SIGNALS

Subject- Computer Network

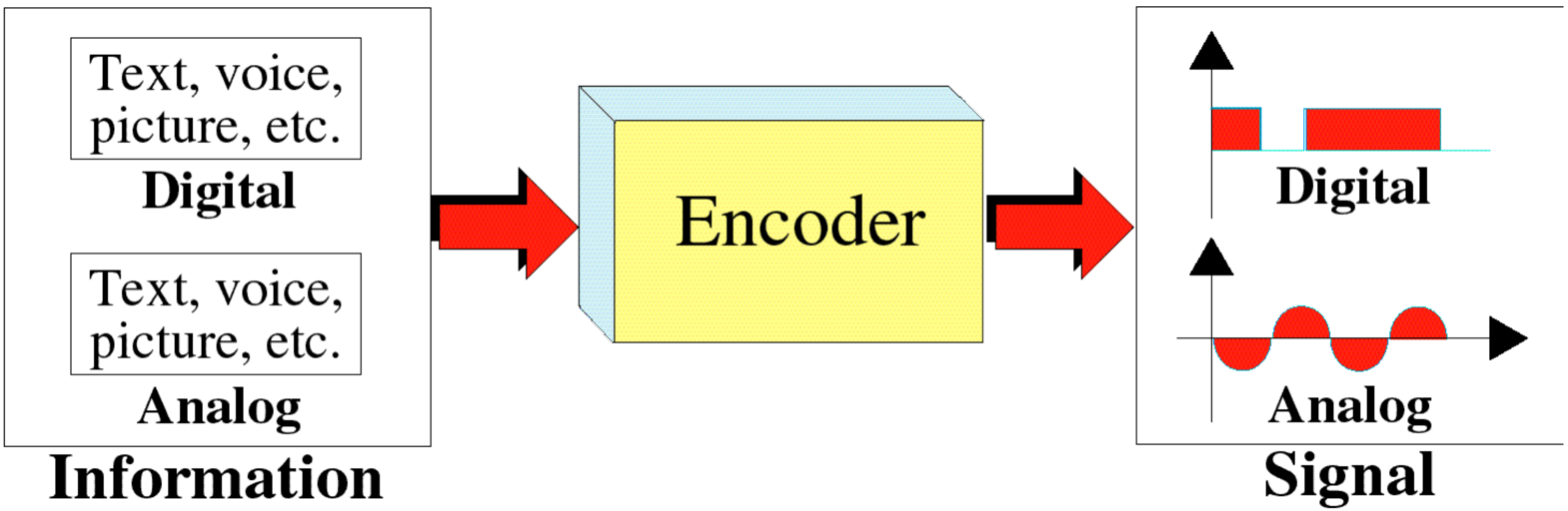
Slides compiled by- Sanghamitra De

Analog & Digital Data

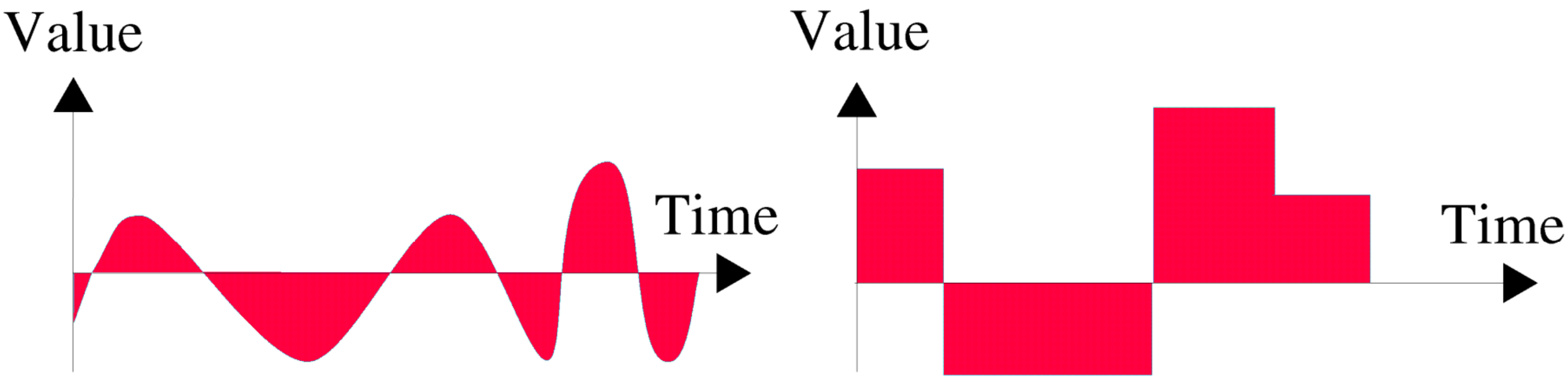
- Analog Data
  - Takes Continuous values
  - e.g. Human voice

- Digital Data
  - Takes Discrete values
  - e.g. data stored in computer memory

Transforming Analog Data to Digital Data



Analog & Digital Signal



a. Analog signal

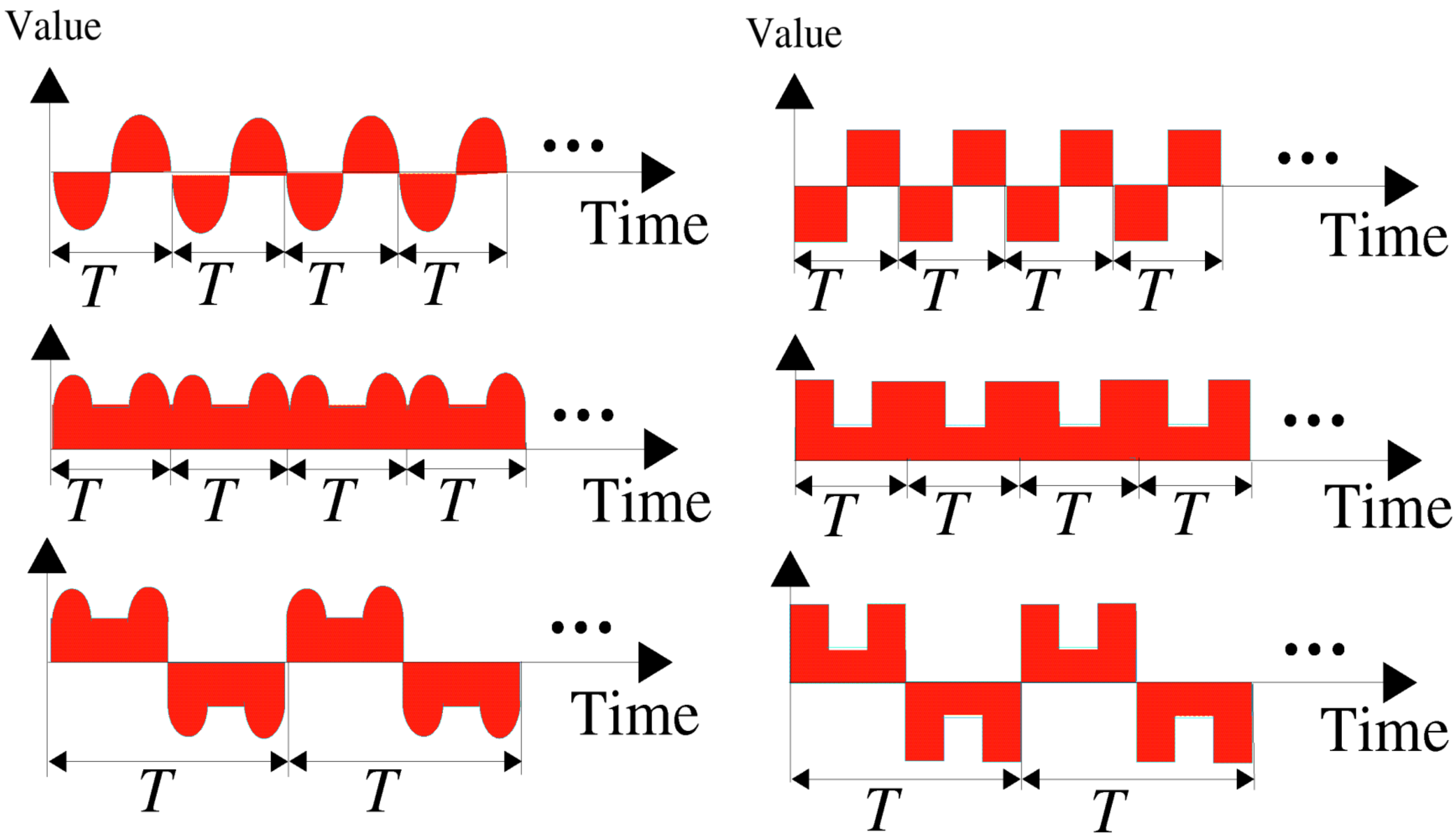
b. Digital signal

Periodic & Aperiodic Signals

- Periodic Signal
  - Follows pattern (Cycle)
  - Repetitive pattern repeated over subsequent identical time periods (Time Period)

- Aperiodic
  - No pattern
  - No repetition

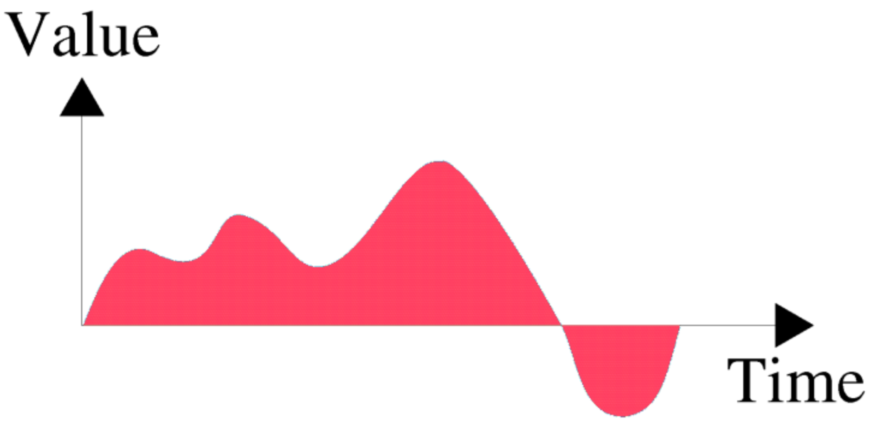
Periodic & Aperiodic Signals



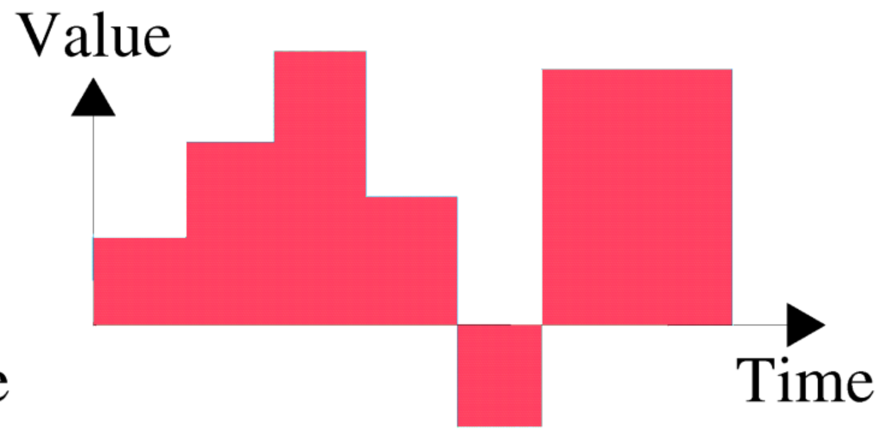
a. Analog

b. Digital

Periodic & Aperiodic Signals

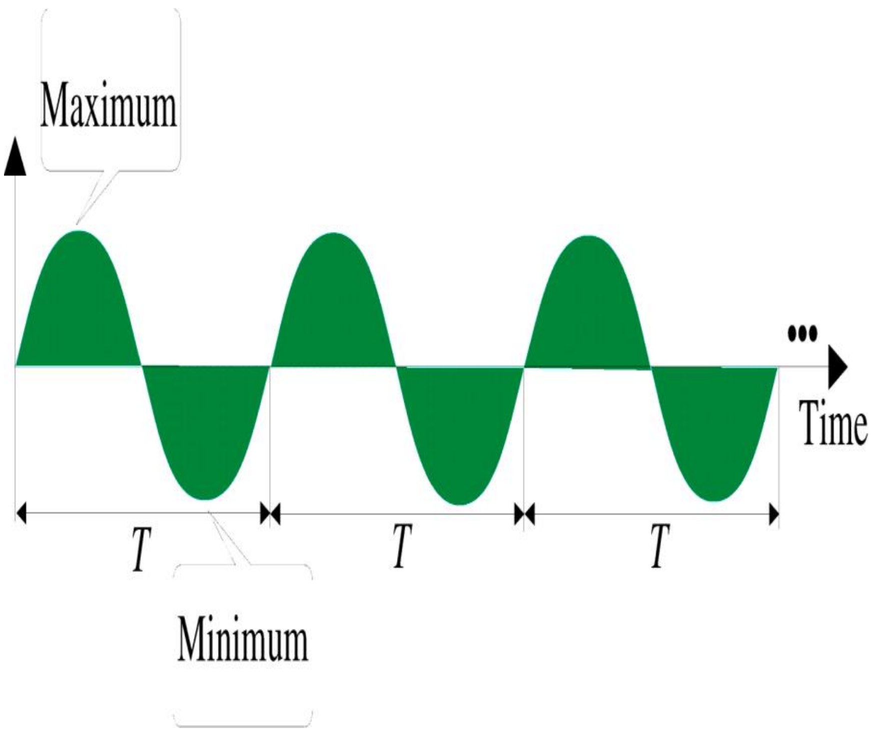


a. Analog



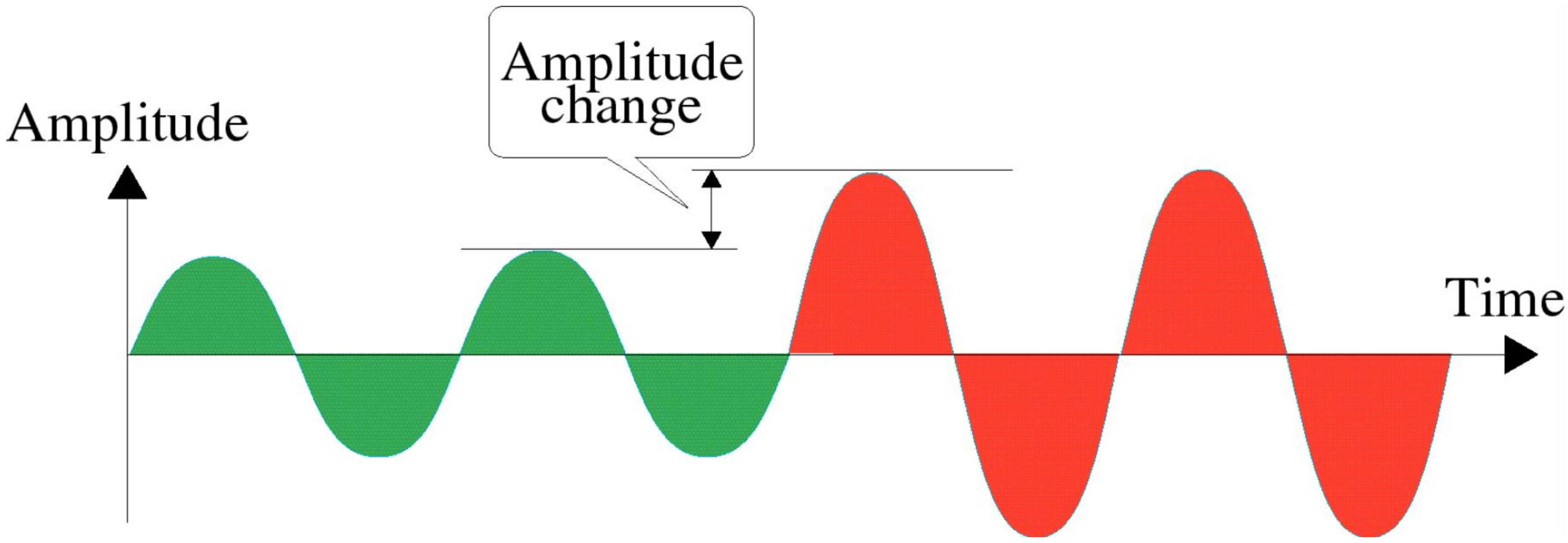
b. Digital

Analog Signals

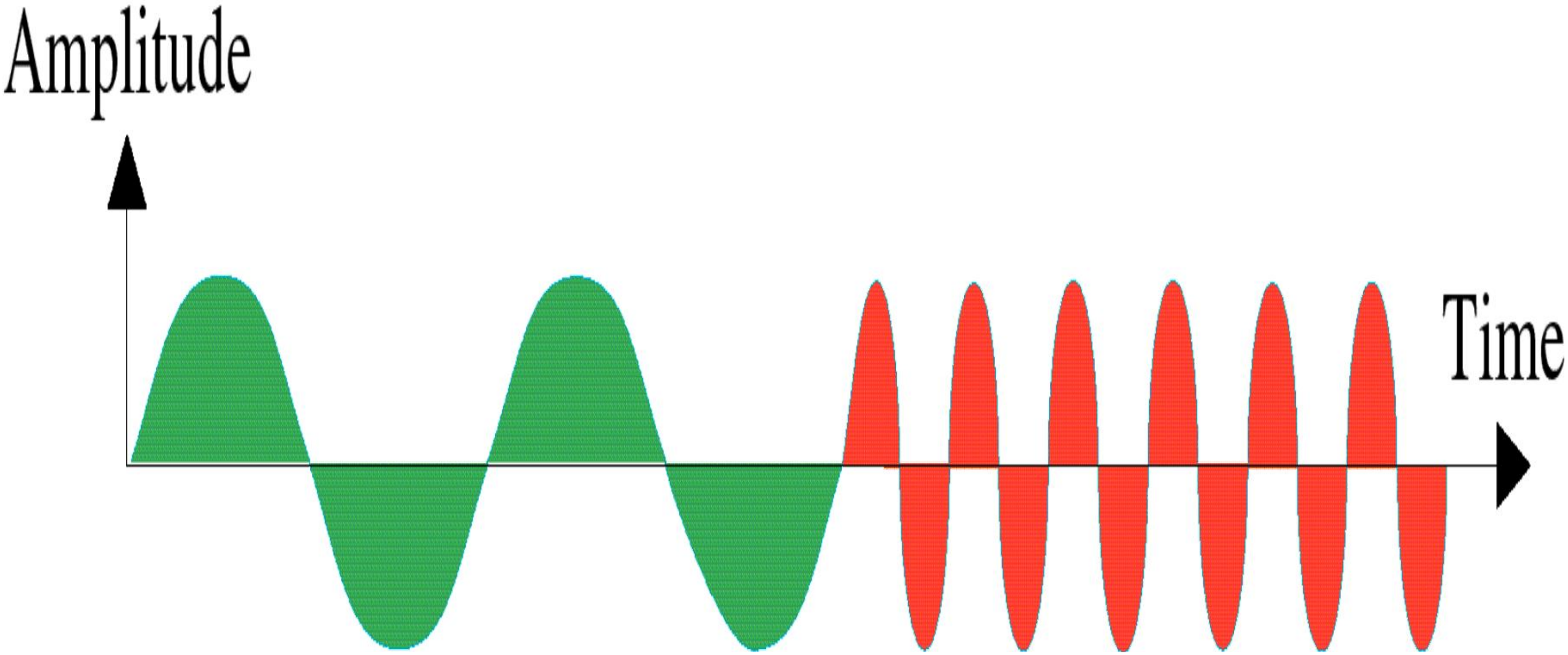


- Amplitude
- Period/Time Period
- Frequency (cycles per sec)
- Phase

Amplitude Change



Frequency Change



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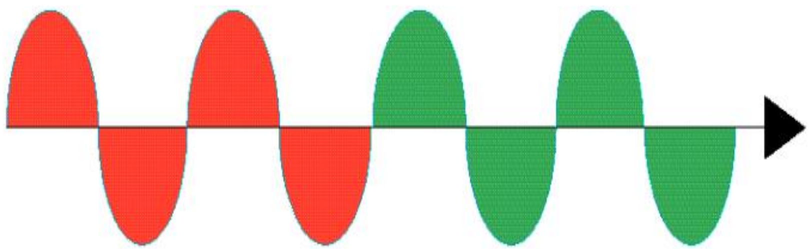
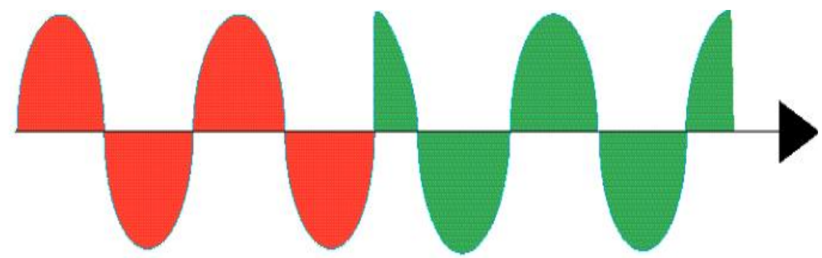
- High frequency: signal value changes over short time span
- Low frequency: signal value changes over long time span
- Zero frequency: no change in signal. Constant voltage level
- Infinite frequency: Signal changes instantaneously.

