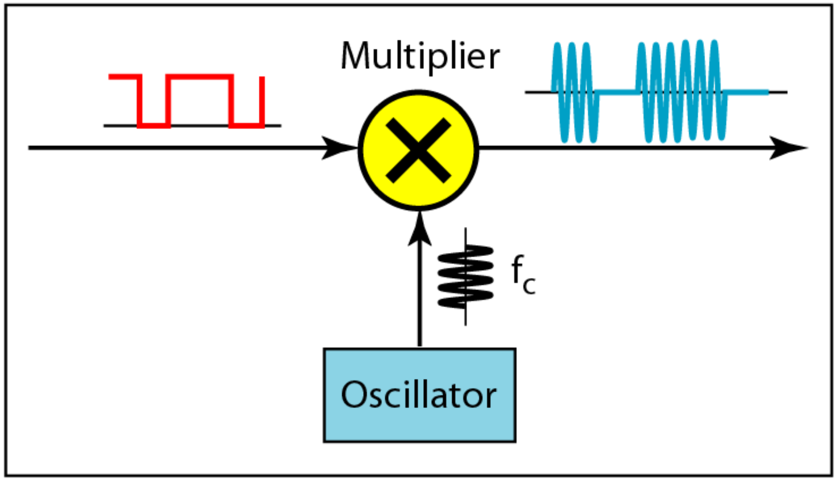
$$s(t) = \begin{cases} A\cos(2\Pi f_c t) & binary1 \\ 0 & binary0 \end{cases}$$

On/Off keying



- The strength/amplitude of the carrier signal is varied to represent binary 1 and 0.
- Frequency and phase remains the same.
- Highly susceptible to noise interference.
- Used up to 1200 bps on voice grade lines, and on optical fiber.