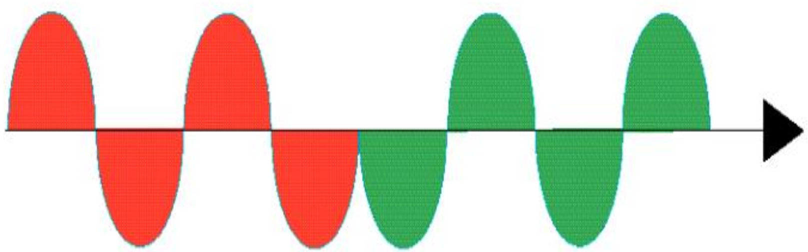
Phase Change



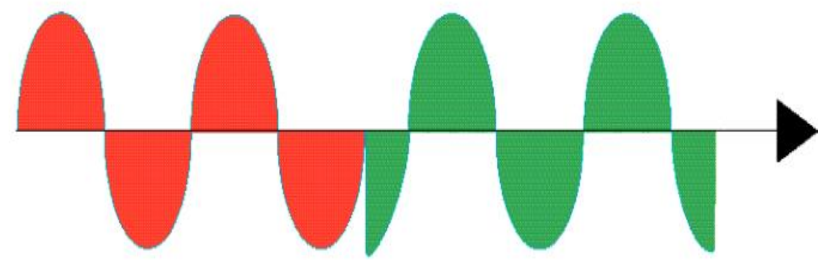
a. No phase change



b. 90 degree phase change

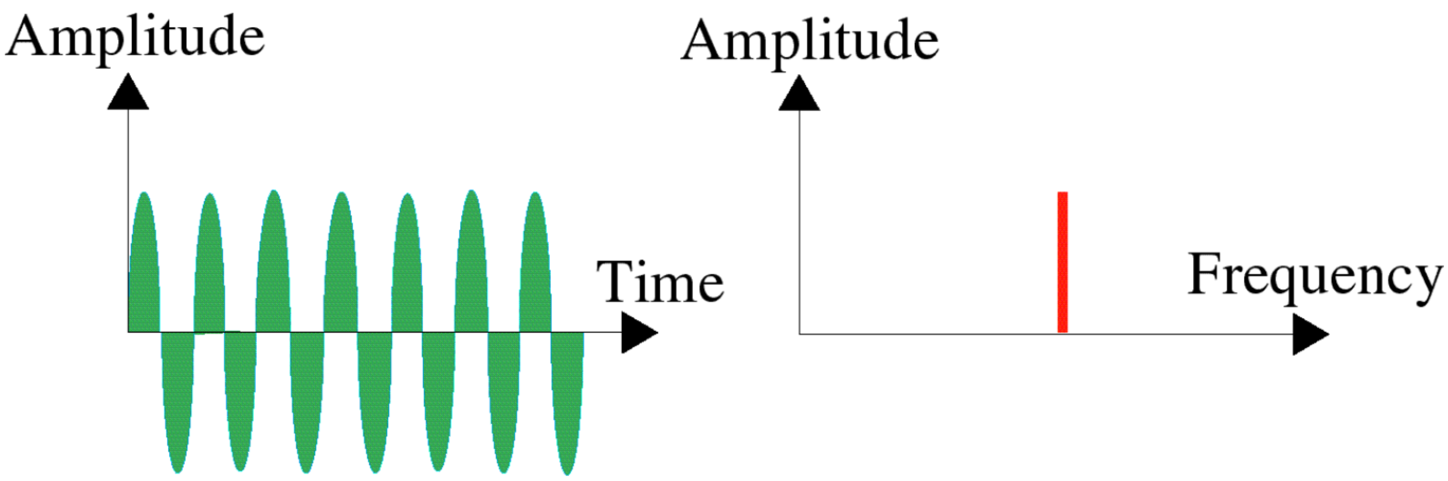


c. 180 degree phase change



d. 270 degree phase change

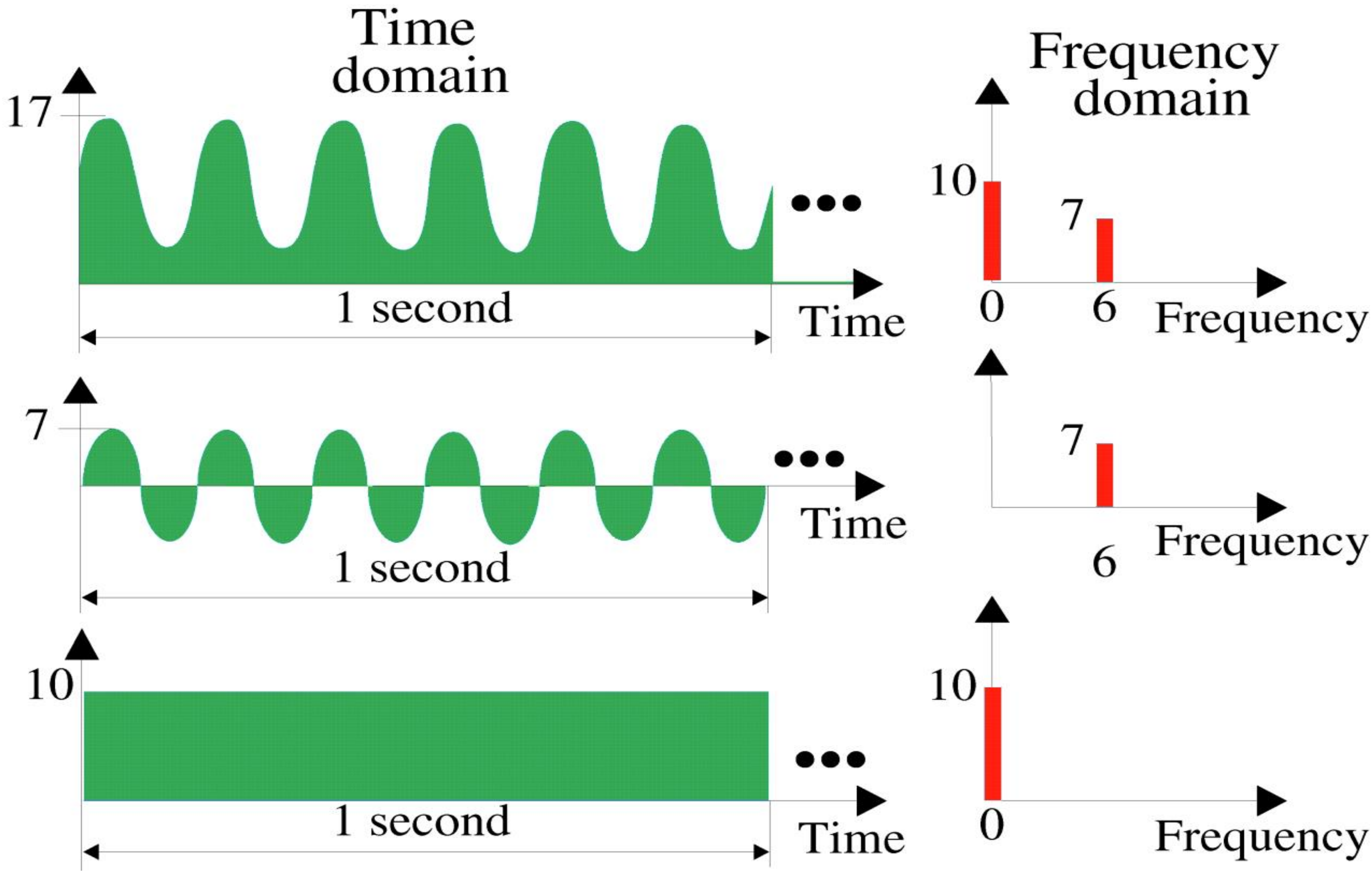
Time & Frequency domain



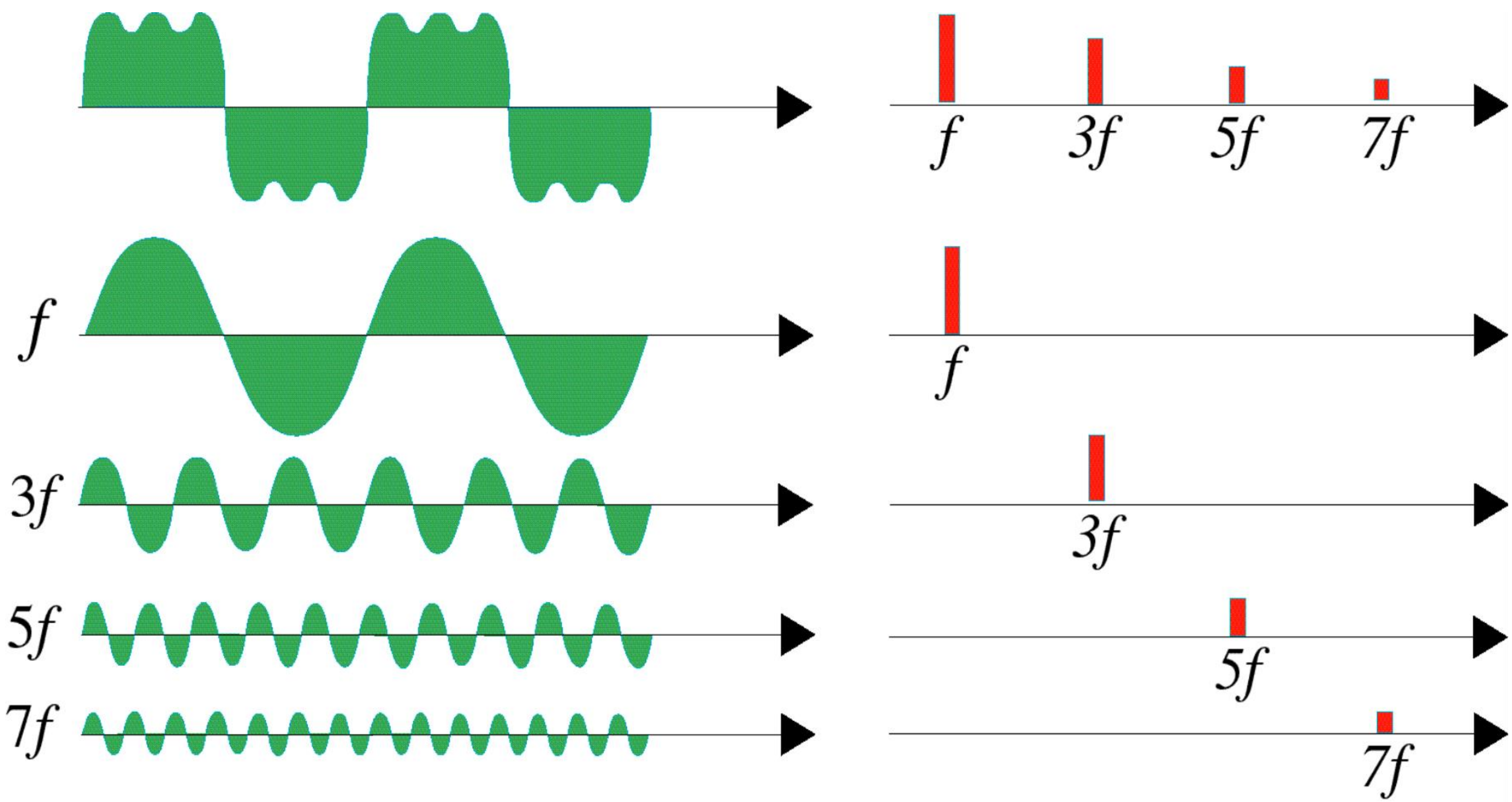
a. Time domain b. Frequency domain

- Time domain plot: changes in signal amplitude with time
- Frequency domain plot: how much of the signal lies within each given frequency band over a range of frequencies

Signal with DC component



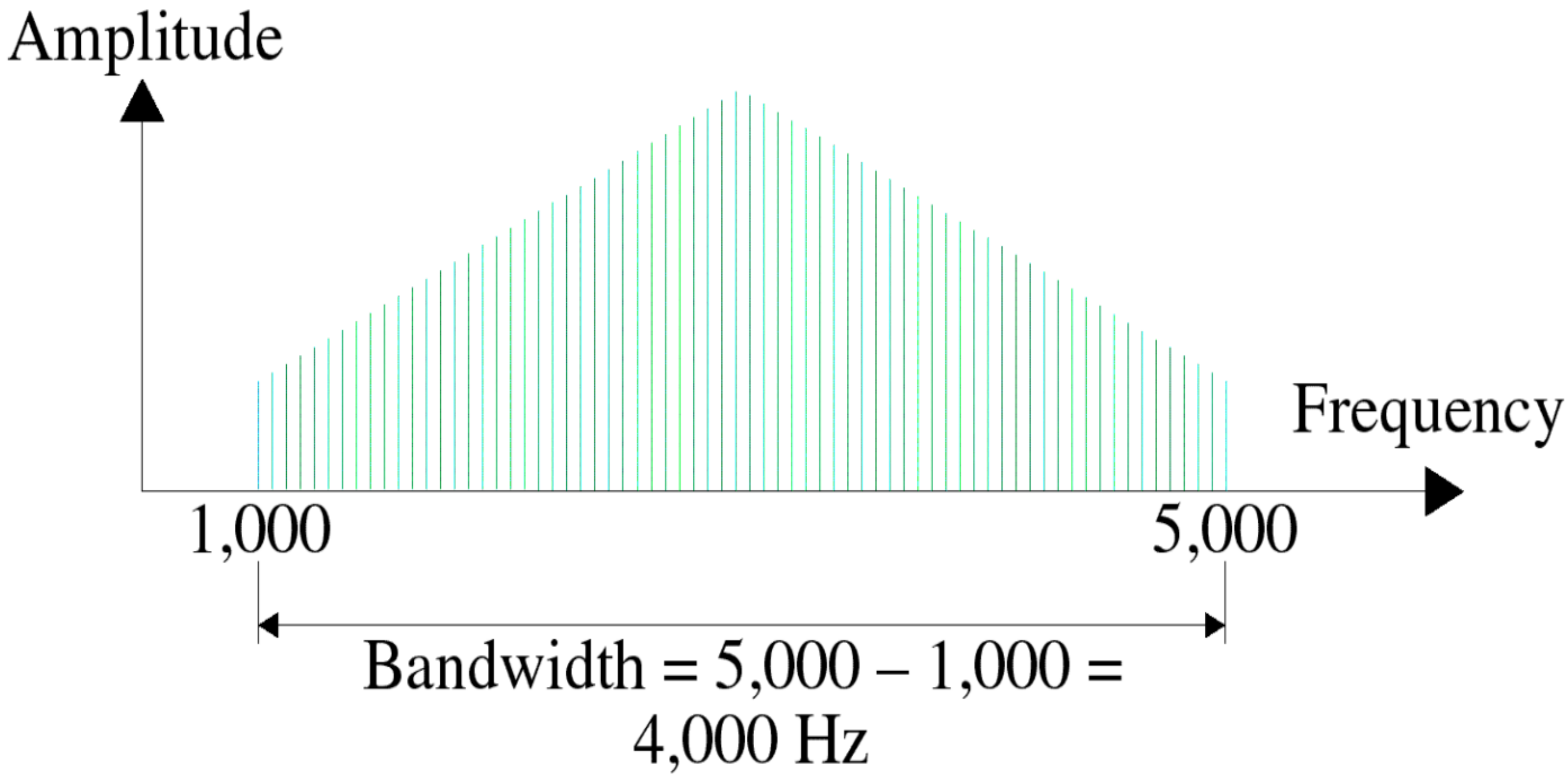
Composite signal



a. Time domain

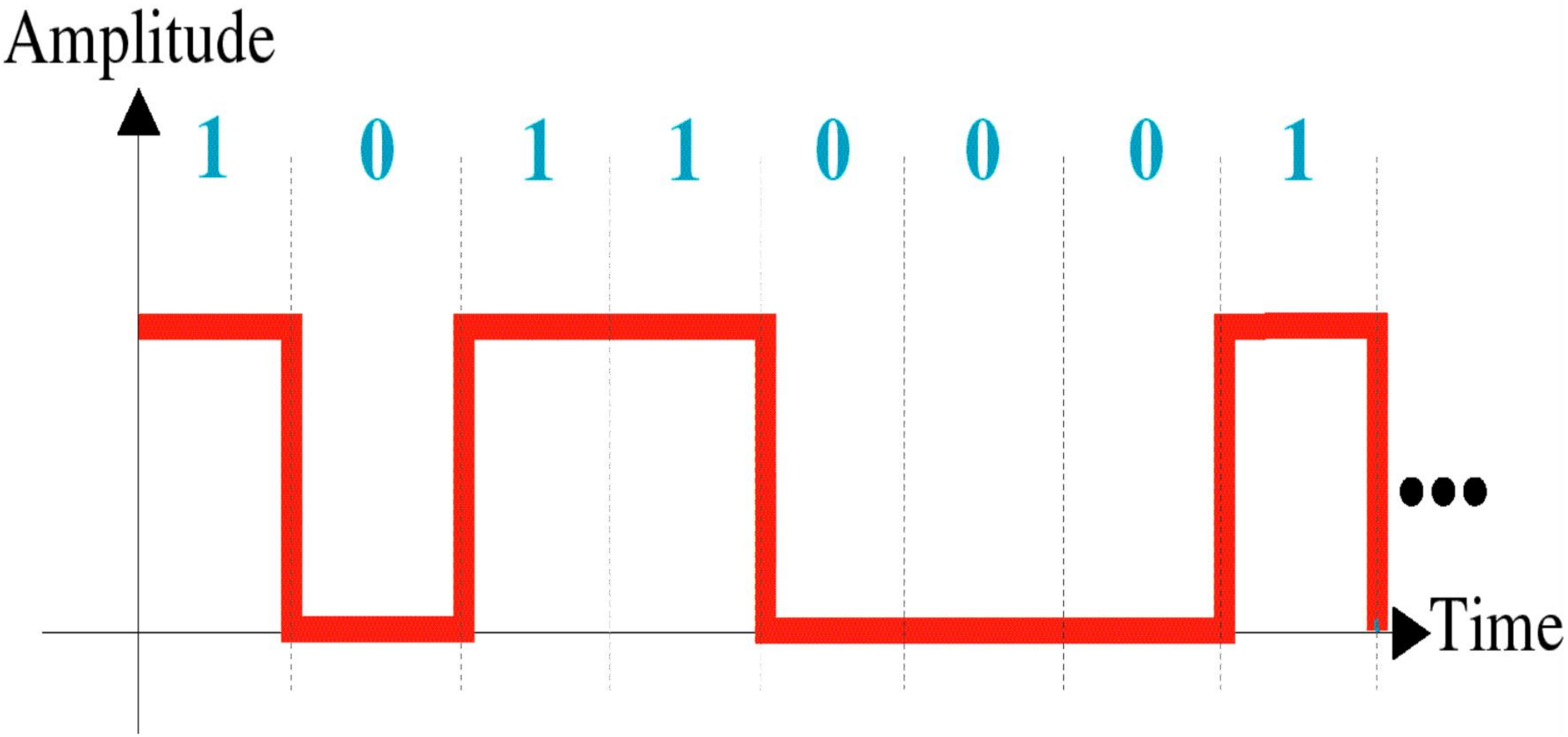
b. Frequency domain

Frequency Spectrum & Bandwidth



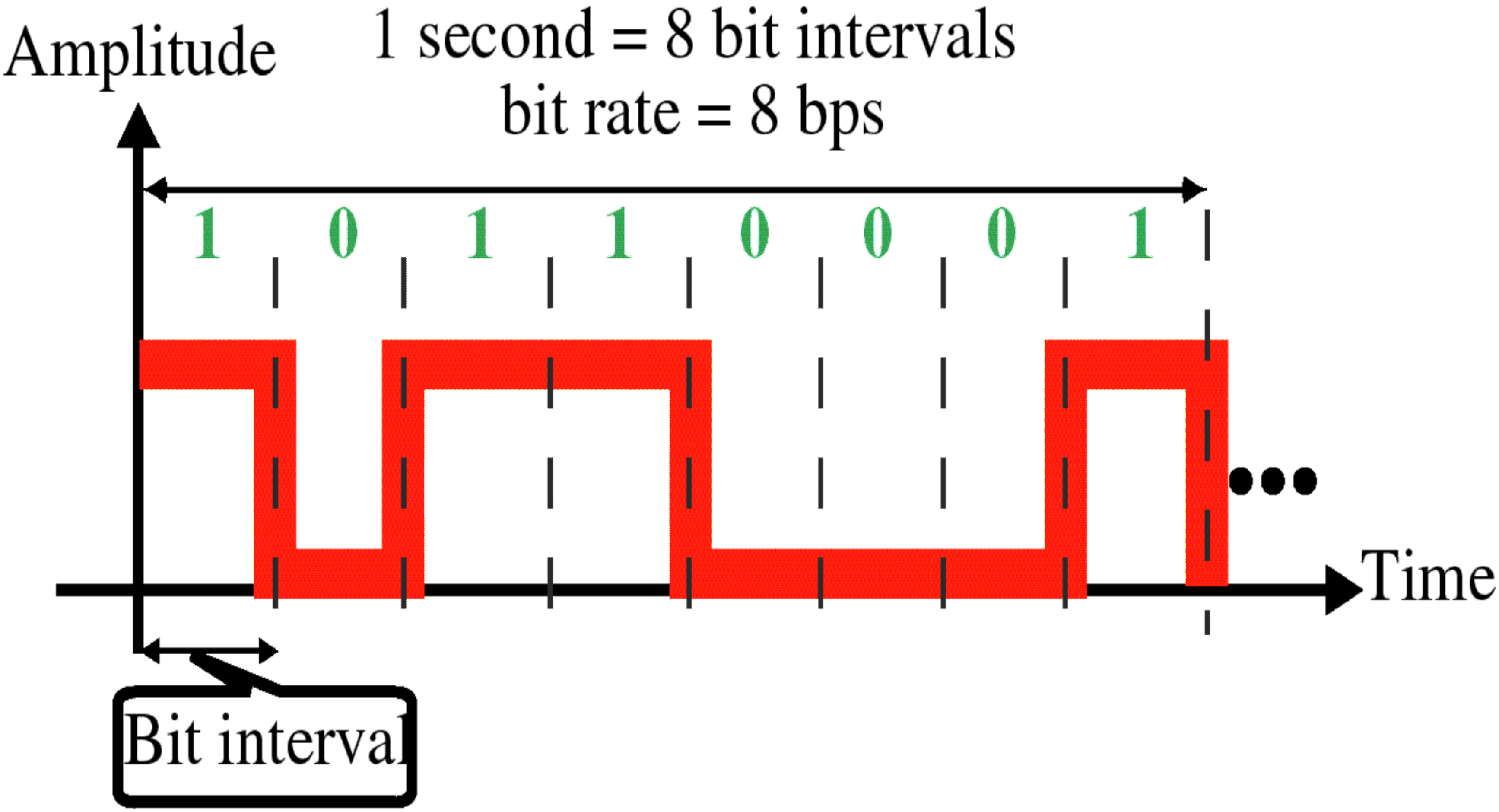
- Frequency spectrum
- Bandwidth of a signal of a signal

Digital Signal

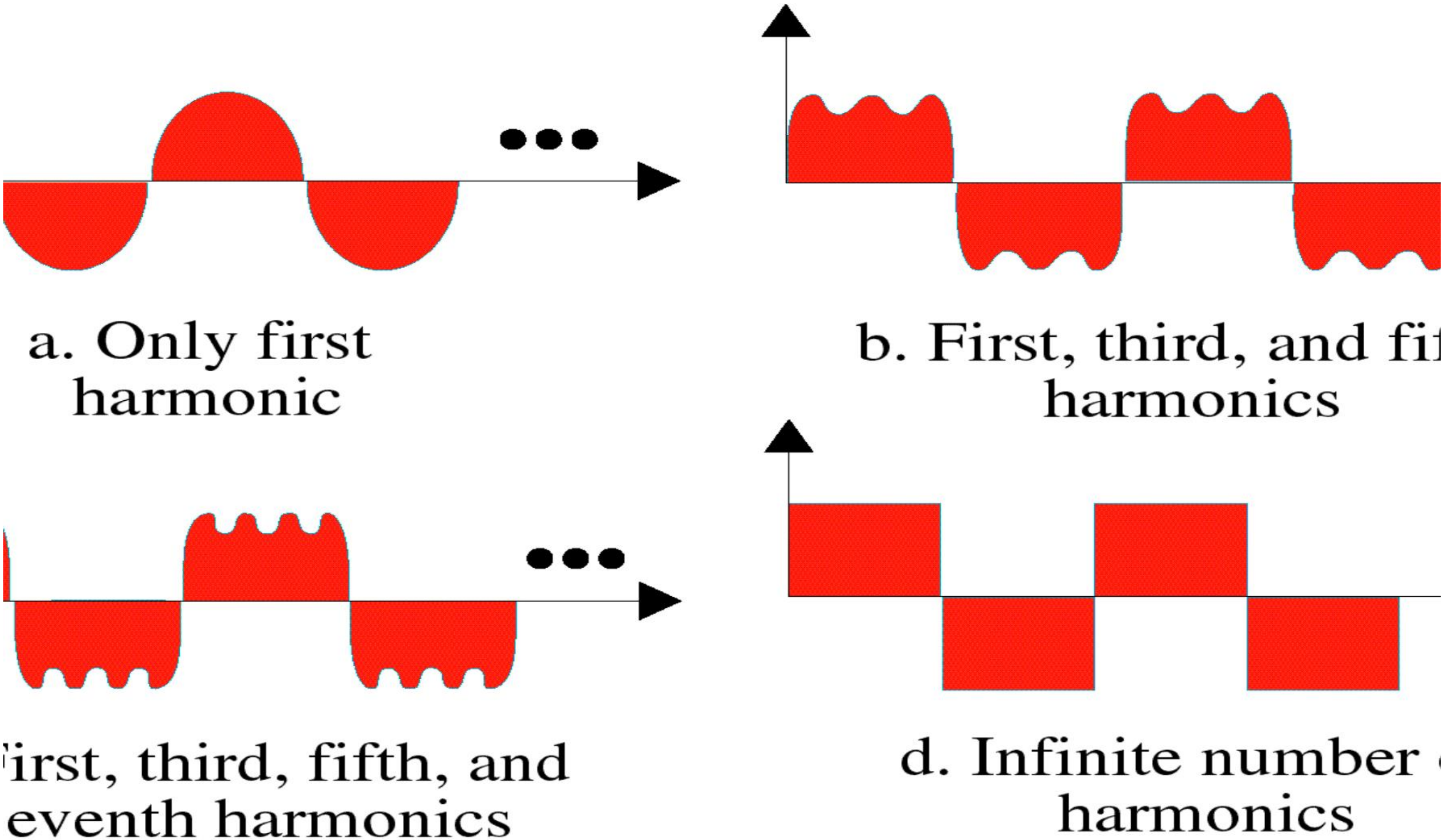


- Most Digital signals are **aperiodic**
- **Bit interval**
- **Bit rate (bps)**

Bit interval & Bit rate

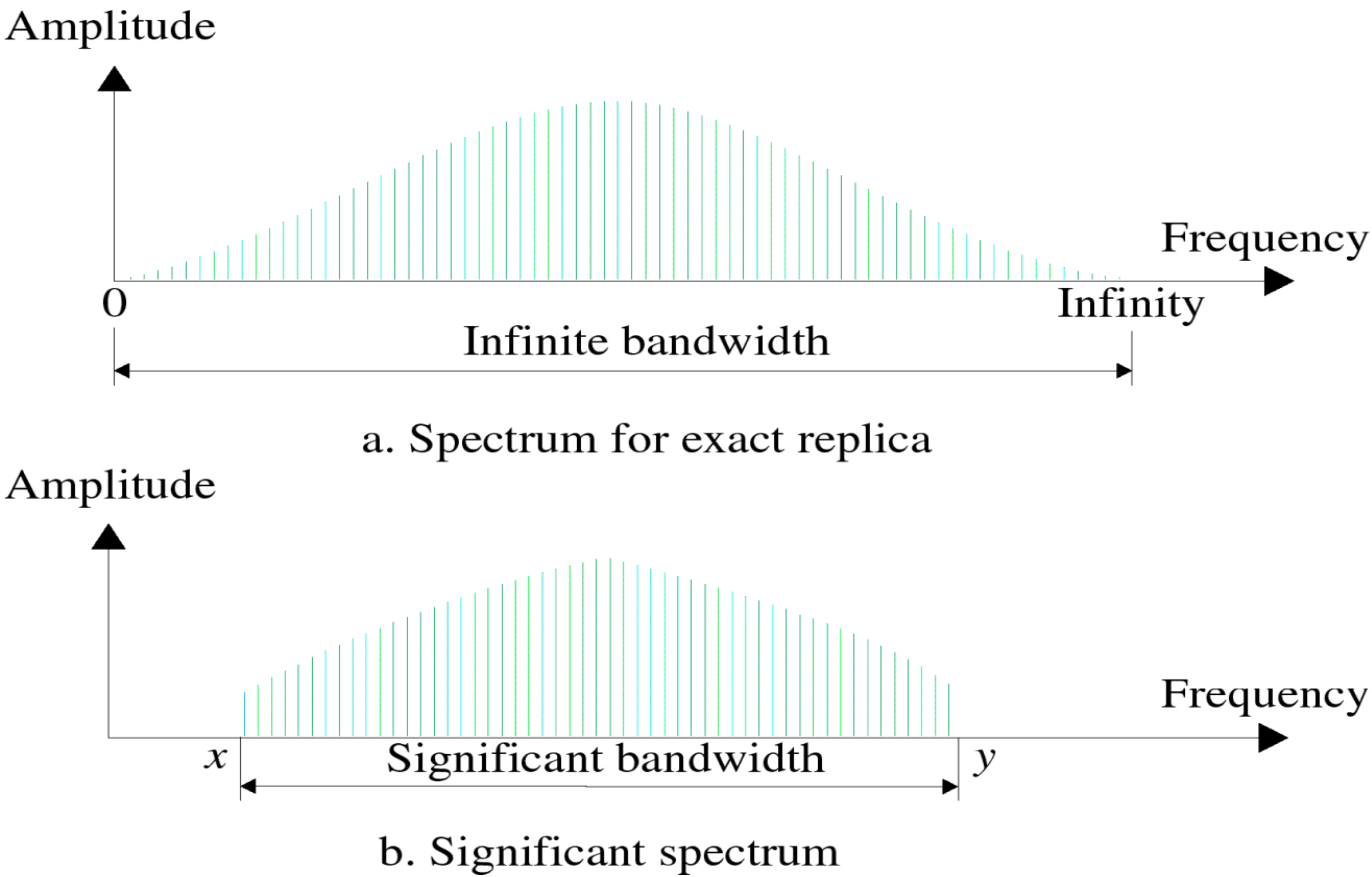


Decomposition of a Digital Signal



Harmonics of a digital signal

Frequency Spectrum of Digital Signal



Encoding & Modulation

Introduction

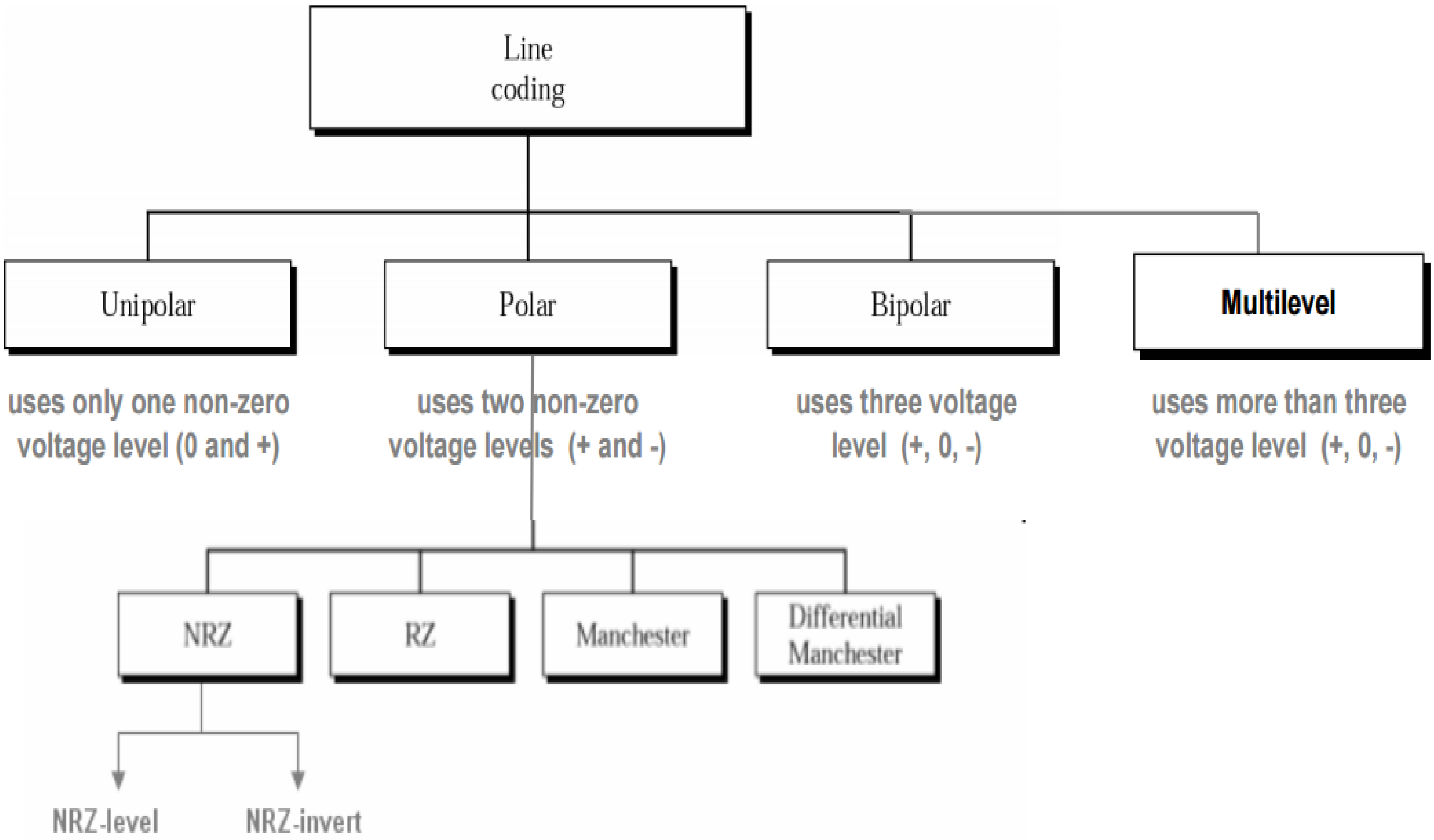
- Data stored in Computer: 0s & 1s
- To be carried from one place to another: data to be converted to digital signals
- Conversion methods:
    - ➢Digital to Digital
    - ➢Digital to Analog
    - ➢Analog to Digital
    - ➢Analog to Analog

Digital to Digital Encoding

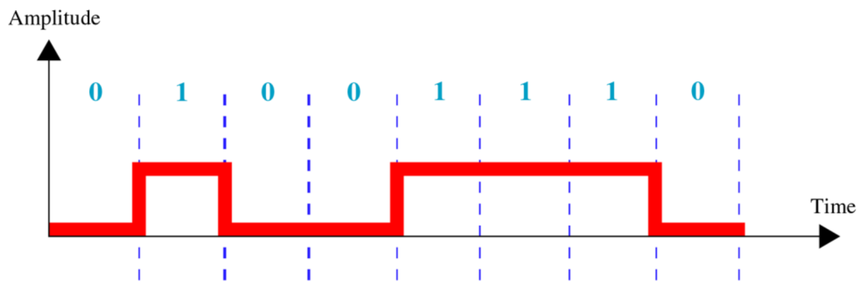
Coding Scheme-types

- Line Coding: process of converting binary data (sequence of bits) to a digital signal
  - digital signal depends 'linearly' on information bits – bits are transmitted 'one-by-one'
- Block Coding: unlike line codes which operate on a stream of information bits, block codes operate on block of information bits
  - redundant bit(s) are added to each block of information bits to ensure synchronization and error detection

Line Coding Schemes – can be divided into four broad categories

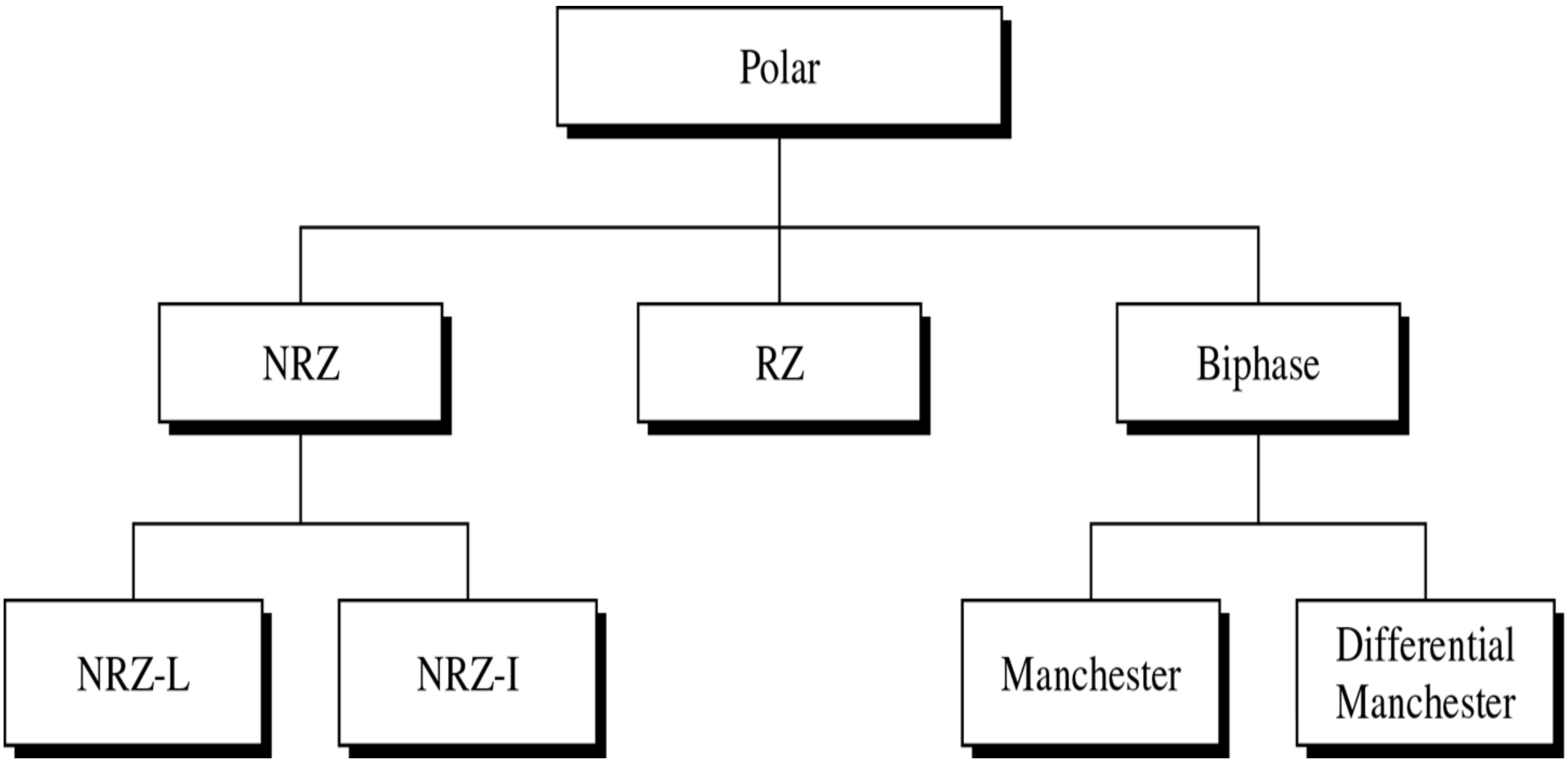


Unipolar Encoding

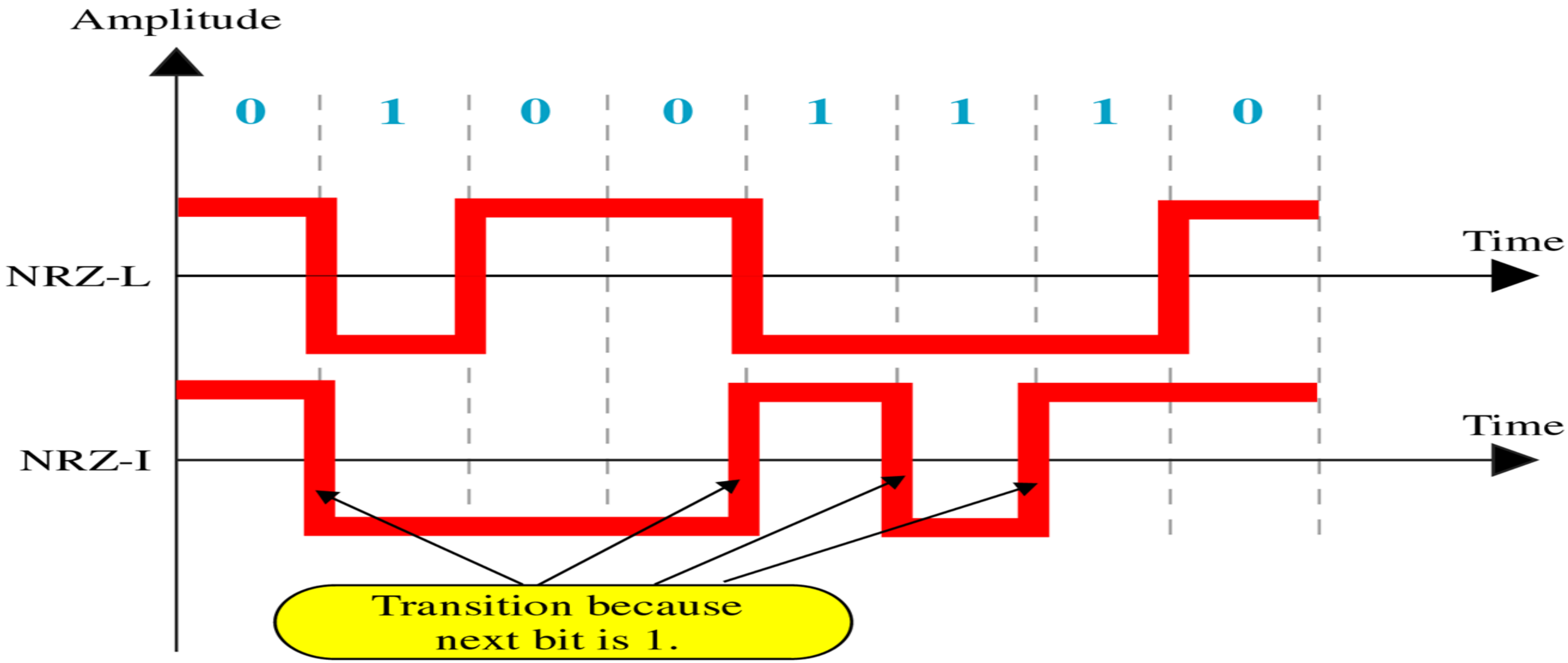


- Issues:
  - Resultant DC component, Transformer
  - No Synchronization

Types of Polar Encoding

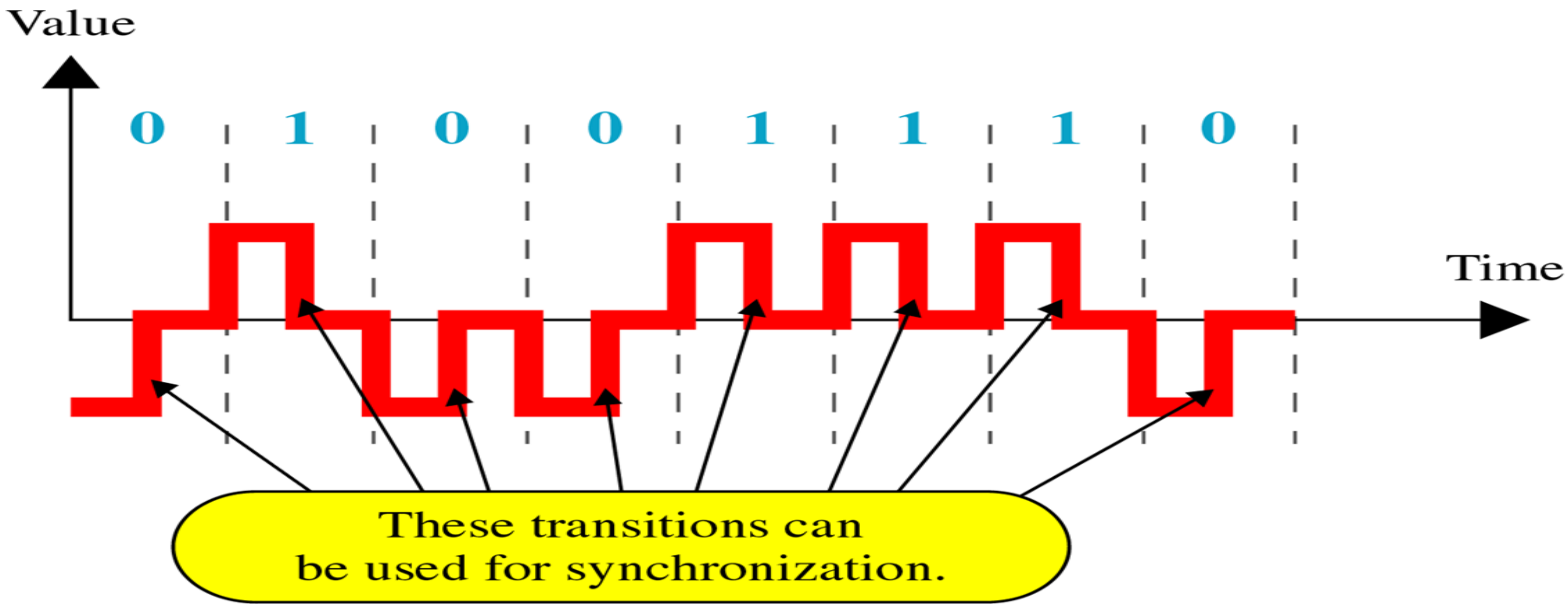


NRZ-L and NRZ-I Encoding



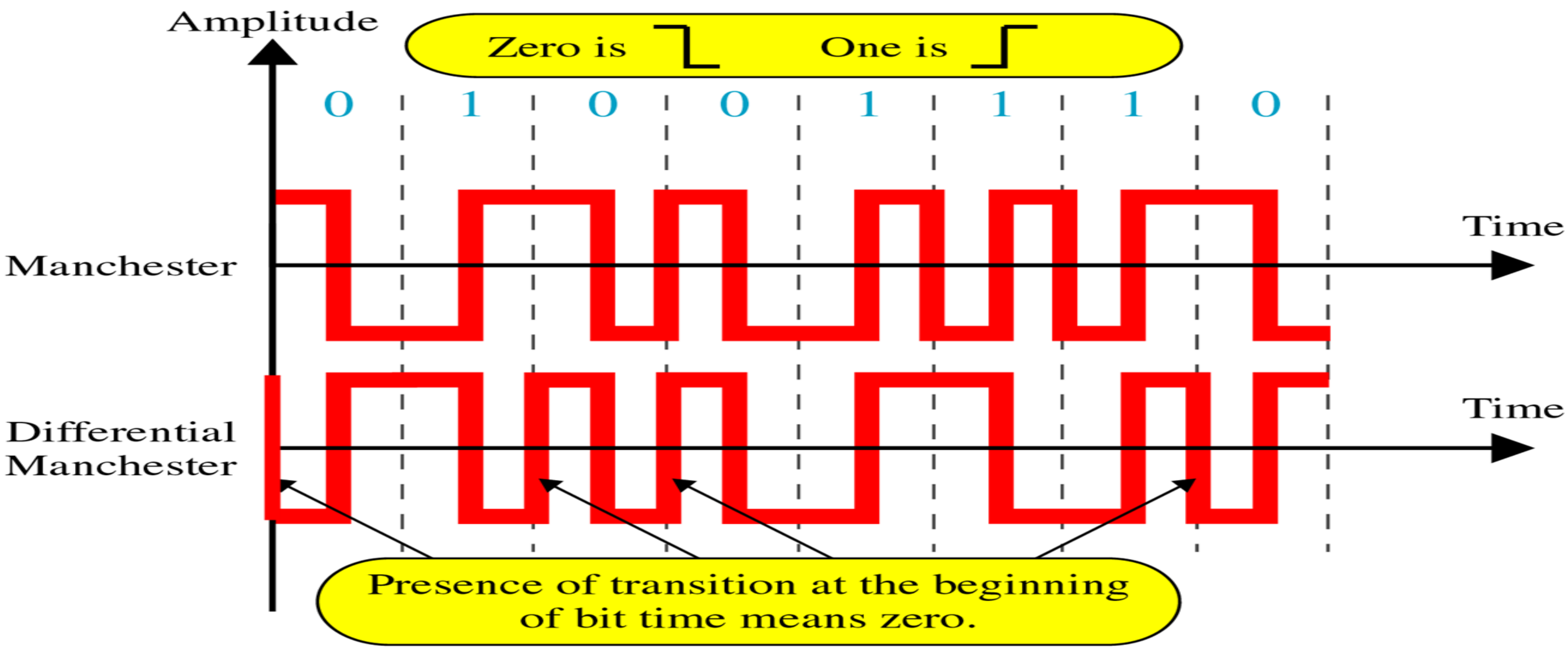
- NRZ-L the level of the voltage determines the value of the bit. In NRZ-I, the inversion or the lack of inversion determines the value of the bit.
- If there is no change, the bit is 0; if there is a change, the bit is 1.

RZ Encoding



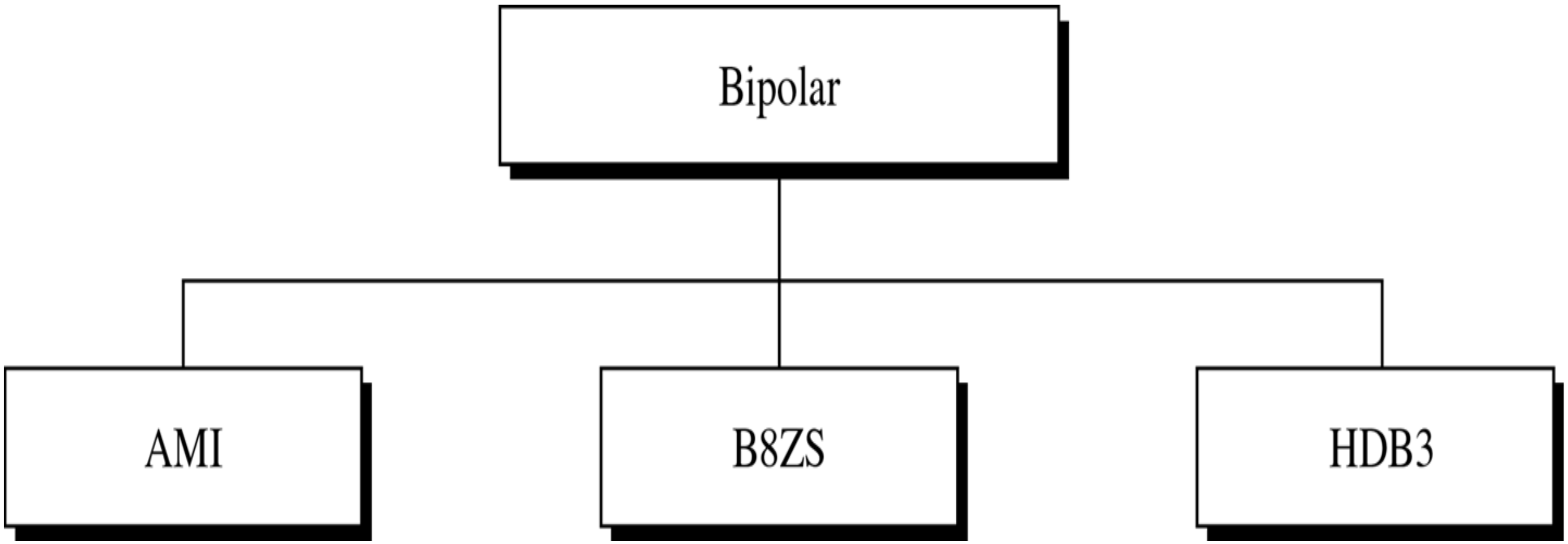
- **Problem with NRZ encoding:** when the sender and receiver clocks are not synchronized. The receiver does not know when one bit has ended and the next bit is starting.
- RZ uses three values: positive, negative, and zero
- Signal changes not between bits but during the bit.
- Signal goes to 0 in the middle of each bit. It remains there until the beginning of the next bit.

Manchester and Differential Manchester Encoding



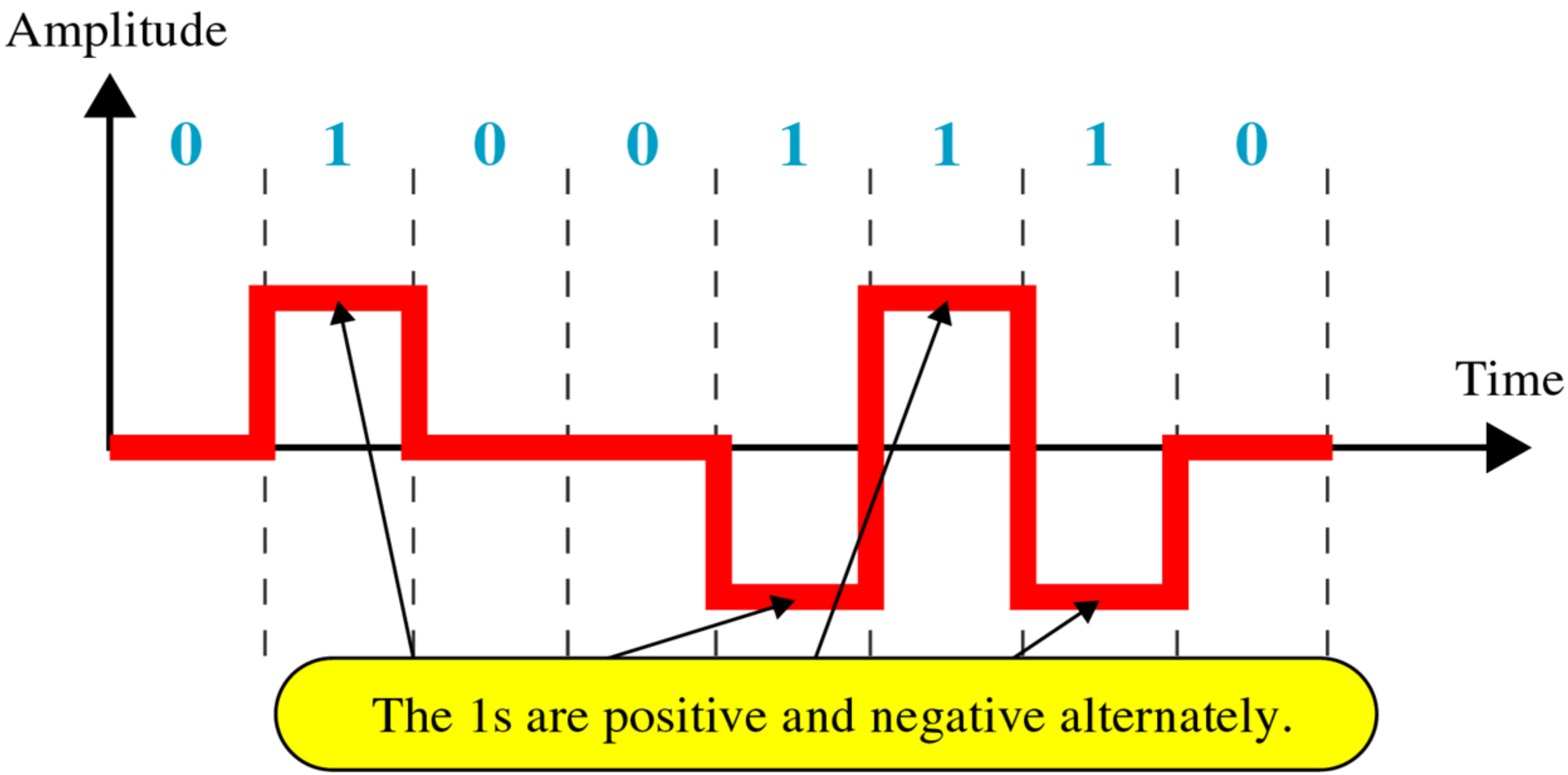
- Manchester encoding: duration of the bit is divided into two halves. The voltage remains at one level during the first half and moves to the other level in the second half.
- The transition at the middle of the bit provides synchronization.
- Differential Manchester: There is always a transition at the middle of the bit, but the bit values are determined at the beginning of the bit.
- If the next bit is 0, there is a transition; if the next bit is 1, there is none.