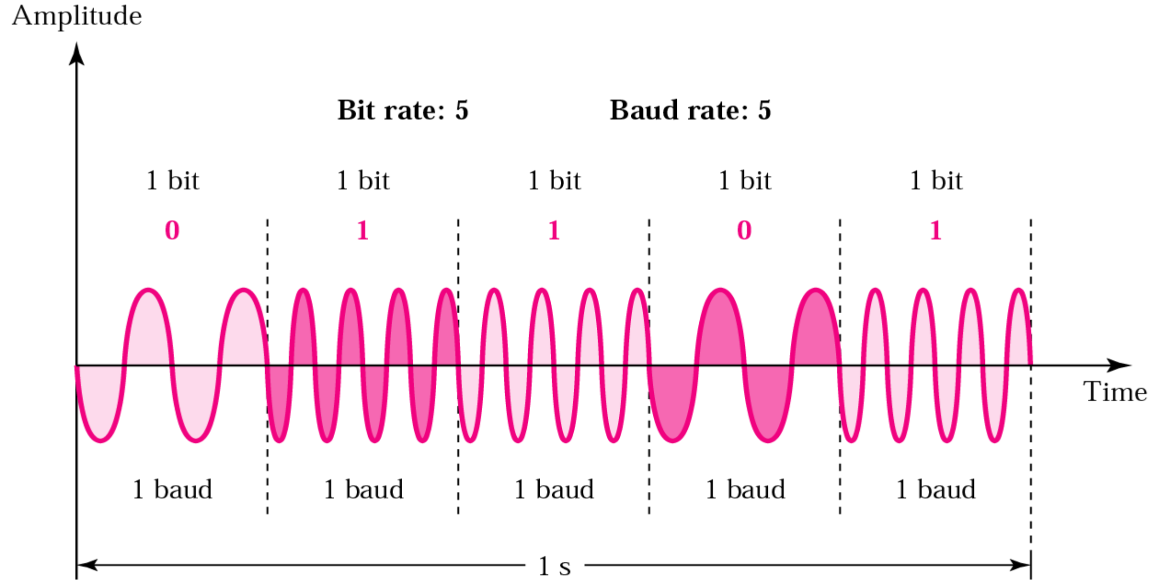

1
0
1
1
0
Carrier signal
Modulated signal


Multiplier
f_c
Oscillator

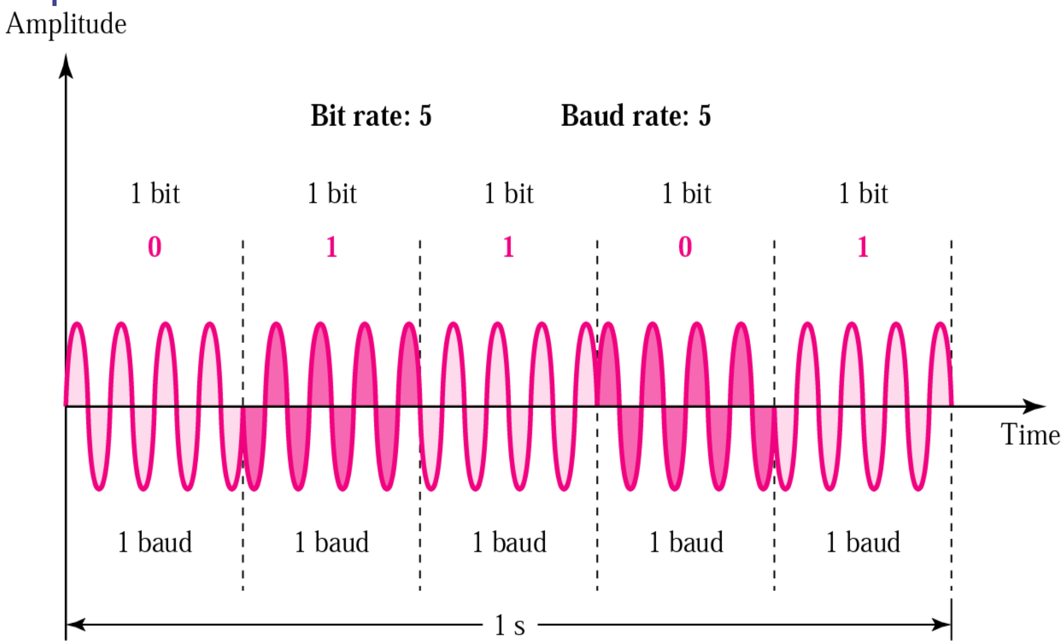
Frequency Shift Keying

- Frequency of the carrier is varied to represent digital data (binary 0/1)
- Peak amplitude and phase remain constant.
- Avoid noise interference by looking at frequencies (change of a signal) and ignoring amplitudes.
- Limitations of FSK is the physical capabilities of the carrier.
- f_1 and f_2 equally offset by equal opposite amounts to the carrier freq.

$$s(t) = \begin{cases} A\cos(2\Pi f_1 t) & binary 1 \\ A\cos(2\Pi f_2 t) & binary 0 \end{cases}$$



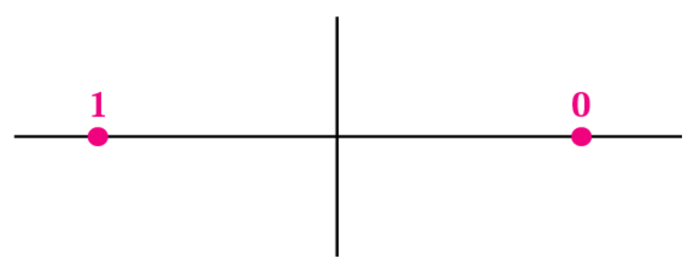
Phase Shift Keying



- Phase of the carrier is varied to represent digital data (binary 0 or 1)
- Amplitude and frequency remains constant.
- If phase 0 deg to represent 0, 180 deg to represent 1. (2-PSK)
- PSK is not susceptible to noise degradation that affects ASK or bandwidth limitations of FSK