Types of Bipolar Encoding



- There are three voltage levels: positive, negative, and zero
- Types: AMI, B8ZS, HDB3

Bipolar AMI Encoding

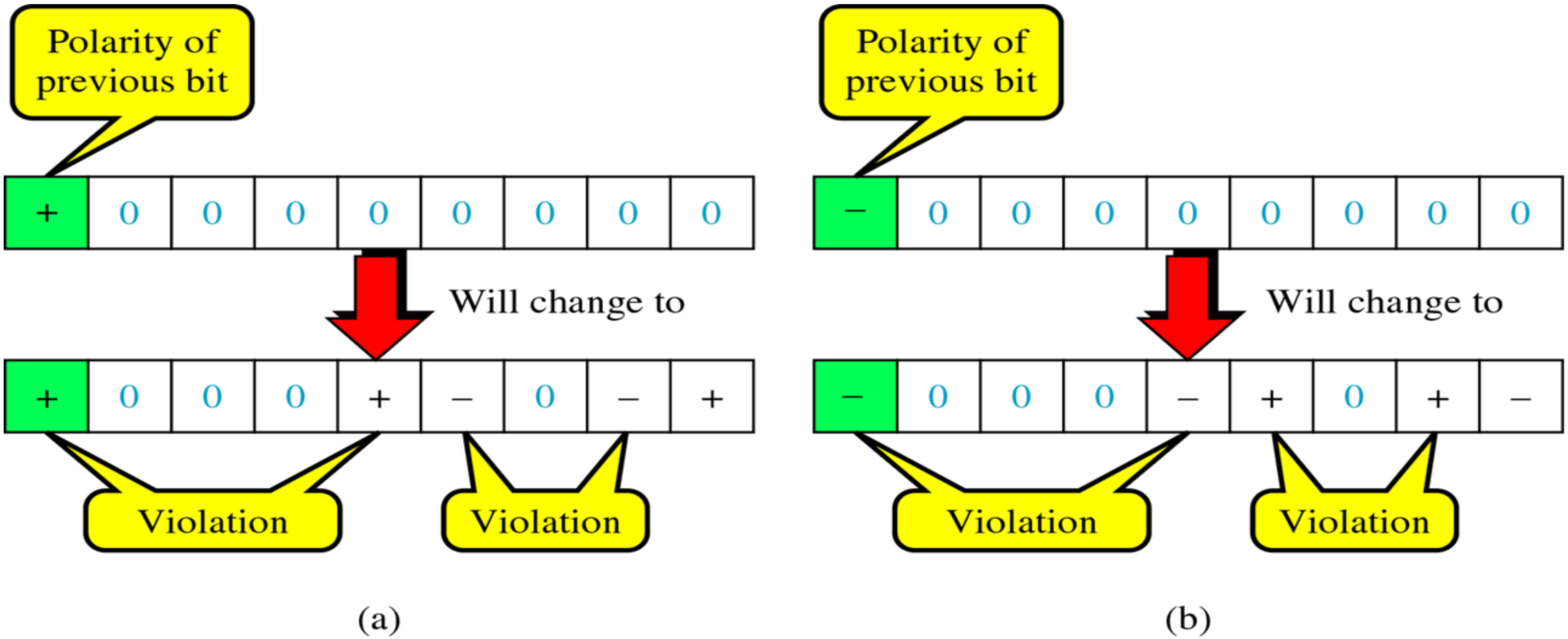


- AMI (alternate mark inversion): A neutral zero voltage represents binary 0. Binary 1s are represented by alternating positive and negative voltages.

Scrambling

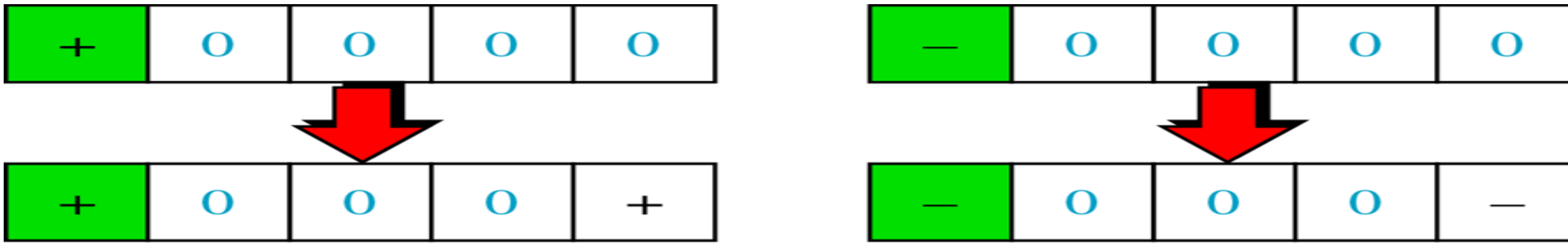
- Biphase schemes are not suitable for long-distance communication due to their wide bandwidth requirement.
- Combination of block coding and NRZ line coding is not suitable for long-distance encoding, because of the DC component.
- Bipolar AMI encoding, has a narrow bandwidth and does not create a DC component.
- A long sequence of 0s upsets the synchronization.
- A long sequence of 0s in the original stream is avoided by substituting long zero-level pulses with a combination of other levels to provide synchronization.
- Number of bits not increased and synchronization assured
- Common scrambling techniques: B8ZS and HDB3

B8ZS Encoding

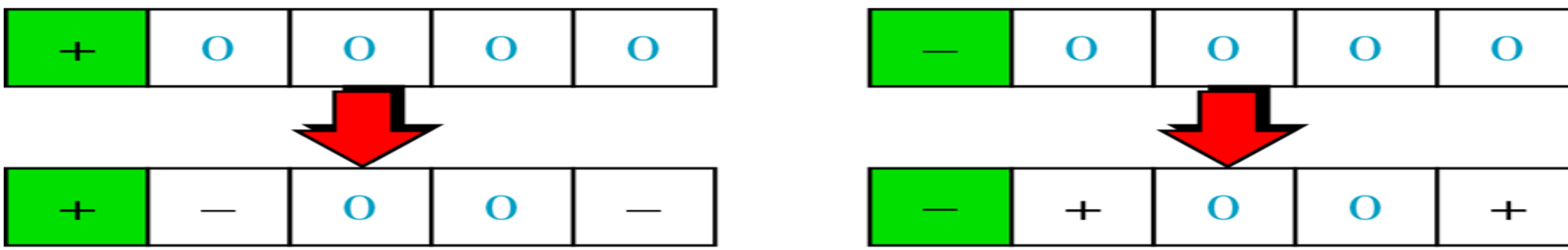


- B8ZS (bipolar with 8-zeros substitution)
- If an octet of all zeros occurs and the last voltage pulse preceding this octet was positive, then the eight zeros of the octet are encoded as 000+ -0- +
- If an octet of all zeros occurs and the last voltage pulse preceding this octet was negative, then the eight zeros of the octet are encoded as 000- +0+ -
- That is the pattern of violation is same, but with inverted polarities

HDB3 Encoding



(a) If the number of 1s since the last substitution is odd



(b) If the number of 1s since the last substitution is even

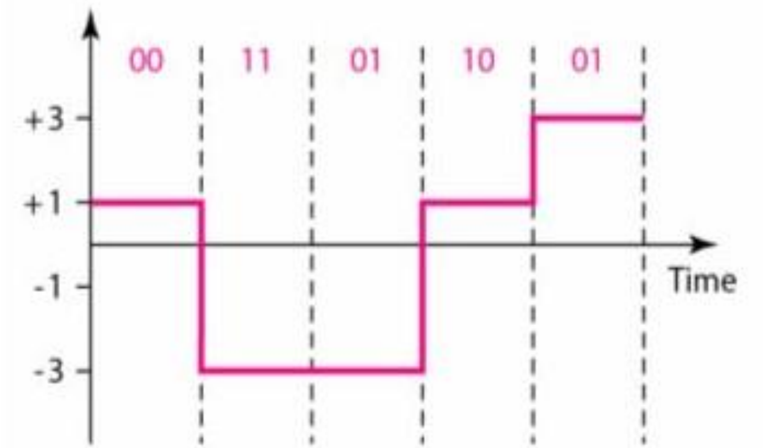
- HDB3 (high-density bipolar-3 zeros)
- Fourth zero is replaced with a code violation.
- In addition, a rule is needed to ensure that successive violations are of alternate polarity so that no dc component is introduced.
- Thus, if the last violation was positive, this violation must be negative and vice versa.

Multilevel – 2B1Q

- 2B1Q (2 Binary 1 Quaternary) Coding
- Data patterns of size 2 bits are encoded as one signal element belonging to a four-level signal
  - data is sent two time faster than with NRZ-L
  - receiver has to discern 4 different thresholds

	Previous level: positive	Previous level: negative
Next bits	Next level	Next level
00	+1	-1
01	+3	-3
10	-1	+1
11	-3	+3

Transition table



Assuming positive original level

Block Coding

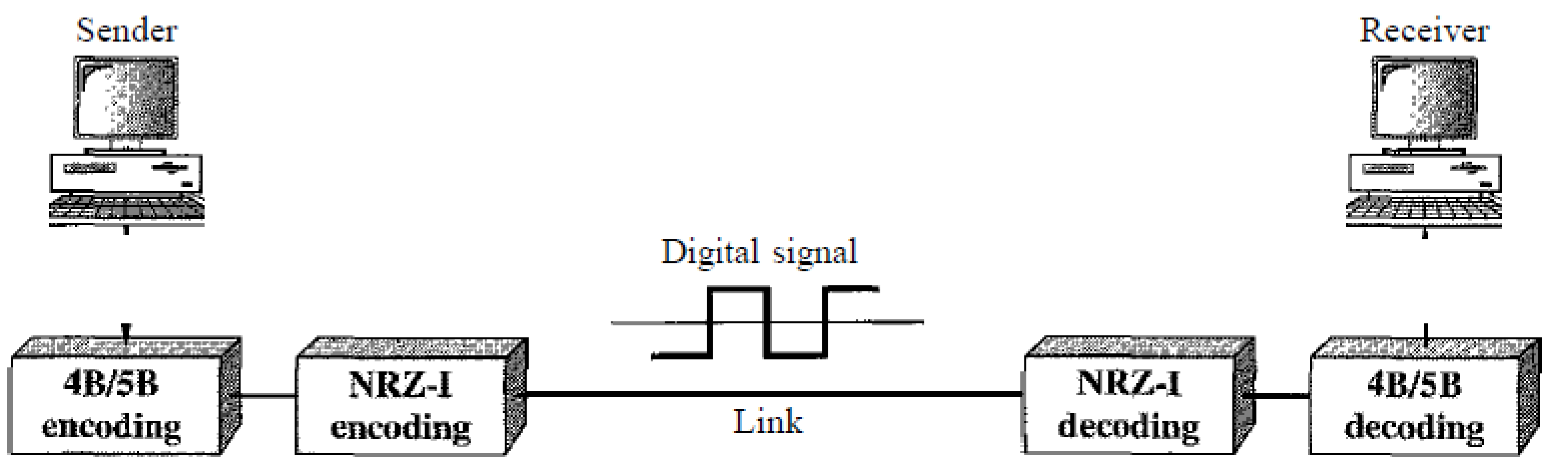
- Block coding changes a block of *m* bits into a block of *n* bits, where *n* is larger than *m.*
- Block coding is referred to as an *mB/nB* encoding technique
- Normally involves three steps: division, substitution, and combination.
  - Division step: a sequence of bits is divided into groups of *m* bits.
  - Substitution step: substitute an m-bit group for an n-bit group.
  - Combination step: n-bit groups are combined together to form a stream.

4B/5B

- Four binary/five binary (4B/5B) coding scheme designed to be used in combination with NRZ-I.
- NRZ-I good signal rate but synchronization problem.
- In 4B/5B, the 5-bit output that replaces the 4-bit input has no more than one leading zero (left bit) and no more than two trailing zeros (right bits).

4B/5B

Figure Using block coding 4B/5B with NRZ-I line coding scheme



4B/5B

Table *4B/5B mapping codes*

Data Sequence	*Encoded Sequence*	*Control Sequence*	*Encoded Sequence*
0000	11110	Q (Quiet)	00000
0001	01001	I (Idle)	11111
0010	10100	H (Halt)	00100
0011	10101	J (Start delimiter)	11000
0100	01010	K (Start delimiter)	10001
0101	01011	T (End delimiter)	01101

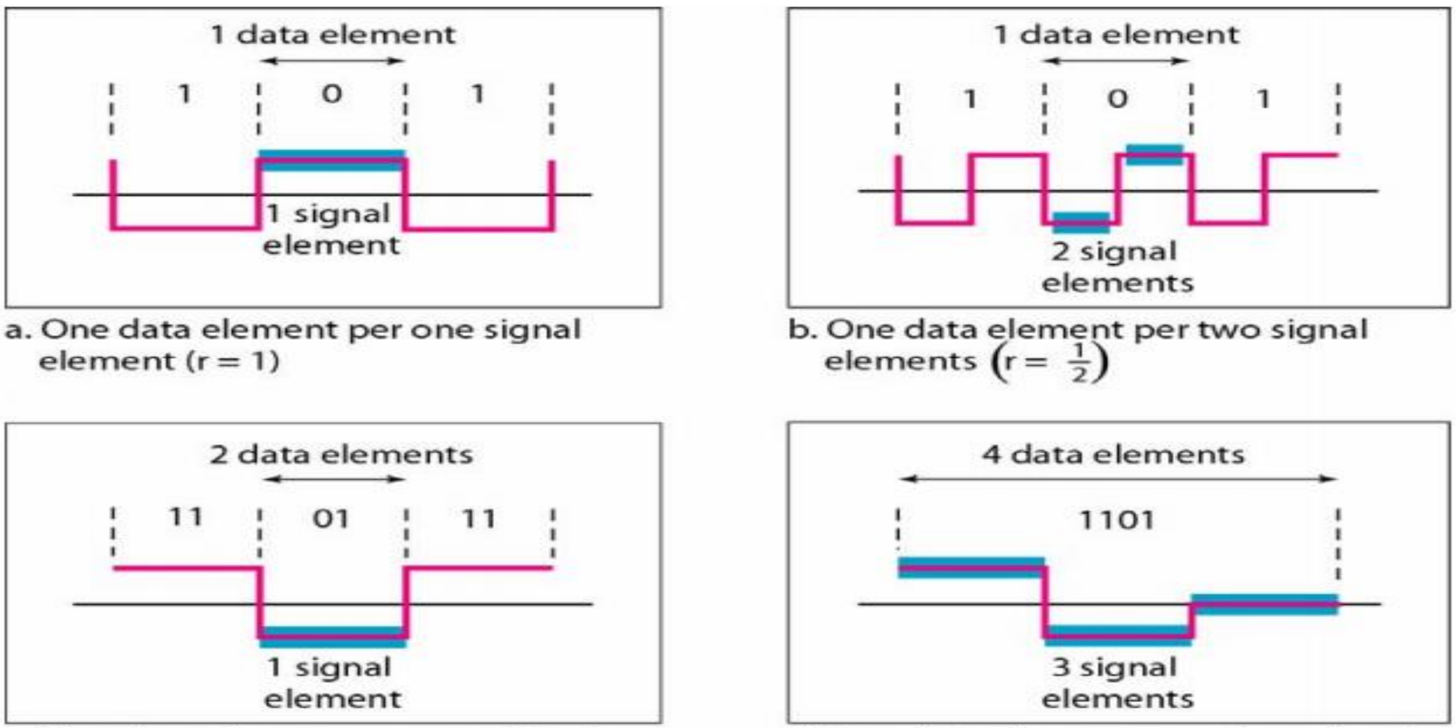
4B/5B contd.

Table *4B/5B mapping codes (continued)*

Data Sequence	*Encoded Sequence*	*Control Sequence*	*Encoded Sequence*
0110	01110	S (Set)	11001
0111	01111	R (Reset)	00111
1000	10010		
1001	10011		
1010	10110		
1011	10111		
1100	11 010		
1101	11011		
1110	11100		
1111	11101		

Data Rate vs. Baud Rate

- Data Rate – # of data elements (bits) sent in 1 sec – unit: bps
- Signal Rate – # of signal elements/pulses sent in 1 sec – unit: baud



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

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

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

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

Data Rate vs. Baud Rate

- One goal of data communications is to increase data rate (speed of transmission) while decreasing signal rate (bandwidth requirements).

r = data rate / signal rate – **ratio between data & signal rate**

Signal rate observed in case of a particular data-bit stream: – **depends on N [bps], 1/r [bit/pulse], and <u>the actual data pattern</u>**

- **signal rate for a pattern of all 1-s or all 0-s may be different from that for a patter of alternating 1-s and 0-s**

$$S = \underbrace{c}_{\text{case factor}} \cdot N \cdot \frac{1}{r} \text{ [pulses/sec]}$$

Data Rate vs. Baud Rate

Example [data vs. signal rate]

A signal is carrying data in which one data element is encoded as one signal element (r=1).

If the bit rate is 100 kbps, what is the average value of the baud rate, assuming c is between 0 and 1?

Answer:

$C_{average} = 0.5$

$$S = c \cdot N \cdot \frac{1}{r} = \frac{1}{2} \cdot 100{,}000 \cdot \frac{1}{1} = 50{,}000 \text{ [pulses/sec]} = 50 \text{ [kbaud]}$$

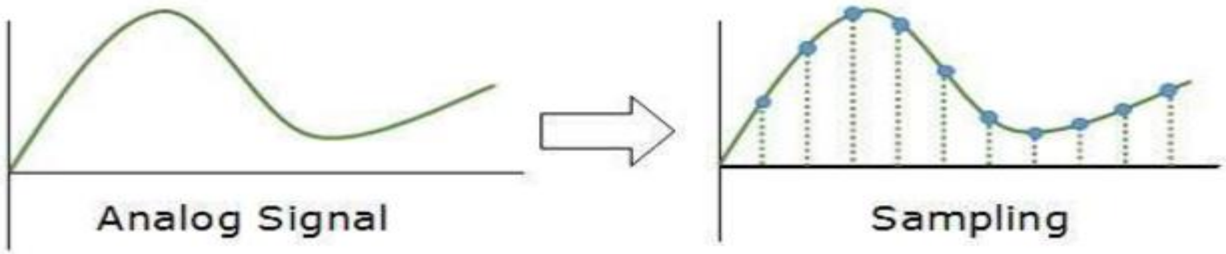
Analog to Digital Encoding

Analog to Digital

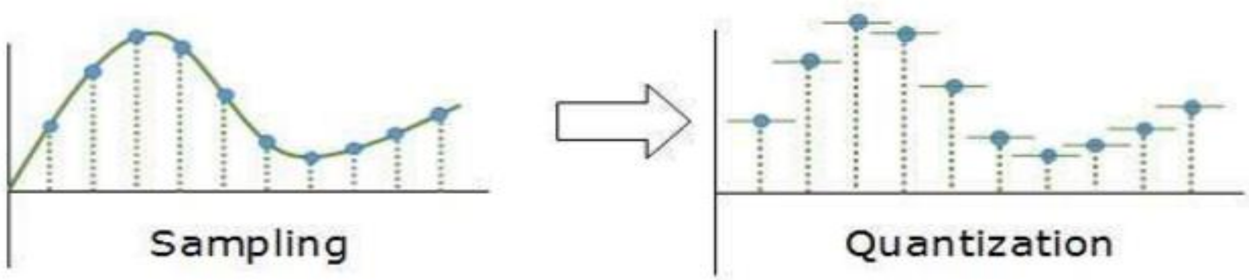
- Microphones create analog voice and camera creates analog videos. To transmit this analog data over digital signals, we need analog to digital conversion.
- To convert analog wave into digital data, we use Pulse Code Modulation (PCM).
- PCM involves three steps:
  - Sampling
  - Quantization
  - Encoding.

PCM

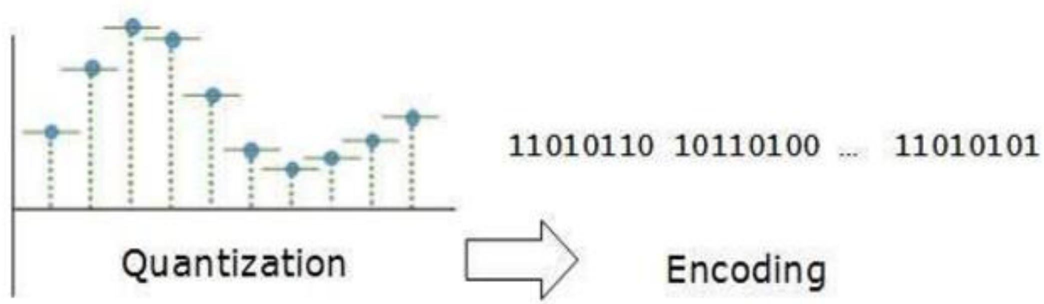
Sampling



Quantization



Encoding



PCM

- Sampling
  - The analog signal is sampled every T interval. Most important factor in sampling is the rate at which analog signal is sampled.
  - **According to Nyquist Theorem, the sampling rate must be at least two times of the highest frequency of the signal.**
- Quantization
  - quantization is done between the maximum amplitude value and the minimum amplitude value.
  - Quantization is approximation of the instantaneous analog value.
- Encoding
  - In encoding, each approximated value is then converted into binary format.

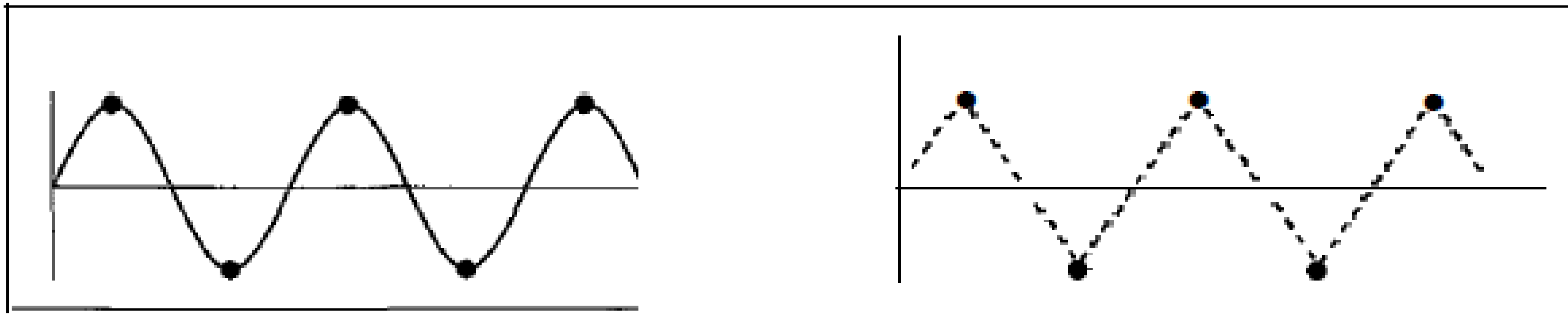
Sampling Rate

Example

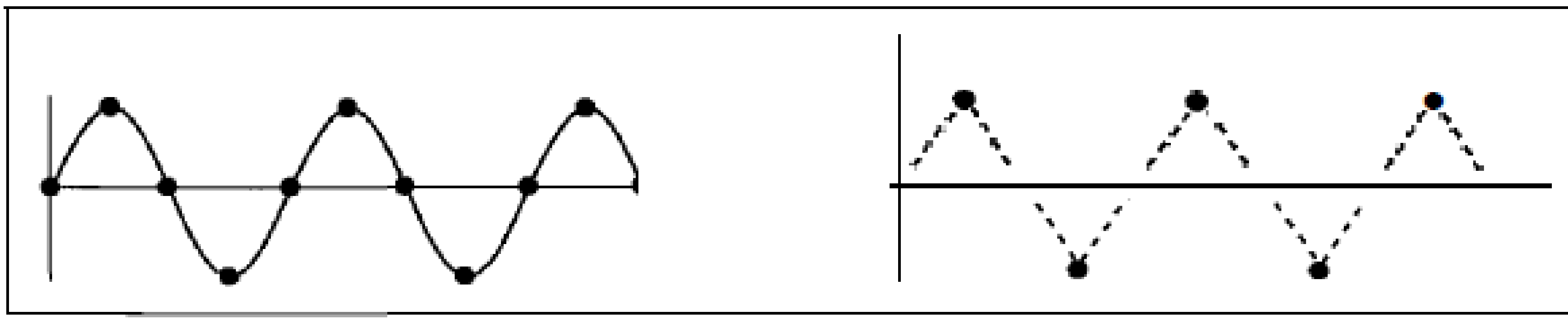
- For an intuitive example of the Nyquist theorem, let us sample a simple sine wave at three sampling rates: *fs = 4f* (2 times the Nyquist rate*) fs = 2f* (Nyquist rate), and *fs =f* (one-half the Nyquist rate). Figure next shows the sampling and the subsequent recovery of the signal.

Ans:

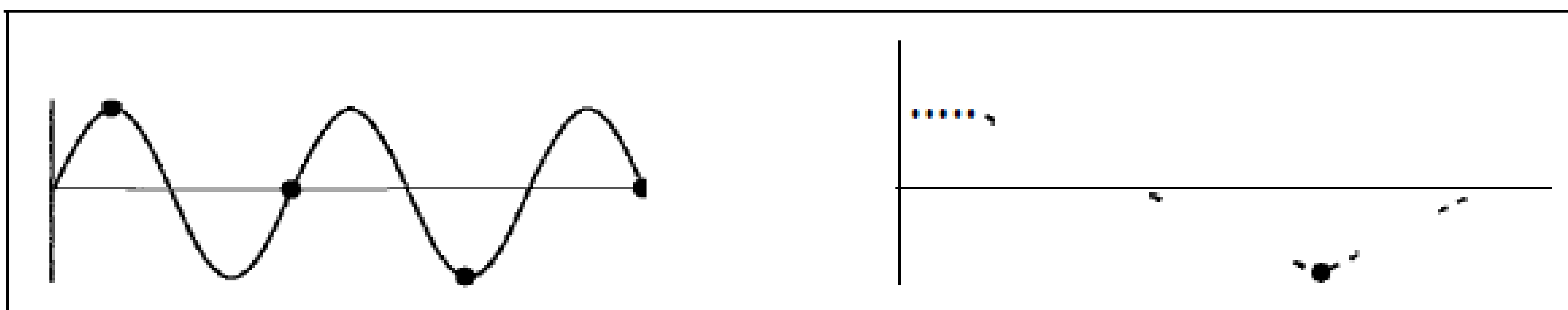
- It can be seen that sampling at the Nyquist rate can create a good approximation of the original sine wave (part a). Oversampling in part b can also create the same approximation, but it is redundant and unnecessary. Sampling below the Nyquist rate (part c) does not produce a signal that looks like the original sine wave.



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



b. Oversampling: $f_s = 4f$



c. Undersampling: $f_s = f$

Sampling rate: examples

Example 1

- Telephone companies digitize voice by assuming a maximum frequency of 4000 Hz. The sampling rate therefore is 8000 samples per second.

Example 2

- 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 3

- 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.

Quantization steps

- Assume that the original analog signal has instantaneous amplitudes between V_{min} and V_{max}
- Divide the range into L zones, each of height Δ(delta).
- $\Delta = (V_{max} - V_{min})/L$
- Assign quantized values of 0 to L - 1 to the midpoint of each zone.
- Approximate the value of the sample amplitude to the quantized values.

Quantization error

- Quantization is an approximation process. The input values to the quantizer are the real values; the output values are the approximated values.
- Difference between actual and approximate gives the quantization error.
- The value of the error for any sample is less than $\Delta/2$. That is: $-\Delta/2 \le \text{error} \le +\Delta/2$
- Contribution of the quantization error to the SNR_{dB} of the signal depends on the number of quantization levels *L,* or the bits per sample *nb*

$$SNR_{dB} = 6.02nb + 1.76 \text{ dB}$$

Examples: Quantization error

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

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

Telephone companies usually assign 7 or 8 bits per sample.

Encoding

Bit rate :::: sampling rate x number of bits per sample:::: f_s x nb

Example 4.14

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

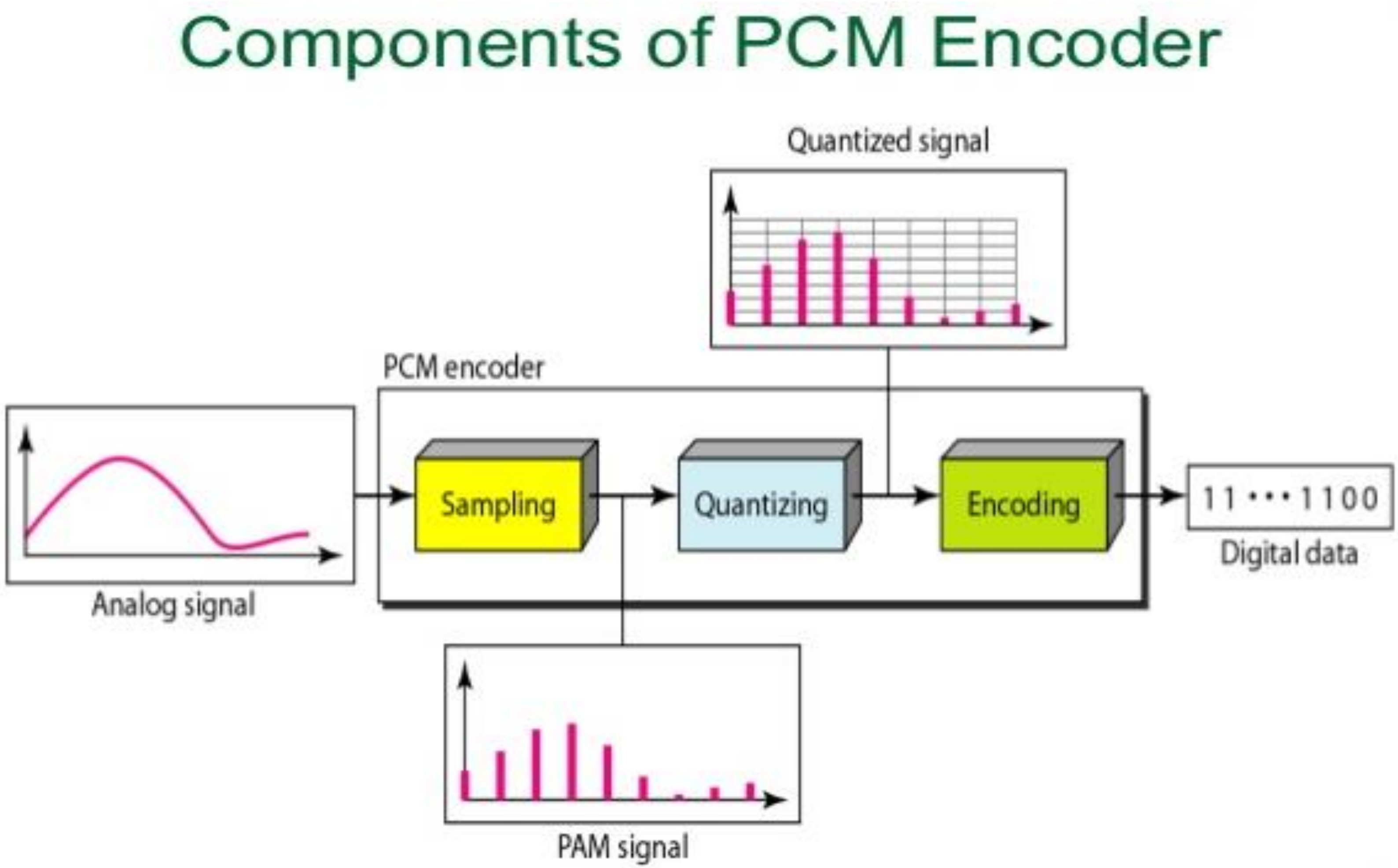
Solution

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

Sampling rate :::: 4000 x 2 :::: 8000 samples/s

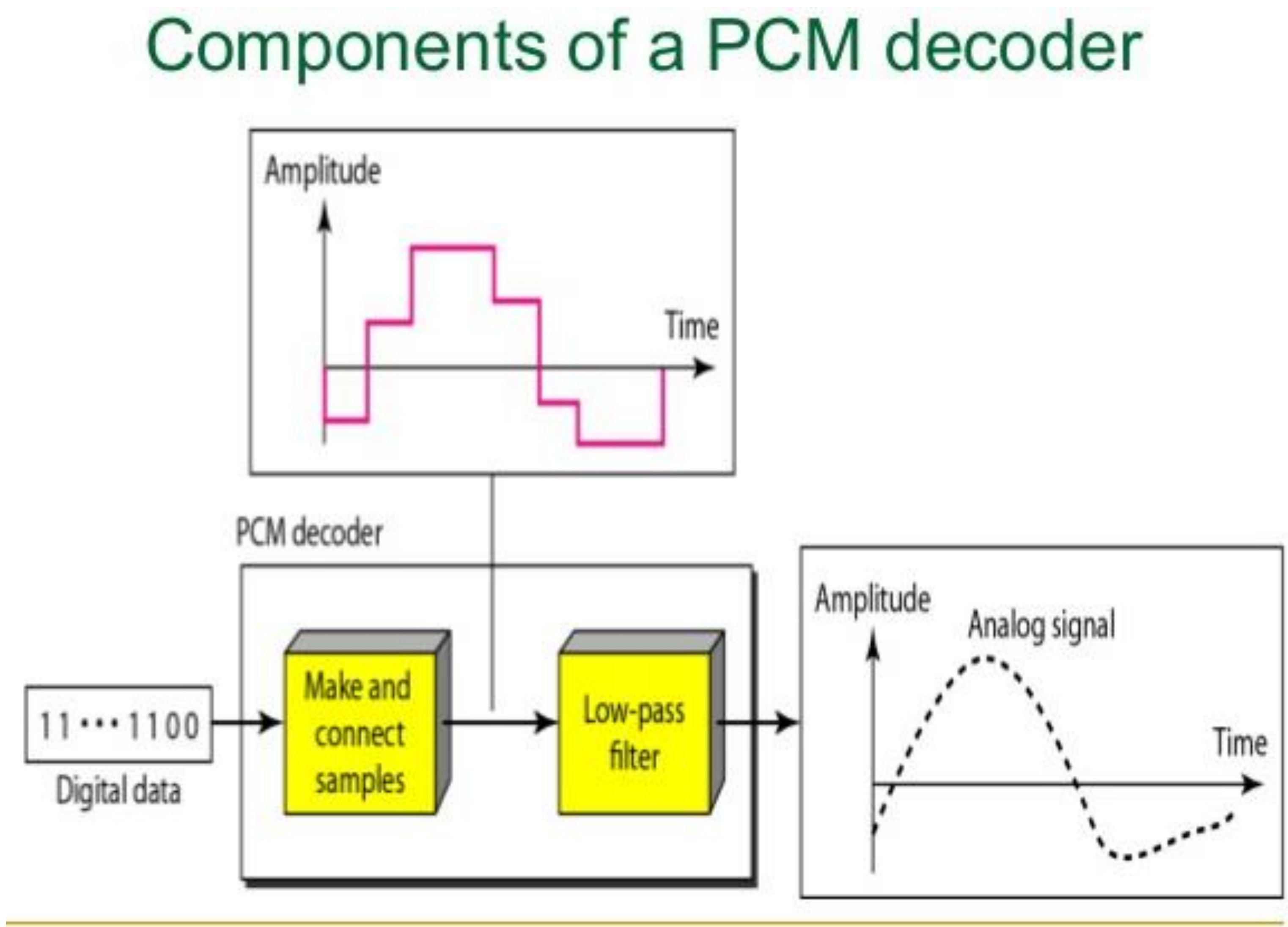
Bit rate = 8000 x 8 :::: 64,000 bps = 64 kbps

PCM Encoder

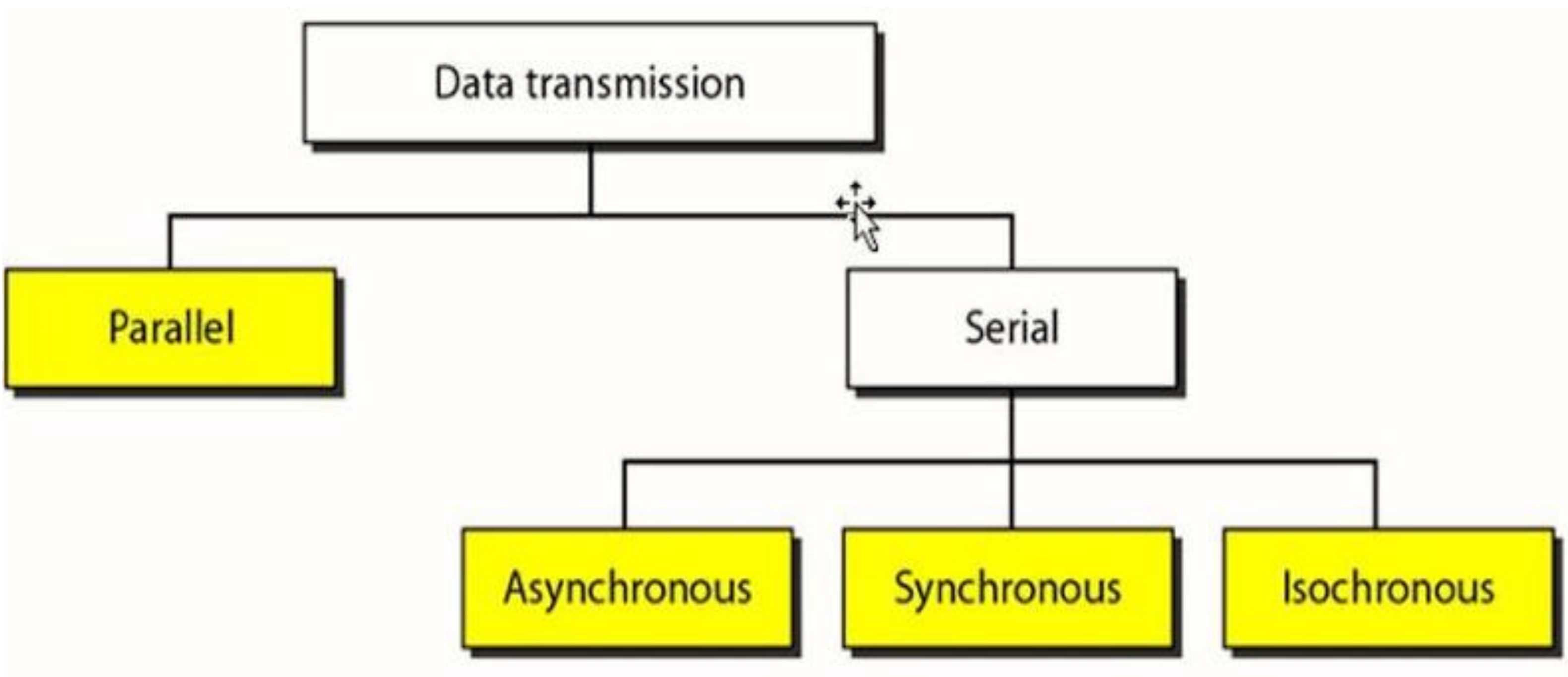


Components of PCM Encoder
Quantized signal
PCM encoder
Sampling
Quantizing
Encoding
11 ••• 1100
Digital data
Analog signal
PAM signal

PCM Decoder


Components of a PCM decoder
Amplitude
Time
PCM decoder
11 ••• 1100
Digital data
Make and connect samples
Low-pass filter
Amplitude
Analog signal
Time

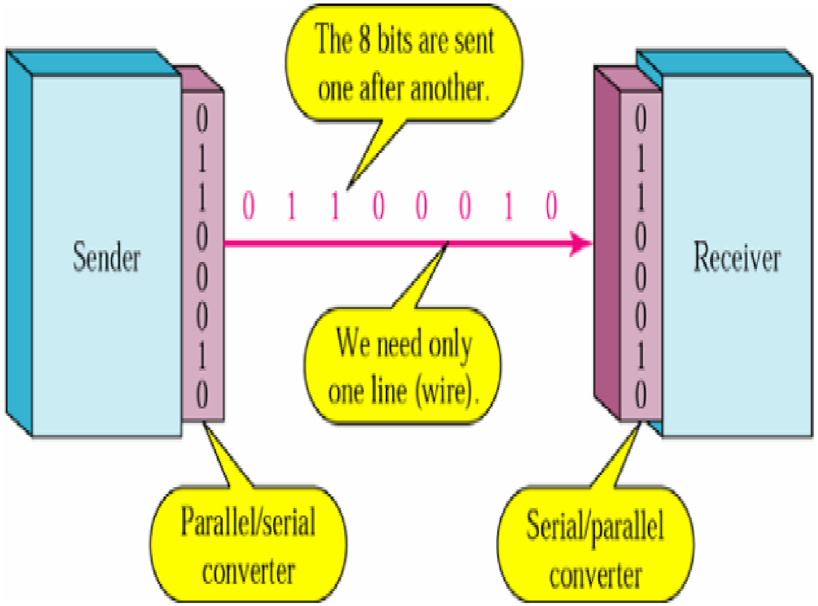
Digital Transmission Modes

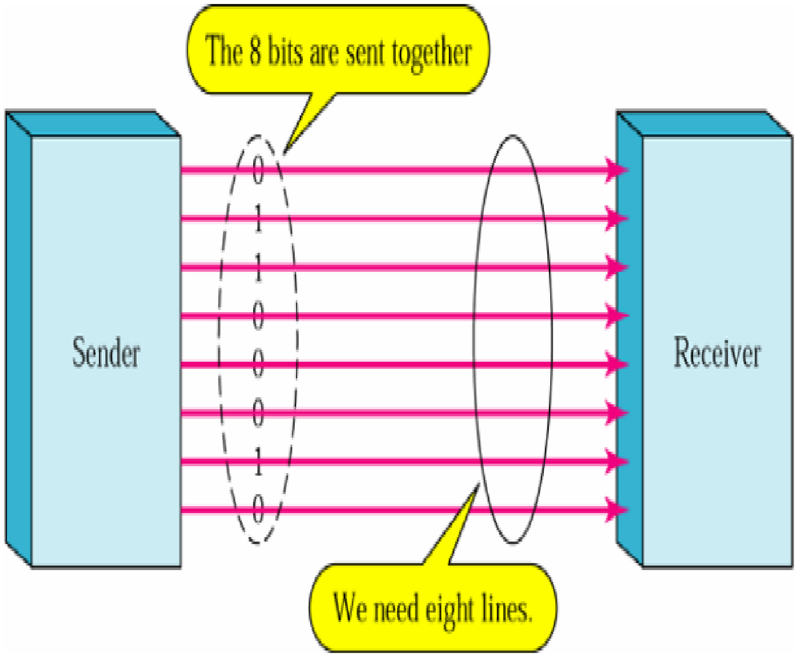


Digital Transmission Modes

- **Serial Mode**: 1 bit is sent with each clock tick
  - one communication channel / wire is needed
- **Parallel Mode**: multiple bits are sent with each clock tick
  - multiple channels / wires, bundled in one cable, are required
  - **advantage**: n-times faster than serial mode
  - **disadvantage**: cost = 8x wires (used only over short distances)

Digital Transmission Modes



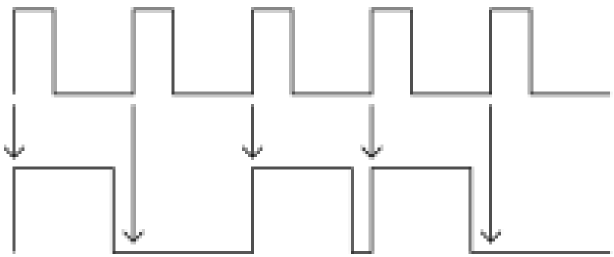


Digital Transmission Modes

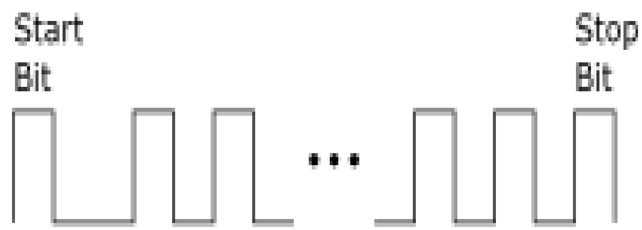
- **Synchronous**
  - ✓ Bits in a synchronous data stream must be transferred in sync with a clock signal.
  - ✓ Systems usually have an error detection mechanism.
  - ✓ If an error is detected the data can be resent.
- **Asynchronous**
  - ✓ Bits in an asynchronous data stream can be transferred at random intervals and the data rate of the stream is not required to be constant.
  - ✓ Systems use a start bit & a stop bit to signal the start & end of a data transmission.
  - ✓ Asynchronous data transfer systems usually have an error detection mechanism.
  - ✓ If an error is detected the data can be resent.

Digital Transmission Modes

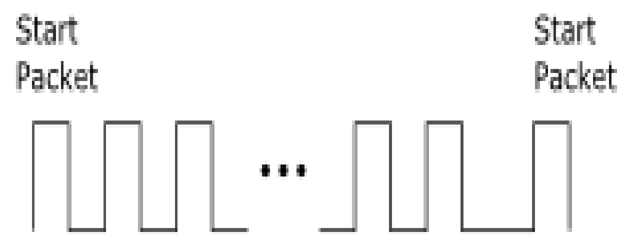
- Isochronous
  - ✓ An isochronous data transfer system combines the features of an asynchronous and synchronous data transfer system.
  - ✓ System sends blocks of data asynchronously, (i.e. the data stream can be transferred at random intervals).
  - ✓ Each transmission begins with a start packet. Once the start packet is transmitted, the data must be delivered with a guaranteed bandwidth.
  - ✓ Isochronous data transfer is commonly used for where data must be delivered within certain time constraints, like streaming video.
  - ✓ Isochronous systems do not have an error detection mechanism (acknowledgment of receipt of packet) because if an error were detected, time constraints would make it impossible to resend the data.



Synchronous



Asynchronous



Isochronous

Digital-to-analog conversion

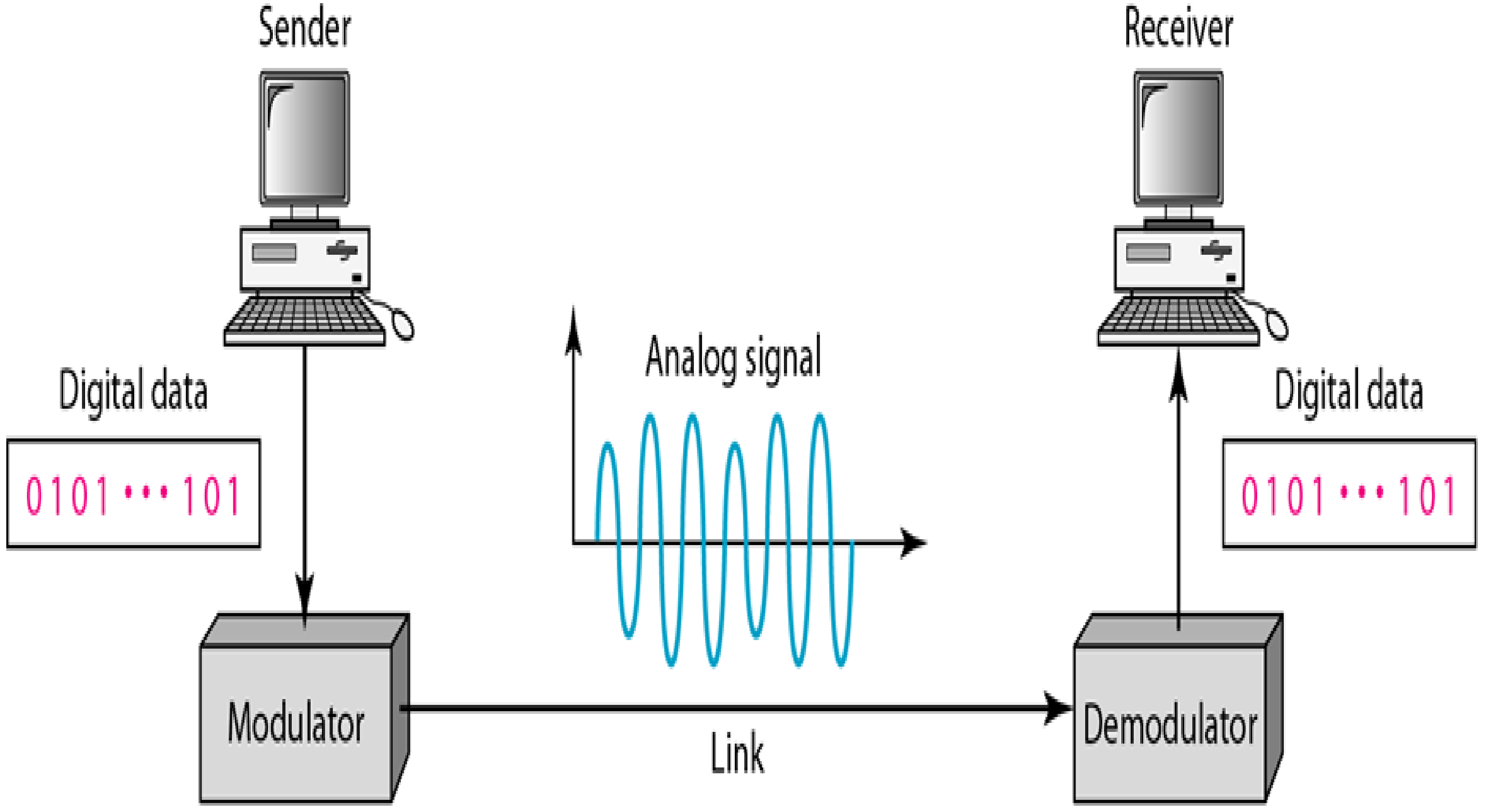
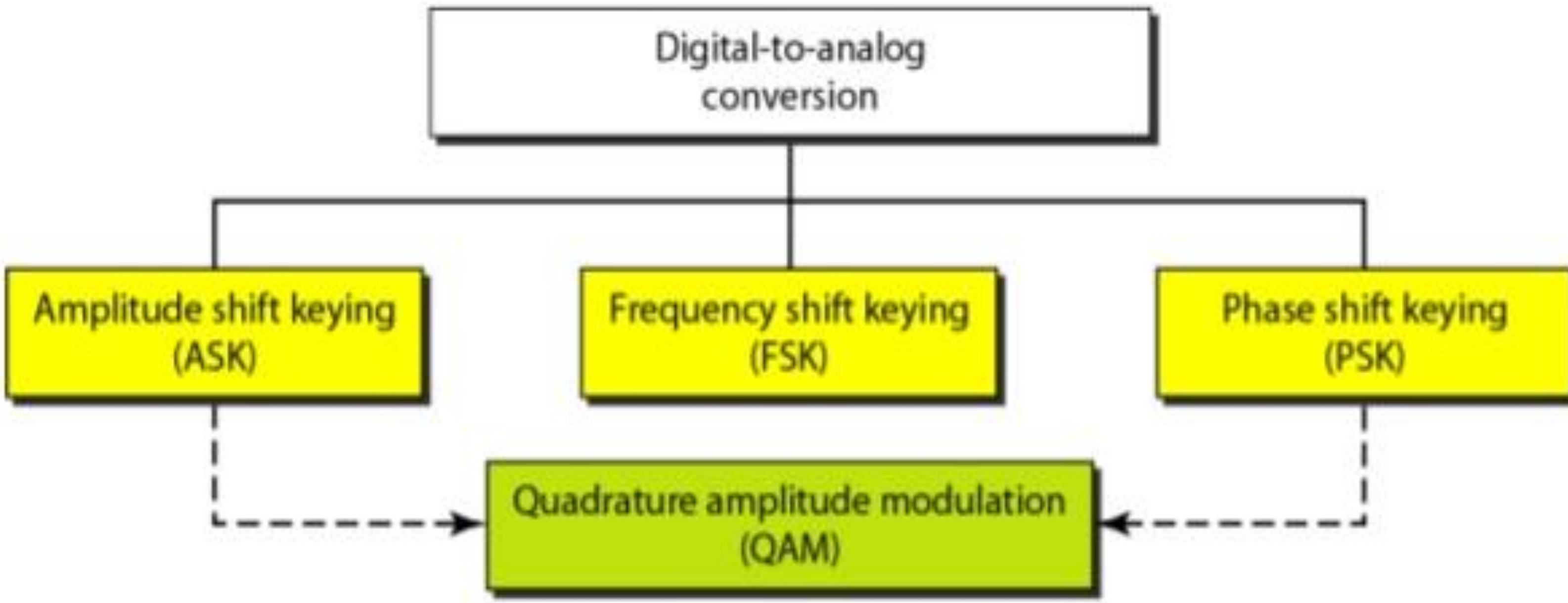
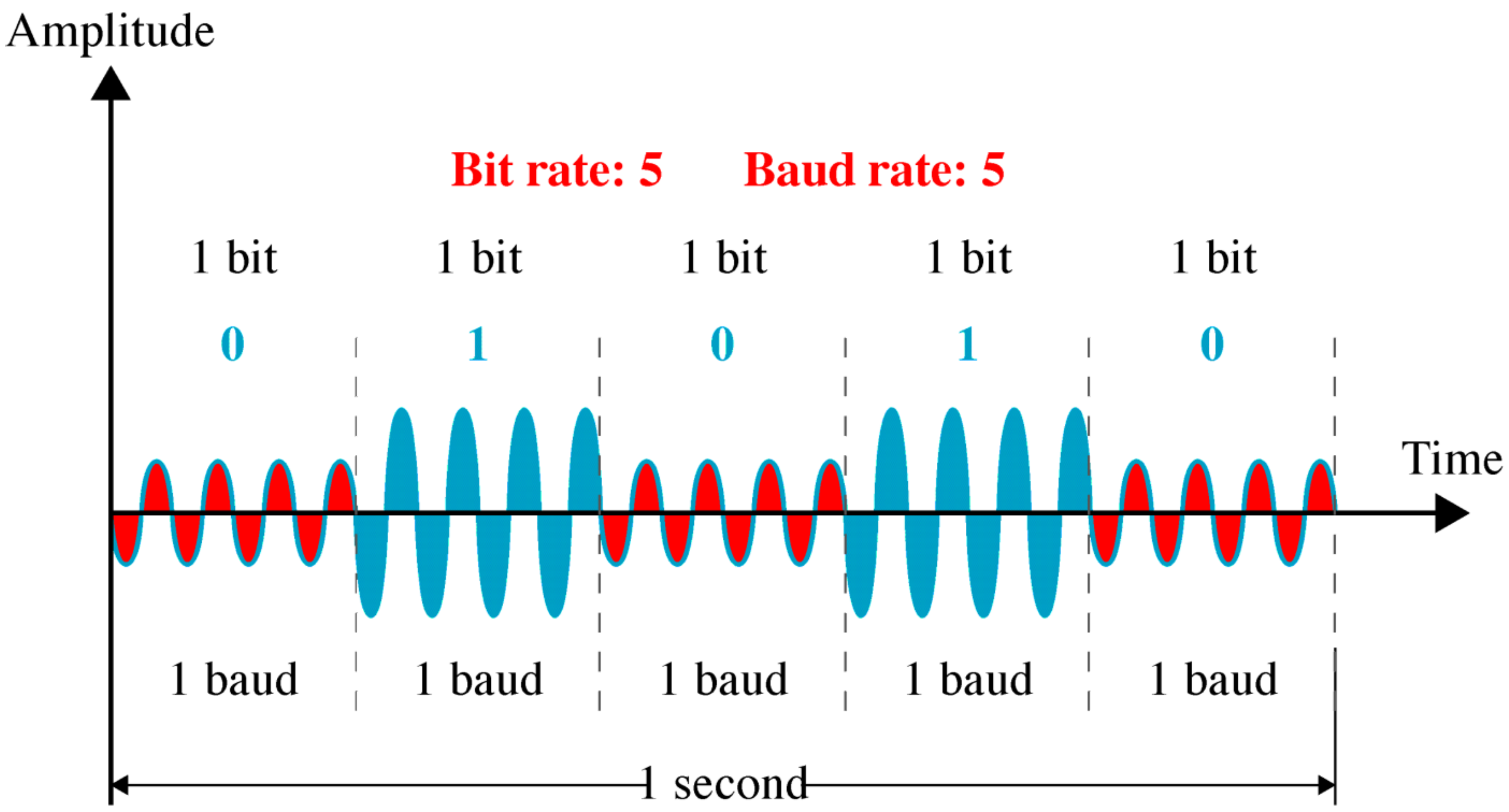