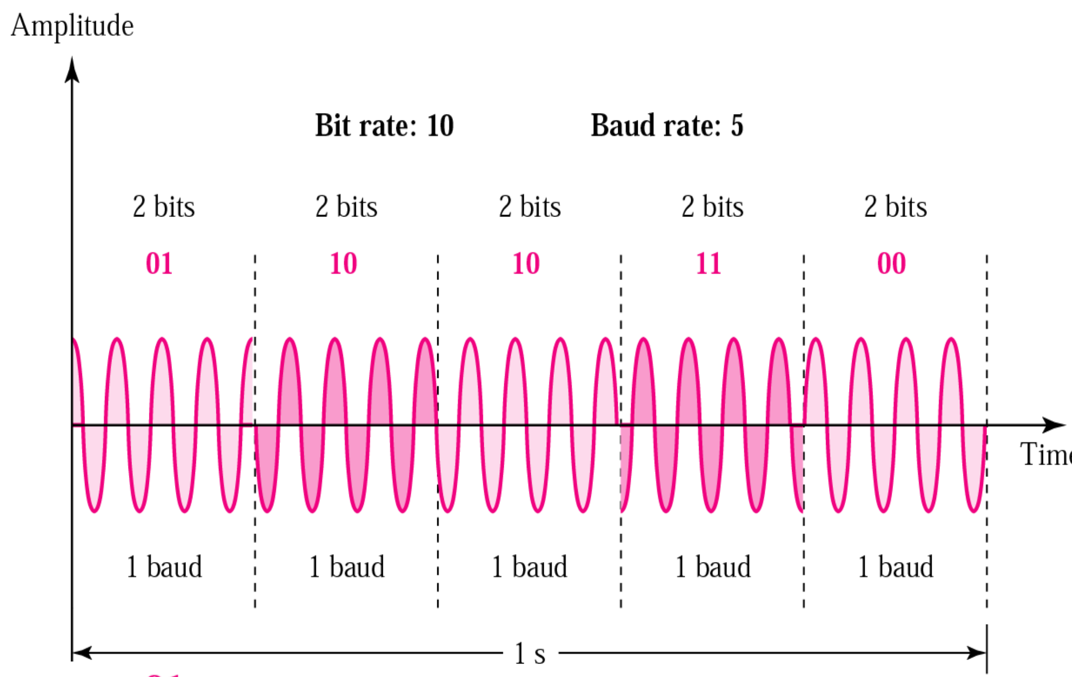
Bit	Phase
0	0
1	180

Bits



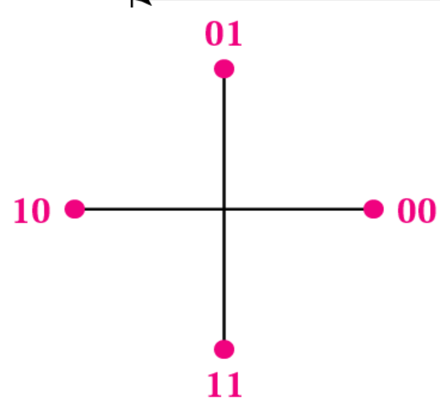
Constellation diagram

4-PSK (QPSK) method



Dibit	Phase
00	0
01	90
10	180
11	270

Dibit
(2 bits)



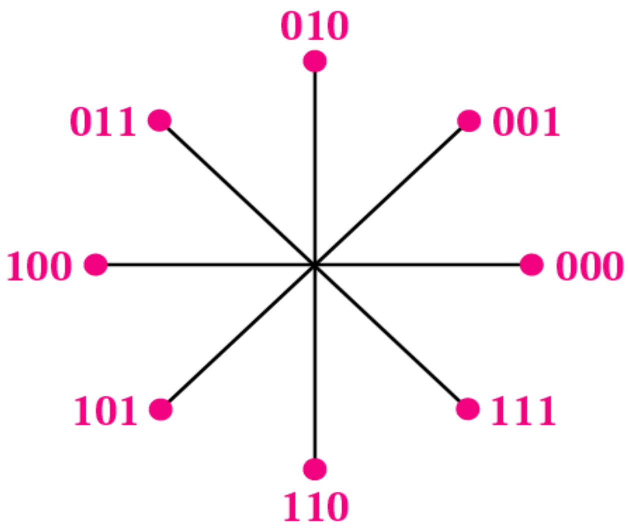
Constellation diagram

8-PSK

- We can extend, by varying the the signal by shifts of 45 deg (instead of 90 deg in 4-PSK)
- With $8 = 2^3$ different phases, each phase can represents 3 bits (tribit).

Tribit	Phase
000	0
001	45
010	90
011	135
100	180
101	225
110	270
111	315

Tribits
(3 bits)

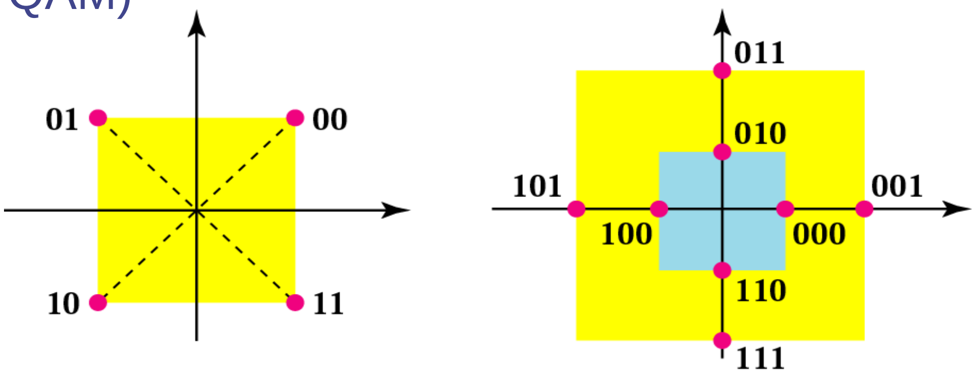


Constellation diagram

Quadrature Amplitude Modulation

- PSK is limited by the ability of the equipment to distinguish between small differences in phases.
  - Limits the potential data rate.
- Quadrature amplitude modulation is a combination of ASK and PSK so that a maximum contrast between each signal unit (bit, dibit, tribit, and so on) is achieved.
  - We can have x variations in phase and y variations of amplitude
  - x • y possible variation (greater data rates)
- Numerous variations. (4-QAM, 8-QAM)

# of phase shifts > # of amplitude shifts



4-QAM
1 amplitude, 4 phases

8-QAM
2 amplitudes, 4 phases