Sender
Receiver
Digital data
0101 ••• 101
Analog signal
Digital data
0101 ••• 101
Modulator
Link
Demodulator

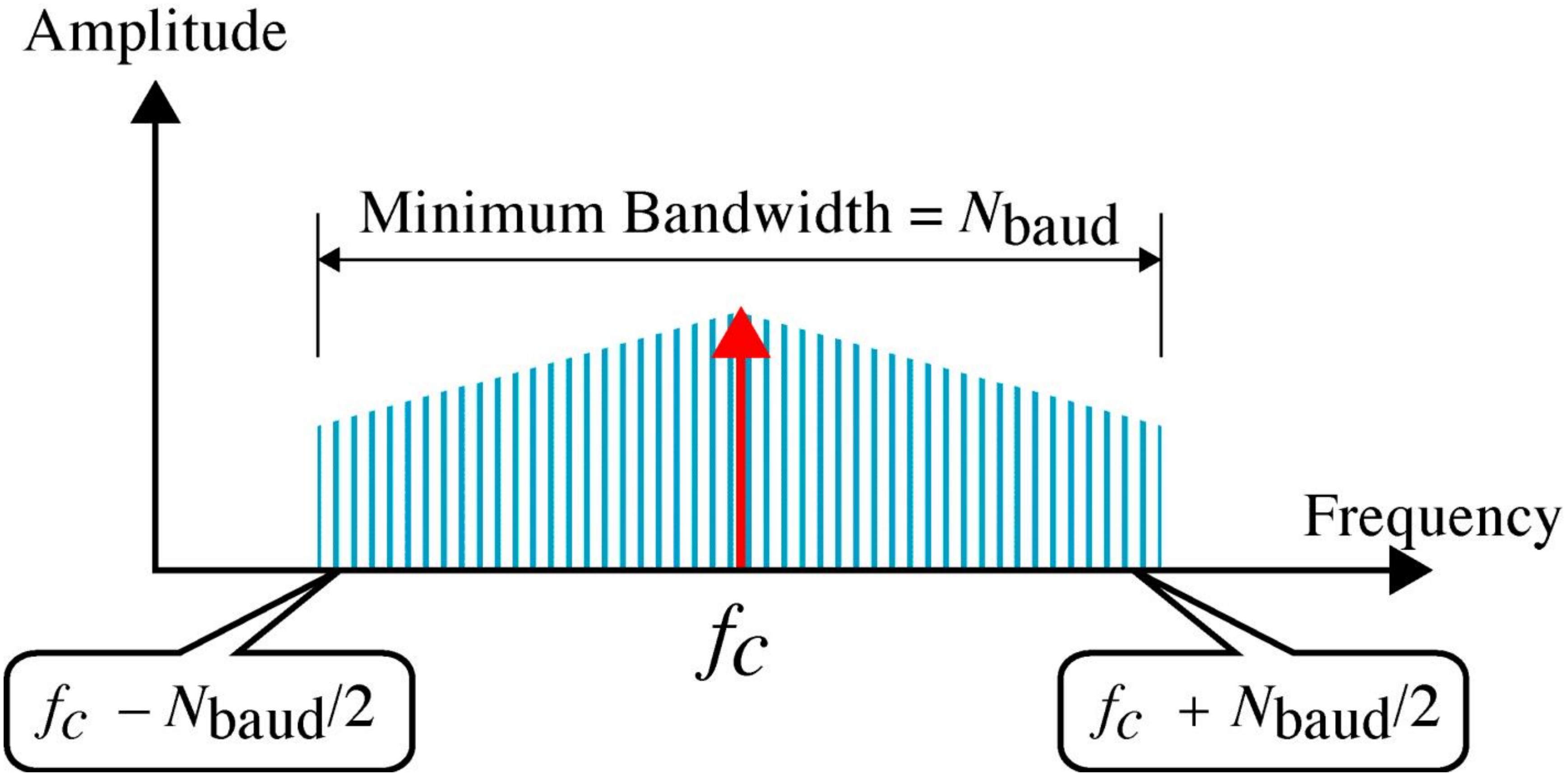
Figure 5.2 *Types of digital-to-analog conversion*



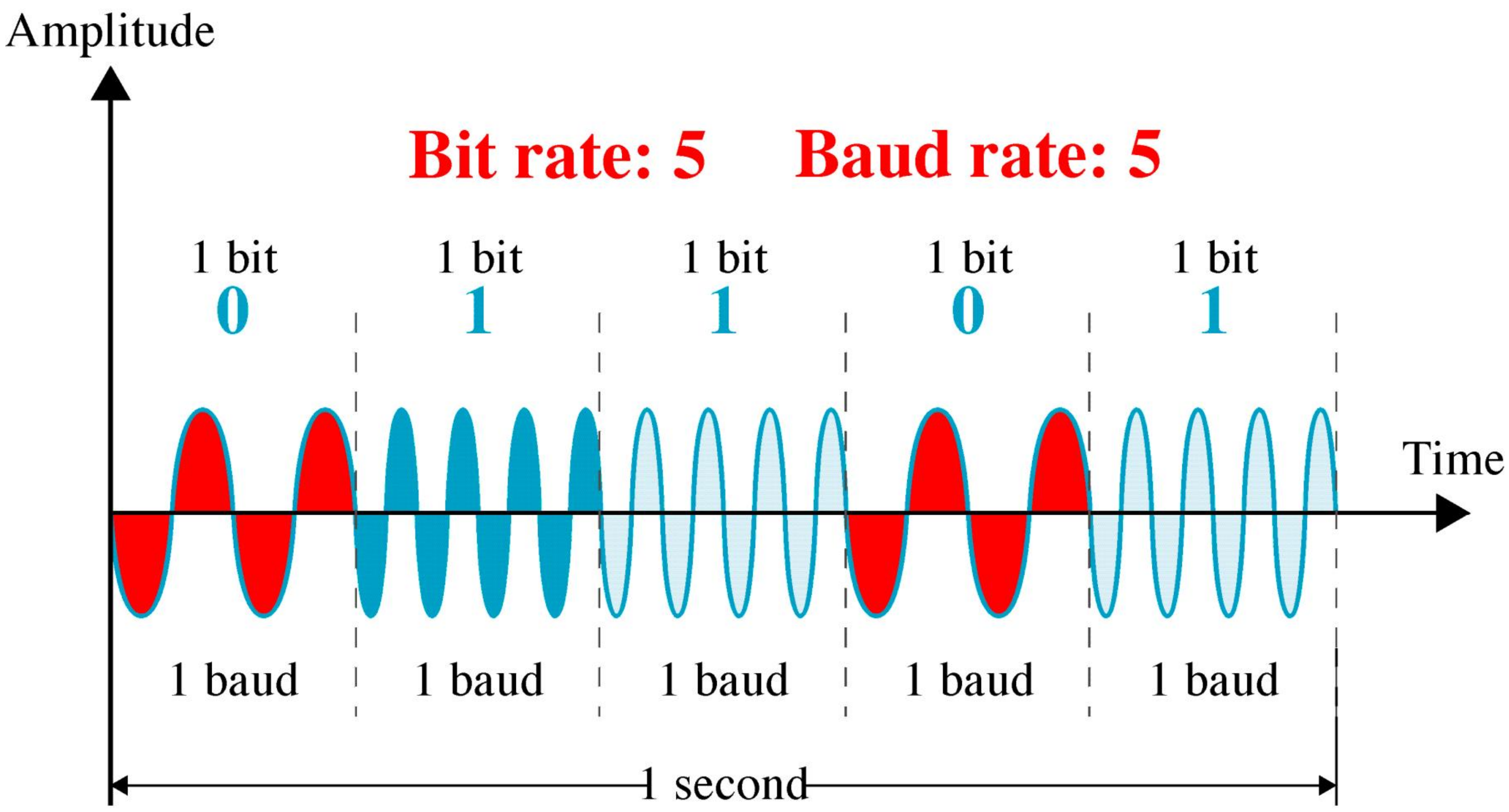
ASK


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

Bandwidth for ASK

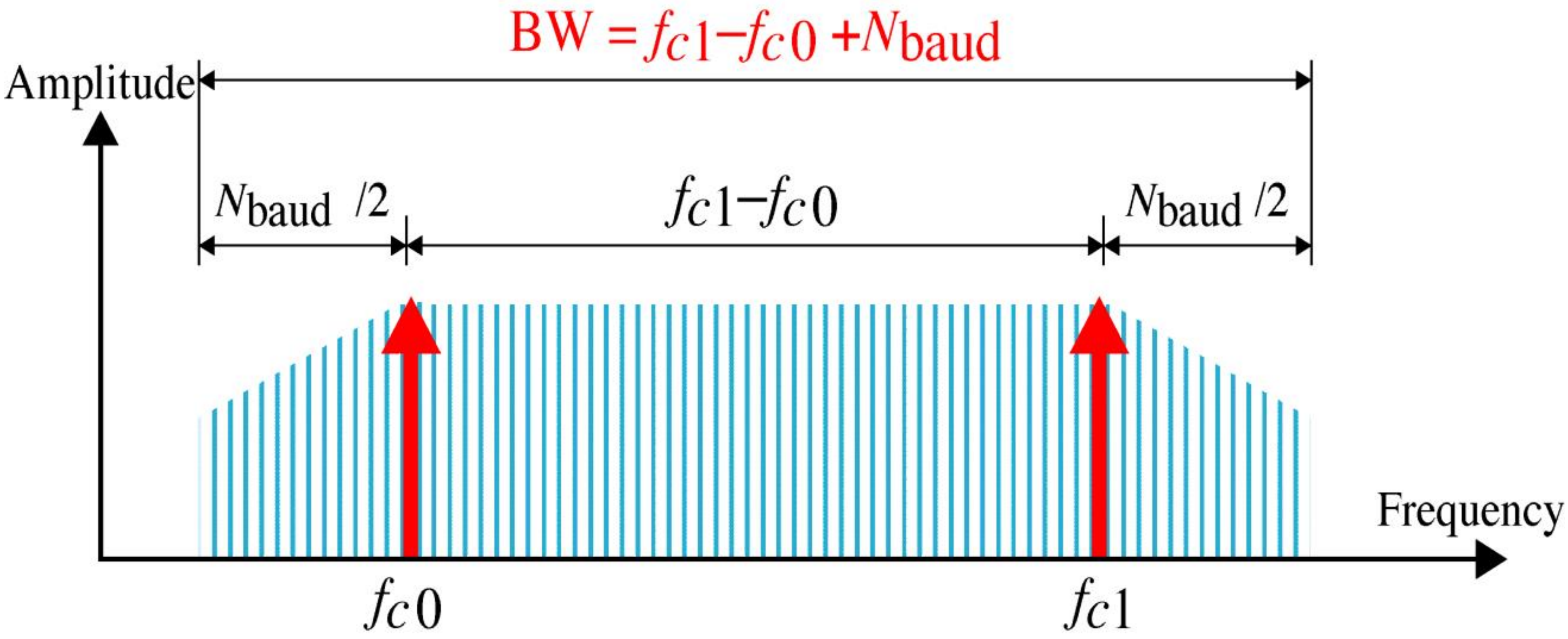


FSK

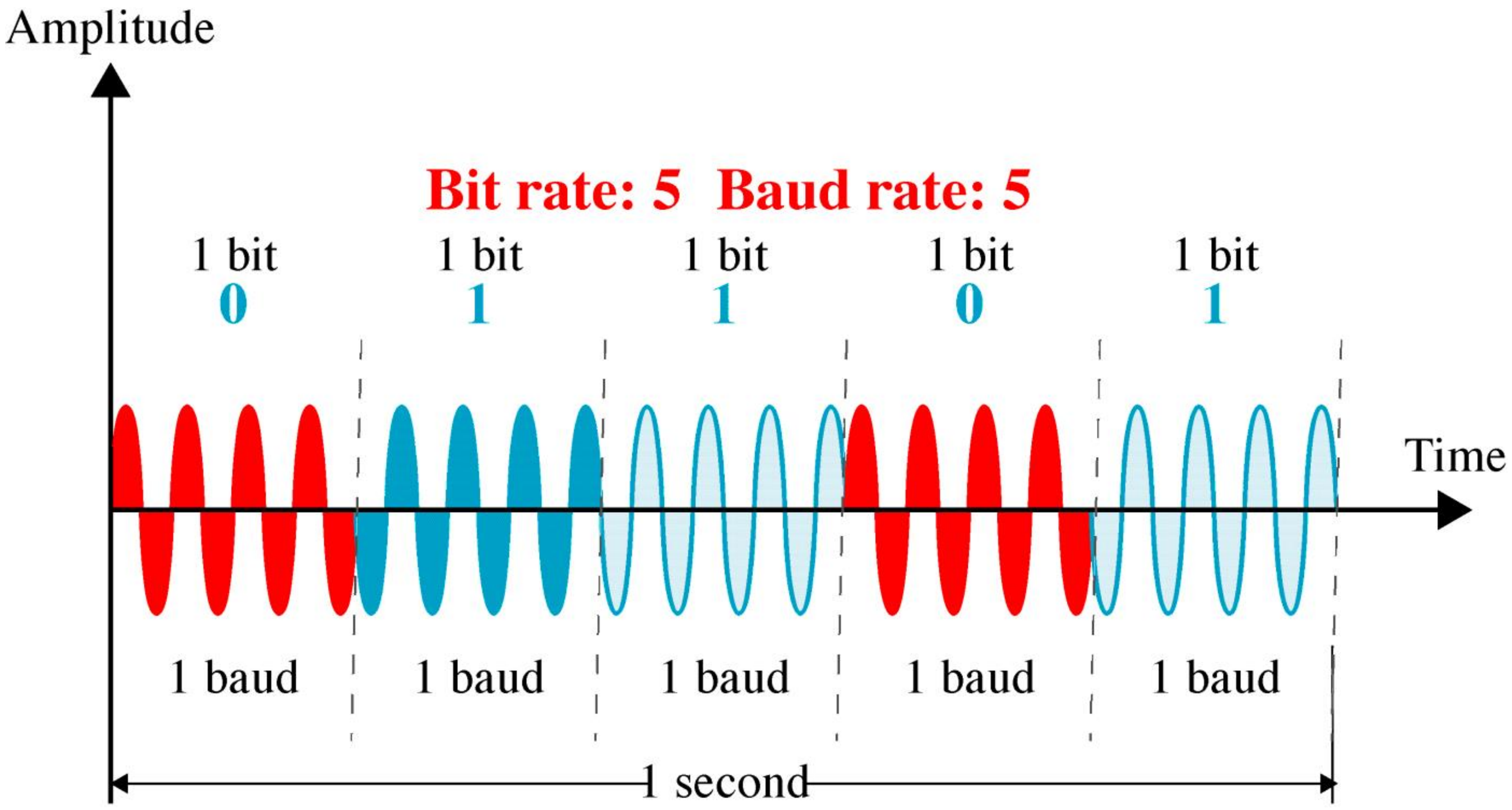


Amplitude
Bit rate: 5
Baud rate: 5
1 bit
0
1 bit
1
1 bit
1
1 bit
0
1 bit
1
Time
1 baud
1 baud
1 baud
1 baud
1 baud
1 second

Bandwidth for FSK



PSK

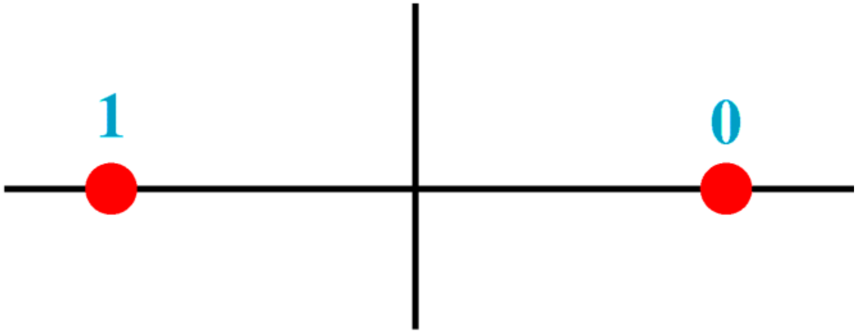


Amplitude
Bit rate: 5 Baud rate: 5
1 bit
0
1 bit
1
1 bit
1
1 bit
0
1 bit
1
Time
1 baud
1 baud
1 baud
1 baud
1 baud
1 second

PSK Constellation

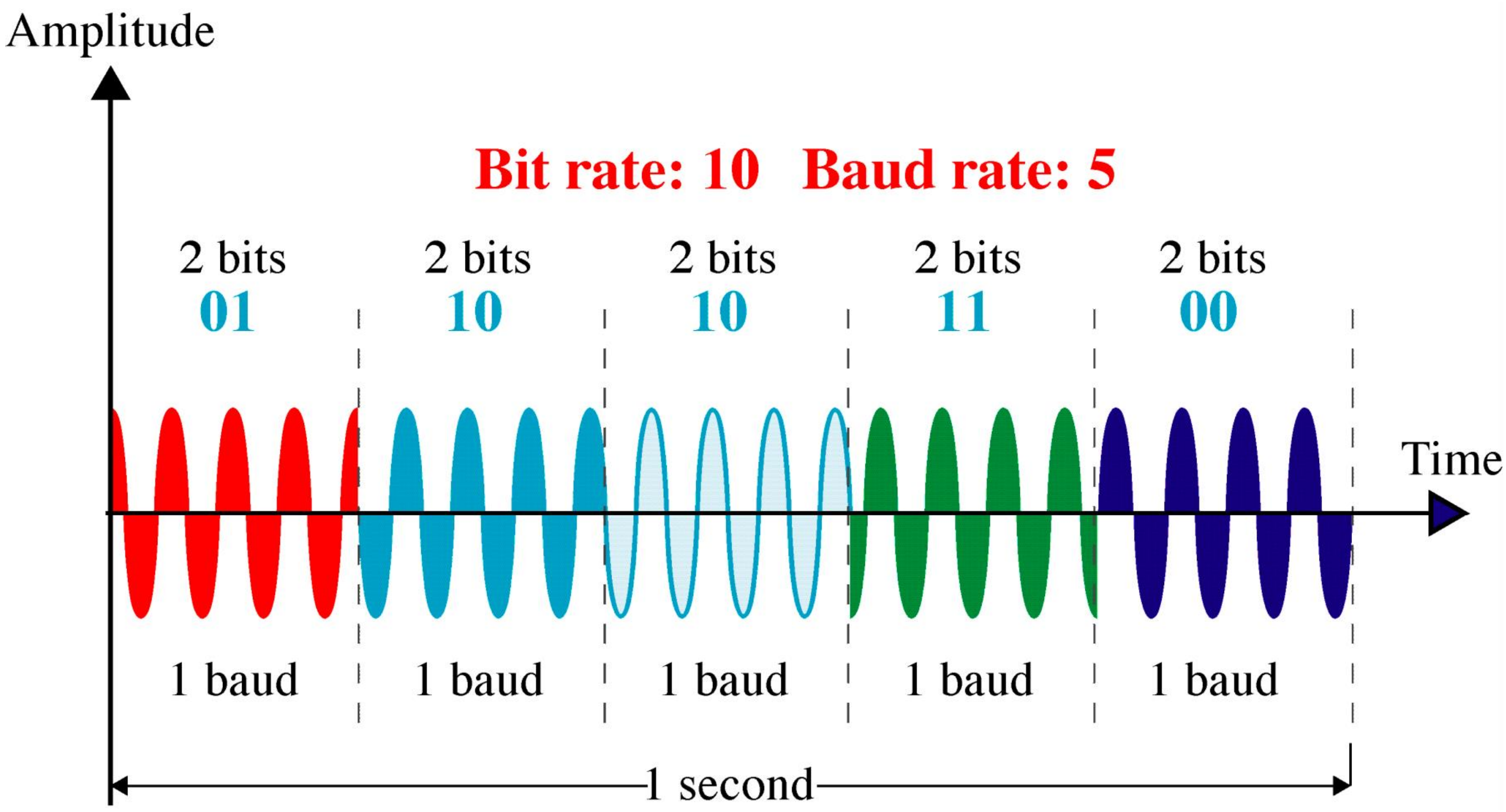
Bit	Phase
0	0
1	180

Bits



Constellation diagram

4-PSK

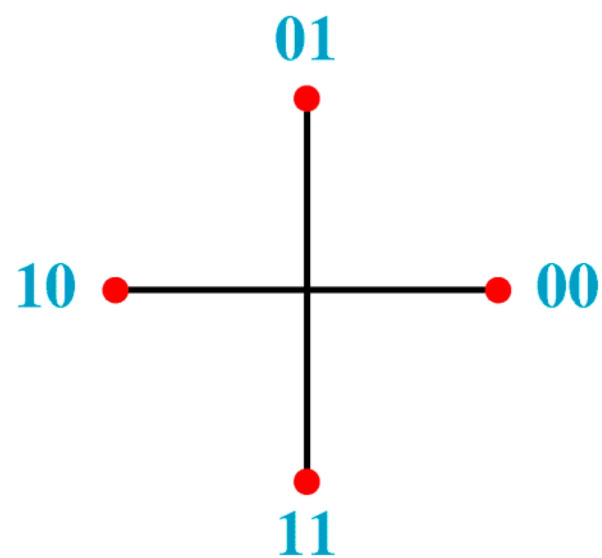


Amplitude
Bit rate: 10 Baud rate: 5
2 bits
01
2 bits
10
2 bits
10
2 bits
11
2 bits
00
Time
1 baud
1 baud
1 baud
1 baud
1 baud
1 second

4-PSK Characteristics

Dibit	Phase
00	0
01	90
10	180
11	270

Dibit
(2 bits)

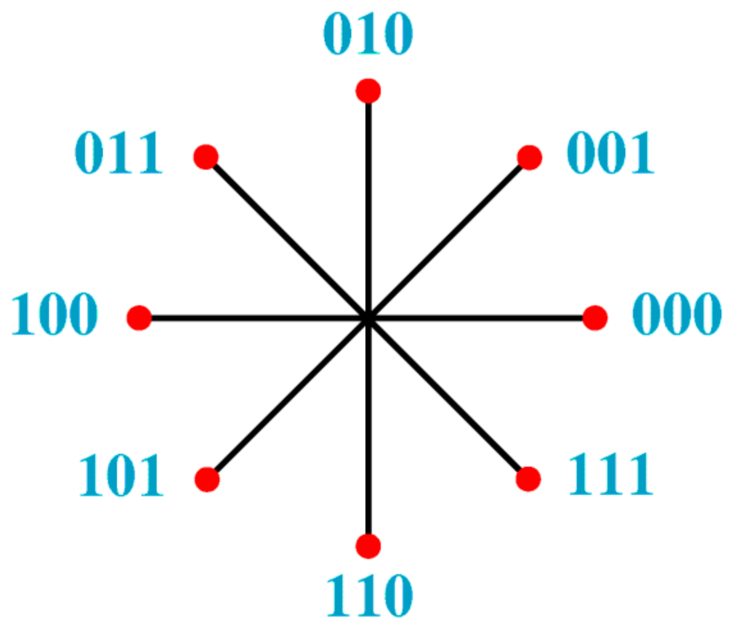


Constellation diagram

8-PSK Characteristics

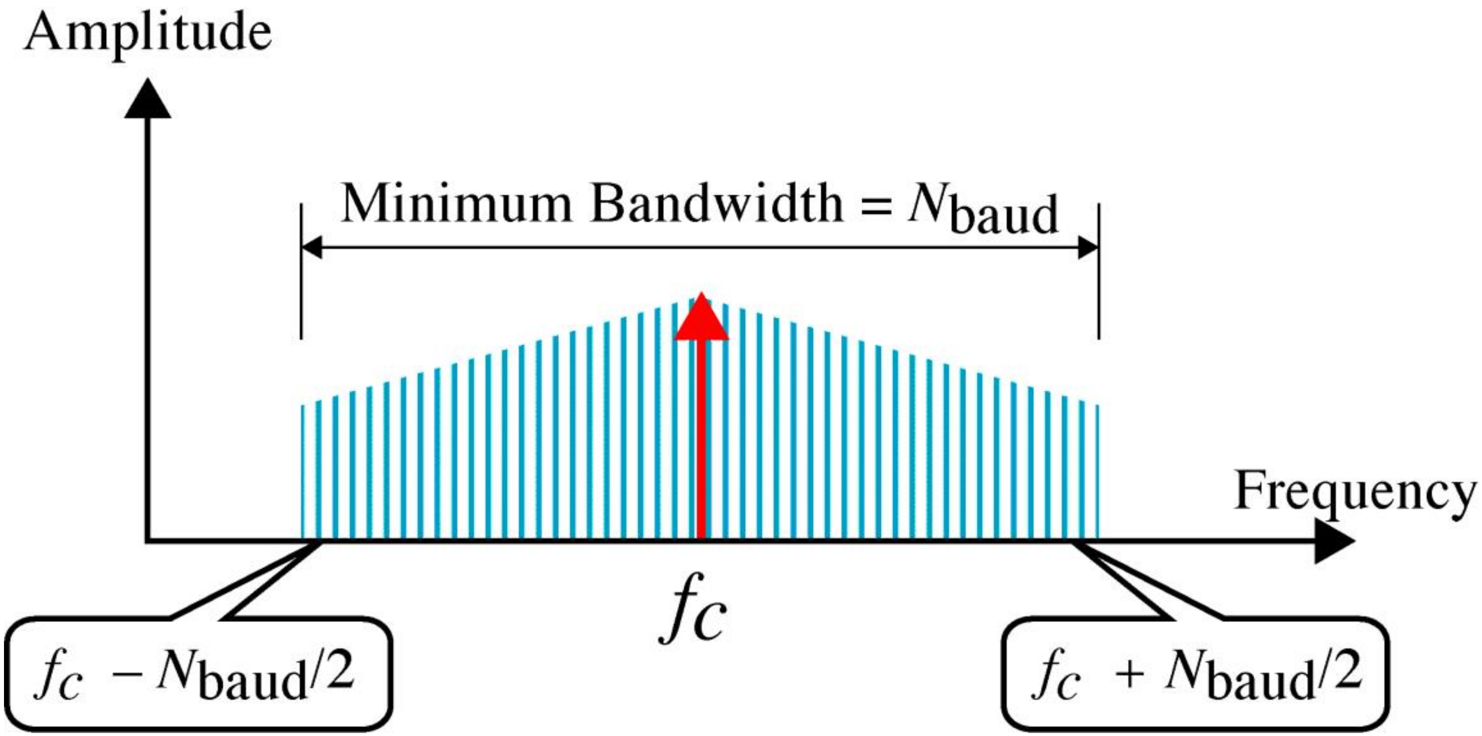
Tribit	Phase
000	0
001	45
010	90
011	135
100	180
101	225
110	270
111	315

Tribits
(3 bits)

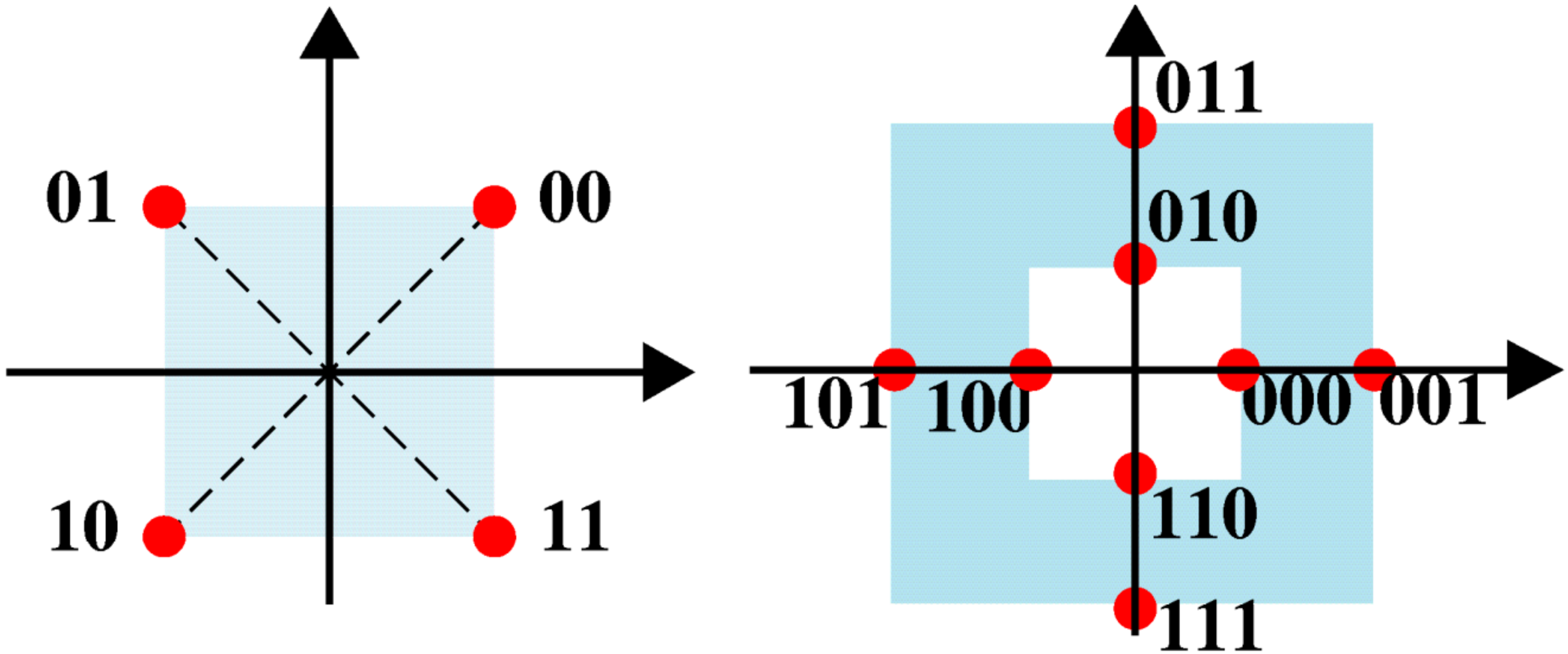


Constellation diagram

PSK Bandwidth



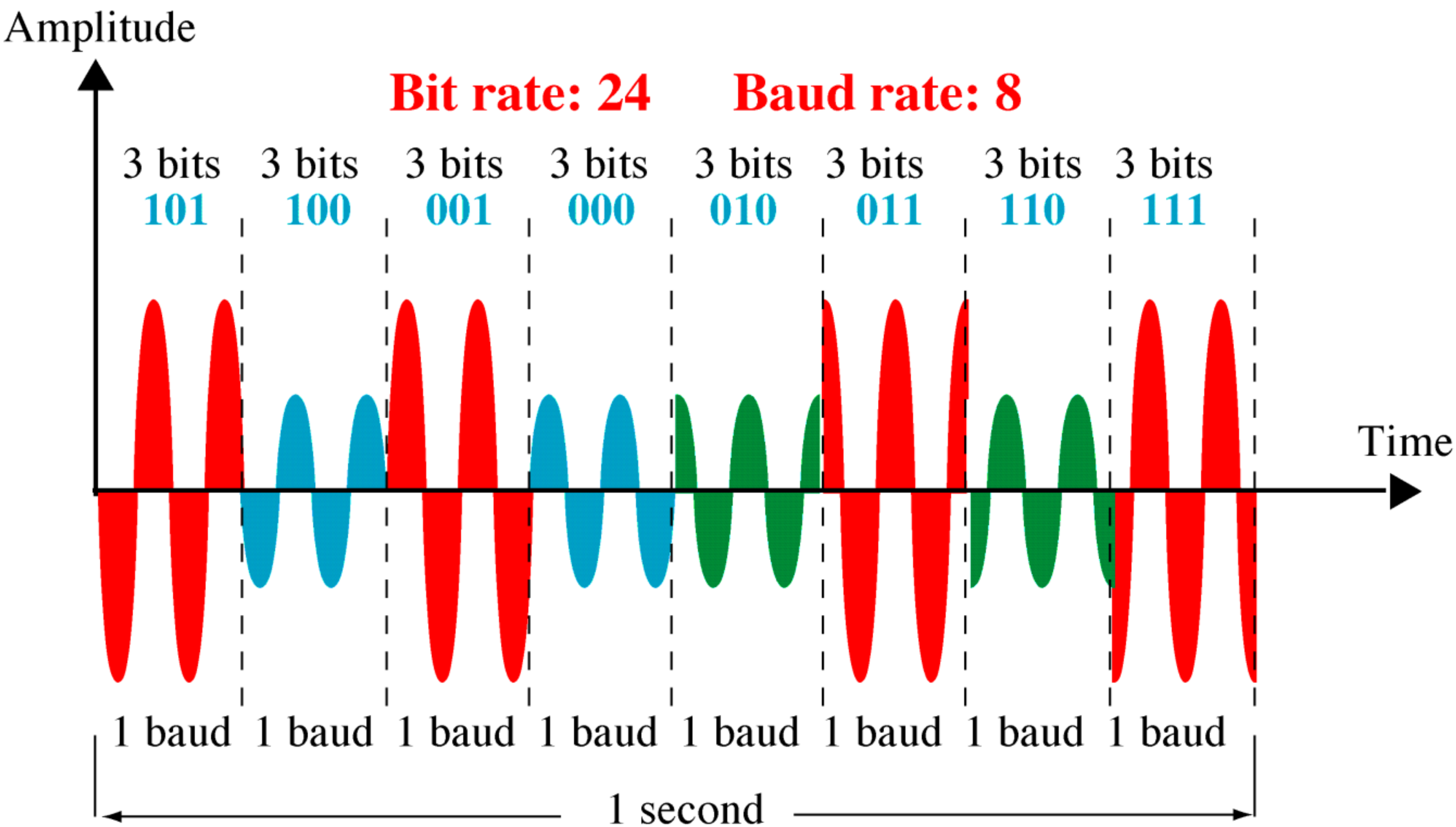
4-QAM and 8-QAM Constellations



4-QAM
1 amplitude, 4 phases

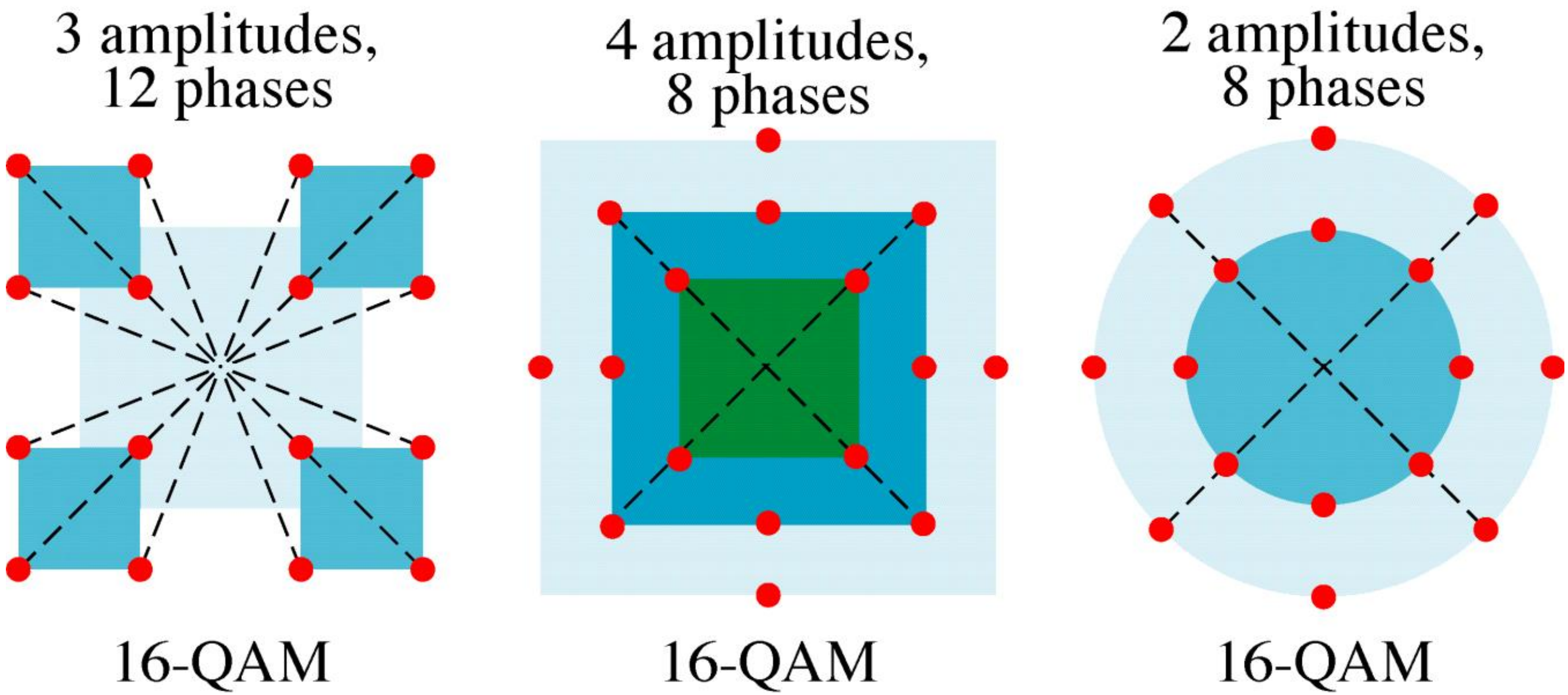
8-QAM
2 amplitudes, 4 phases

8-QAM Signal



Amplitude
Bit rate: 24
Baud rate: 8
3 bits 101
3 bits 100
3 bits 001
3 bits 000
3 bits 010
3 bits 011
3 bits 110
3 bits 111
Time
1 baud
1 baud
1 baud
1 baud
1 baud
1 baud
1 baud
1 baud
1 second

16-QAM Constellation



Quadrature Phase Shift Keying (**QPSK**)modulation.

- QPSK is digital modulation technique.
- It is bandwidth efficient as each signal point represents two bits.
- QPSK uses two separate BPSK modulations; one is in-phase, the other quadrature (out-of-phase).

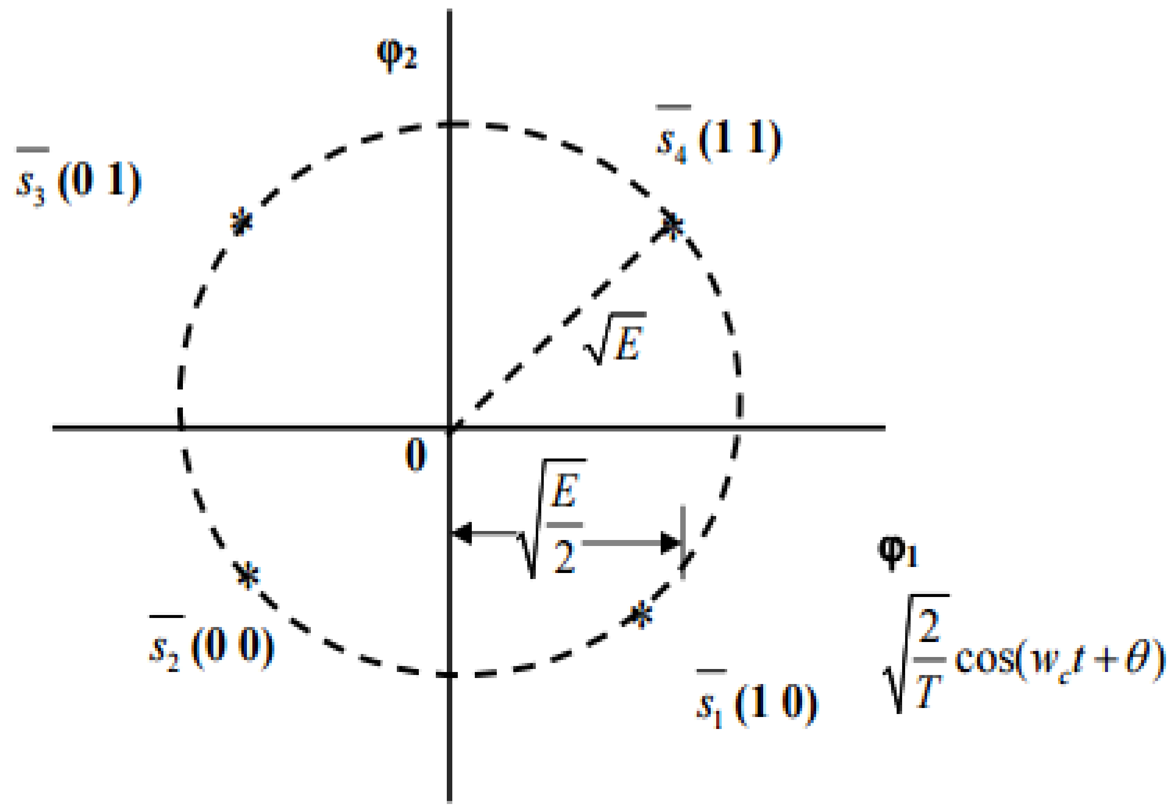


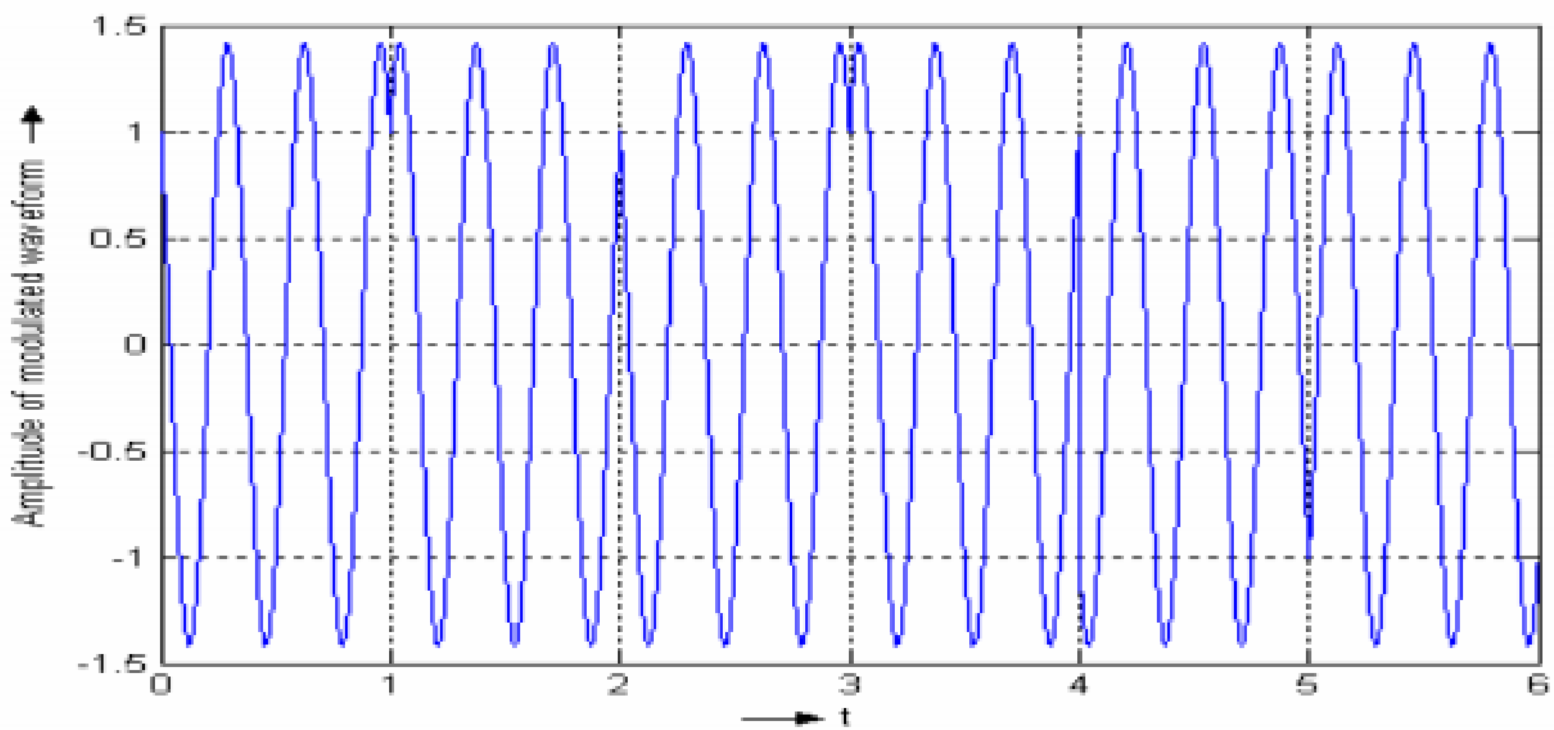
Fig.: *Signal constellation for QPSK. Note that in the above diagram θ has been considered to be zero. Any fixed non-zero initial phase of the basis functions is permissible in general.*

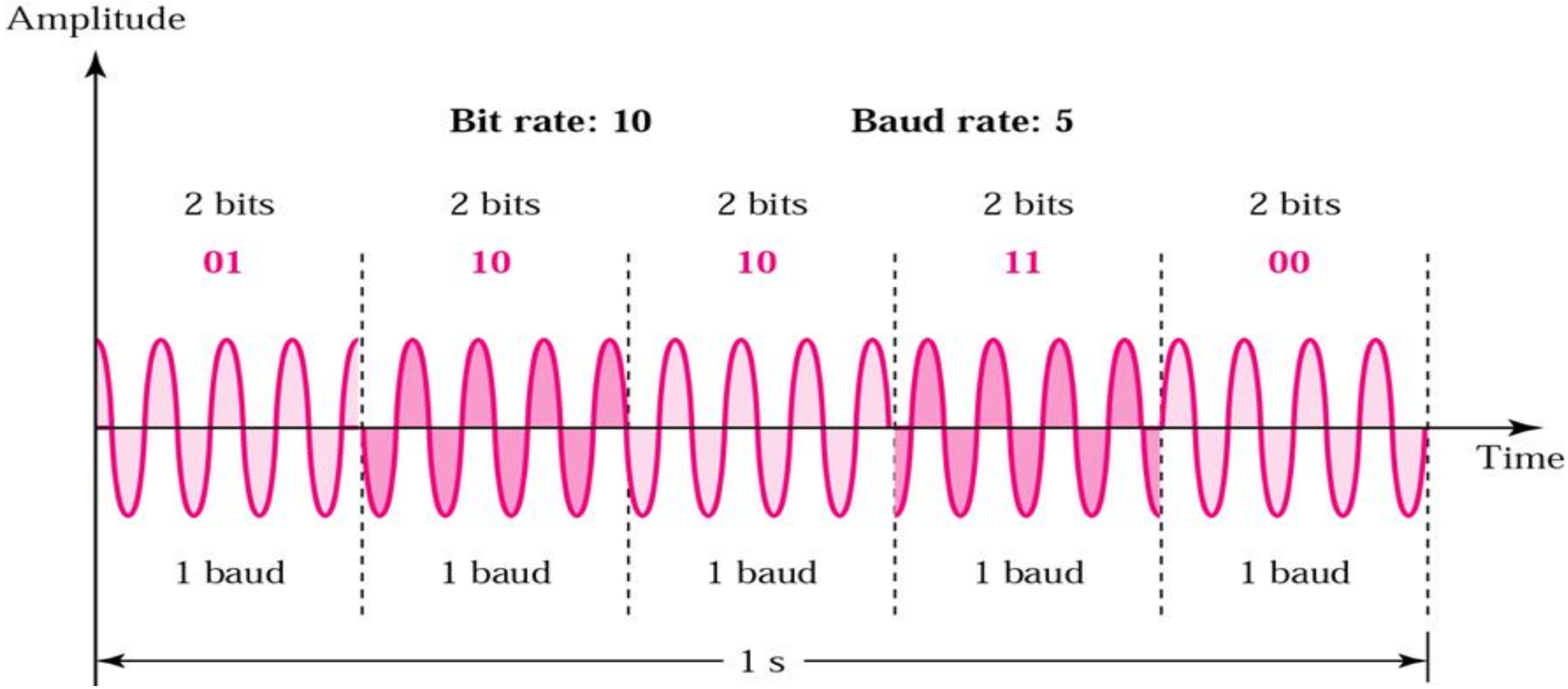
Now, let us consider a random binary data sequence: 10111011000110... Let us designate the bits as 'odd' (b_o) and 'even' (b_e) so that one modulation symbol consists of one odd bit and the adjacent even bit. The above sequence can be split into an odd bit sequence (1111001...) and an even bit sequence (0101010...). In practice, it can be achieved by a 1-to-2 DEMUX. Now, the modulating symbol sequence can be constructed by taking one bit each from the odd and even sequences at a time as {(10), (11), (10), (11), (00), (01), (10), ...}. We started with the odd sequence. Now we can recognize the binary bit stream as a sequence of signal points which are to be transmitted: { $\overline{s}_1$, $\overline{s}_4$, $\overline{s}_1$, $\overline{s}_4$, $\overline{s}_2$, $\overline{s}_3$, $\overline{s}_1$, ...}.

Table summarizes the features of QPSK signal constellation.

Input	Dibit		Phase of QPSK	Coordinates of signal points		
	(b_0)	(b_e)		s_{i1}	s_{i2}	i
$\overline{s_1}$	1	0	$\pi/4$	$+\sqrt{E/2}$	$-\sqrt{E/2}$	1
$\overline{s_2}$	0	0	$3\pi/4$	$-\sqrt{E/2}$	$-\sqrt{E/2}$	2
$\overline{s_3}$	0	1	$5\pi/4$	$-\sqrt{E/2}$	$+\sqrt{E/2}$	3
$\overline{s_4}$	1	1	$7\pi/4$	$+\sqrt{E/2}$	$+\sqrt{E/2}$	4

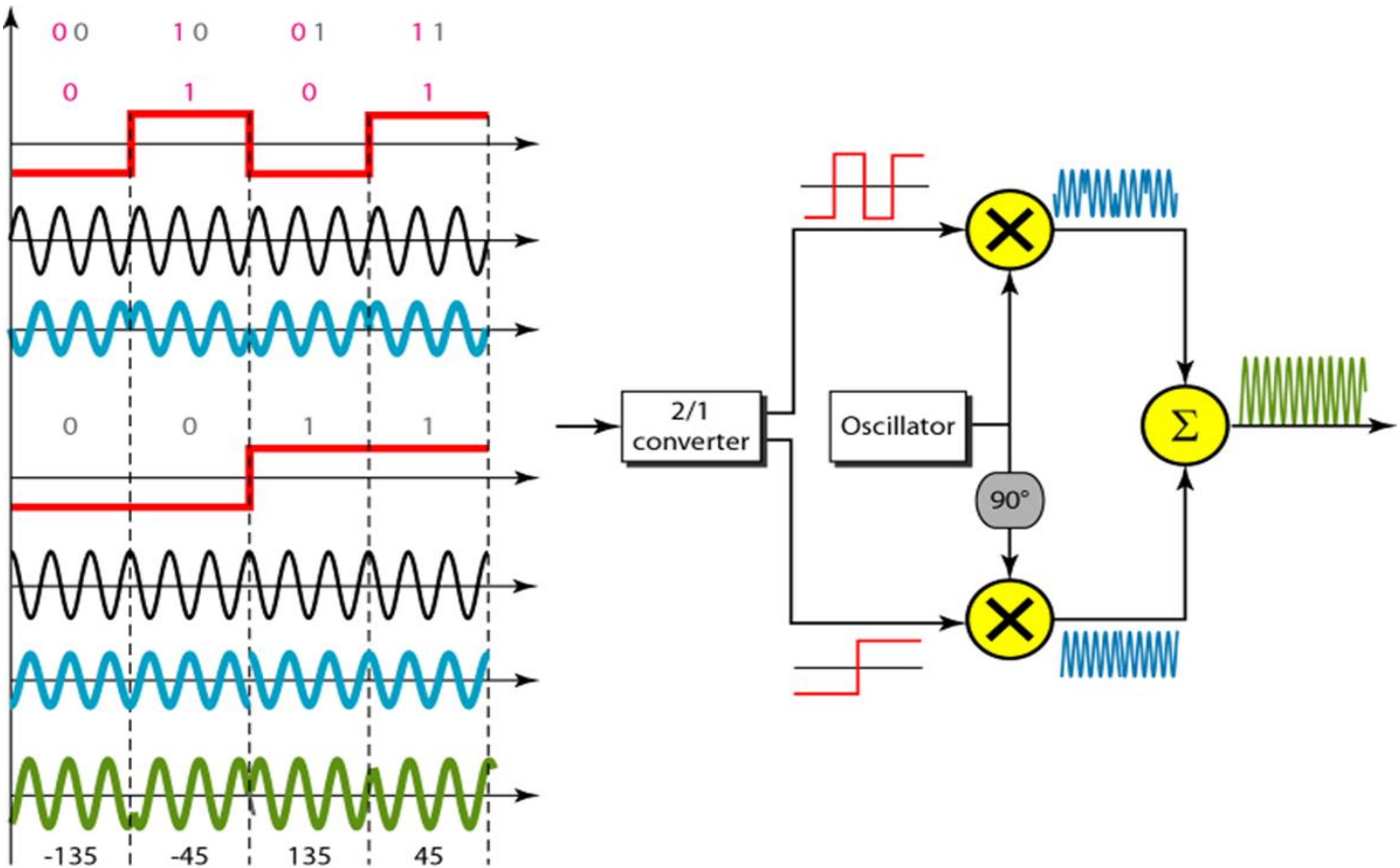
Fig. shows the QPSK modulated waveform for a data sequence 101110110001. For better illustration, only three carrier cycles have been shown per symbol duration.



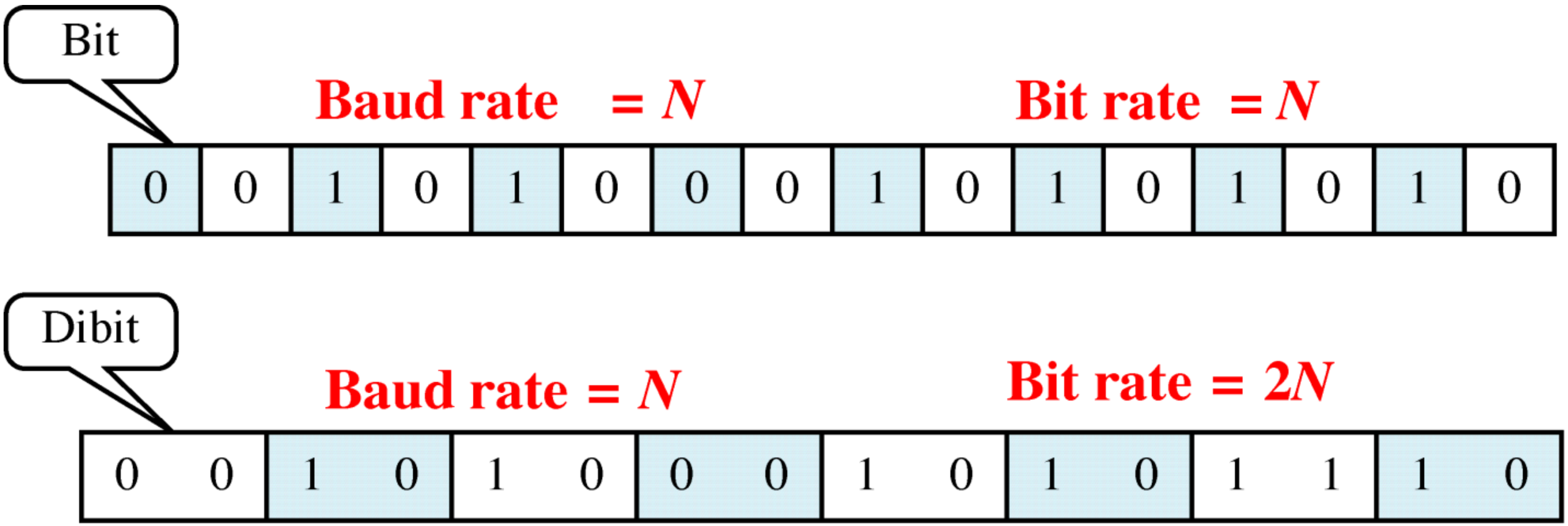


QPSK-- Each signal element carries 2 bits

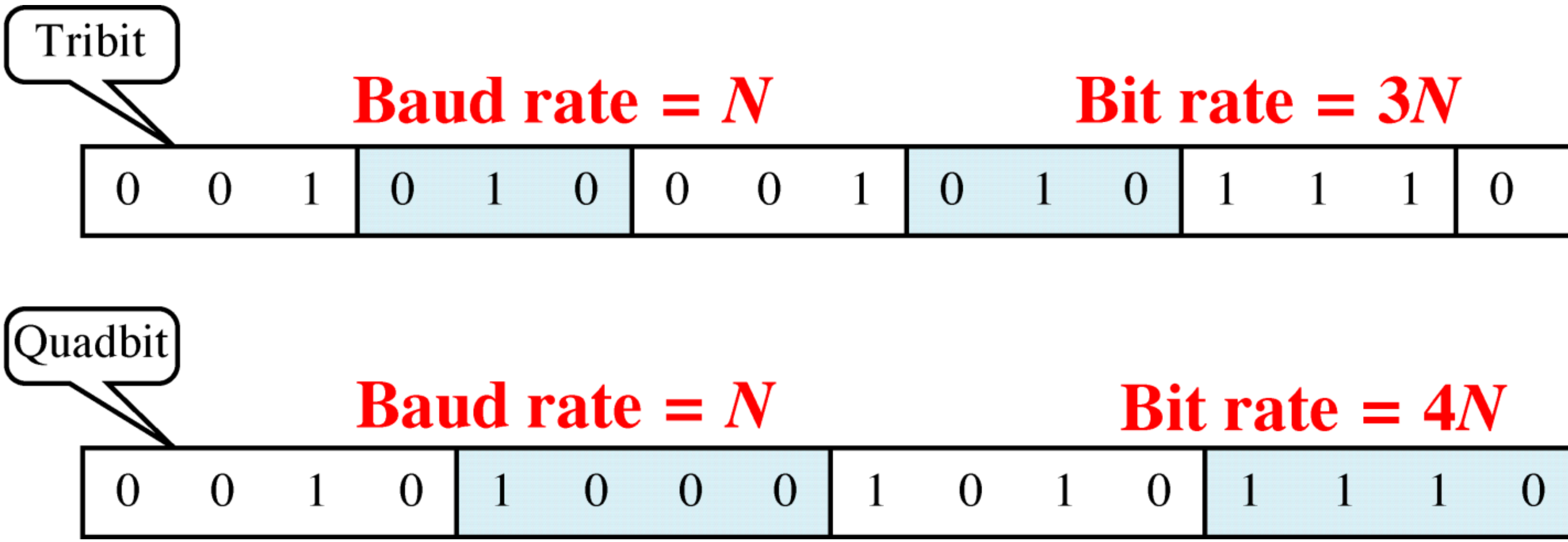
Implementation of QPSK



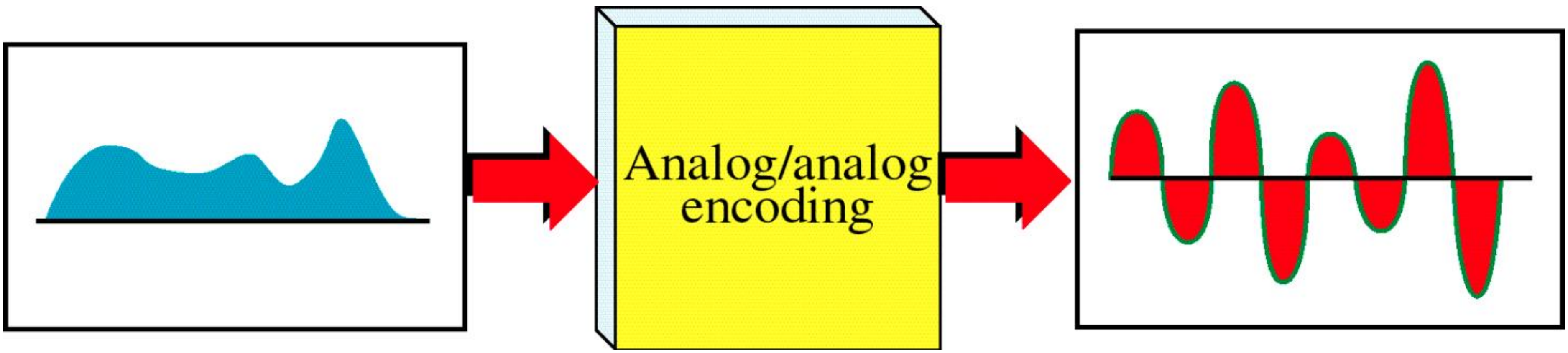
Bit Rate and Baud Rate

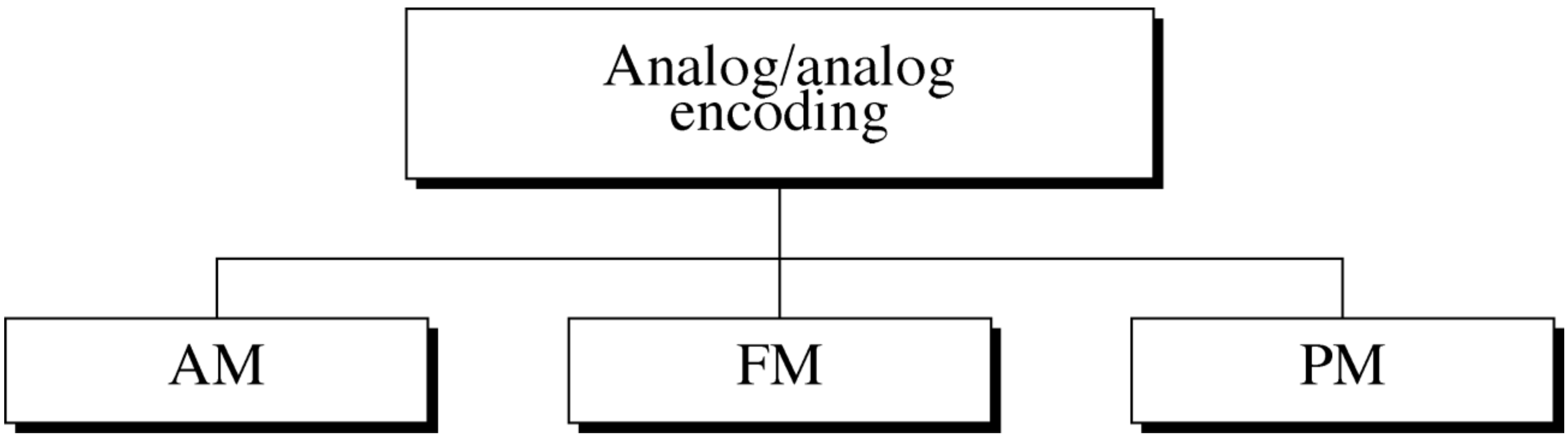


Bit Rate and Baud Rate (contd.)

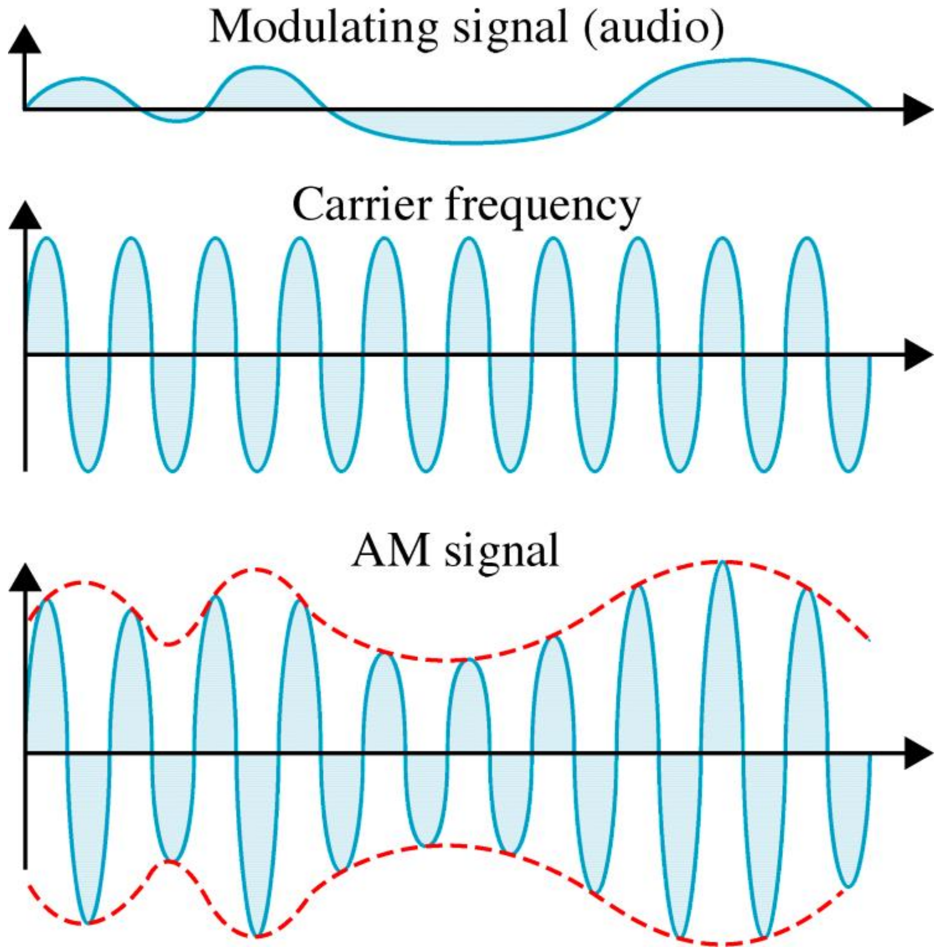


Analog to Analog Modulation


Analog/analog
encoding


Analog/analog encoding
AM
FM
PM

Amplitude Modulation

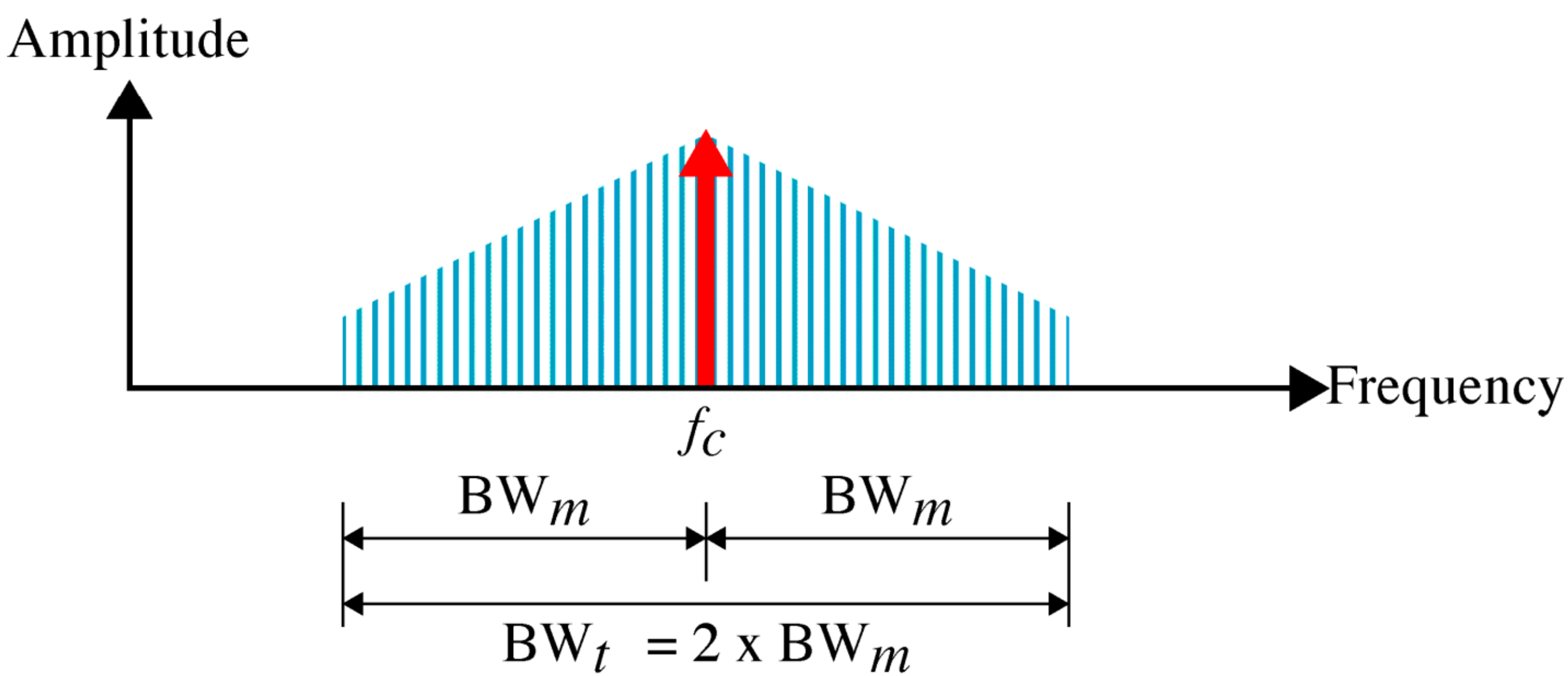


Amplitude Modulation

- 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
- Since on both sides of the carrier freq. fc, the spectrum is identical, we can discard one half, thus requiring a smaller bandwidth for transmission.

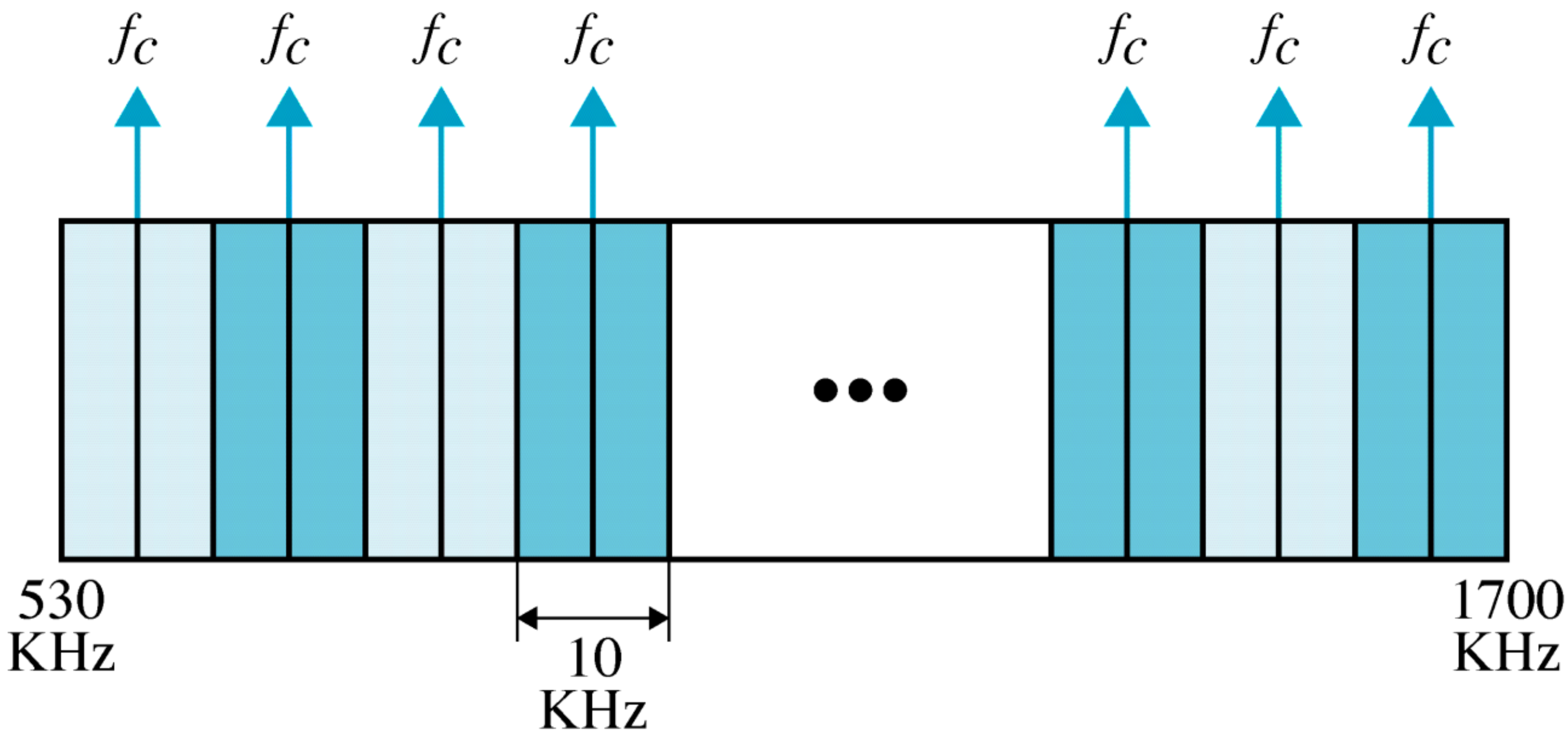
AM Bandwidth

BW_m =Bandwidth of the modulating signal (audio)
BW_t = Total bandwidth (radio)
f_c = Frequency of the carrier

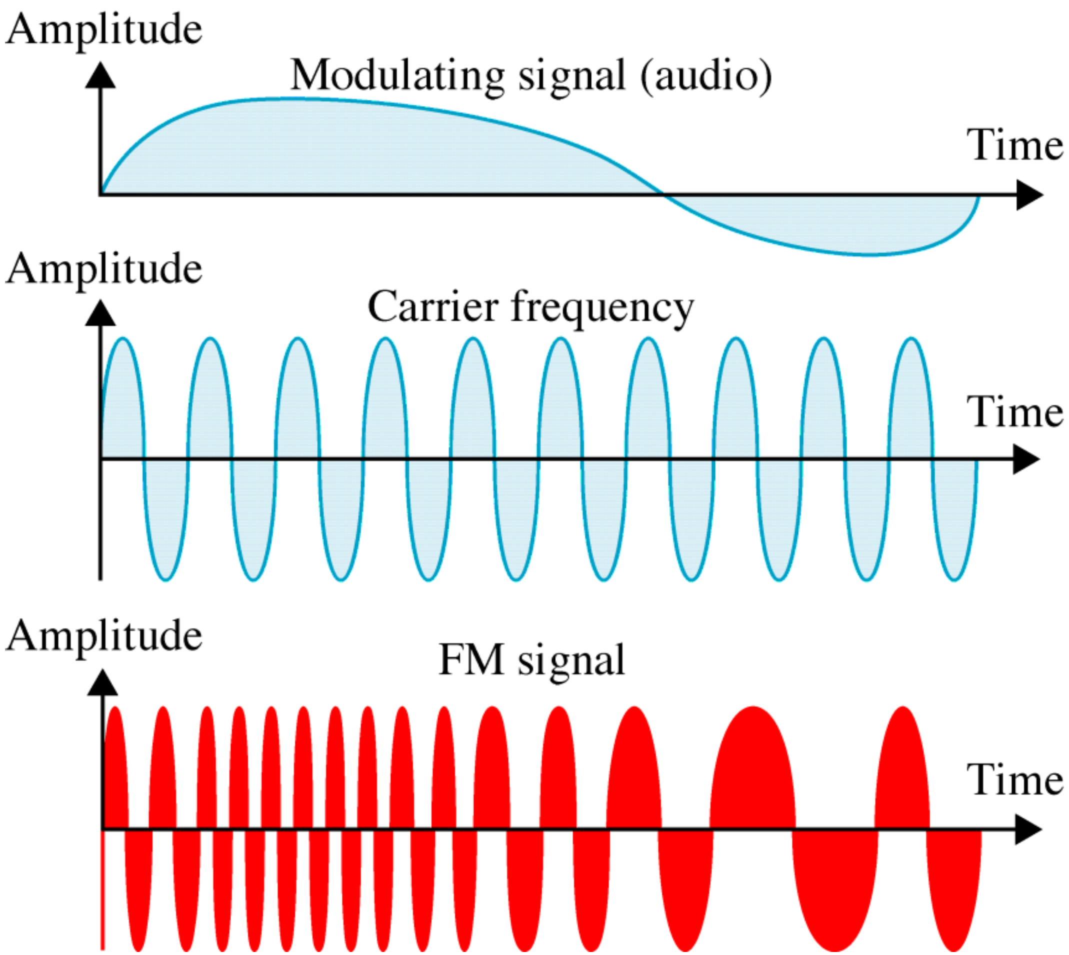


AM Band Allocation

f_c = Carrier frequency of the station



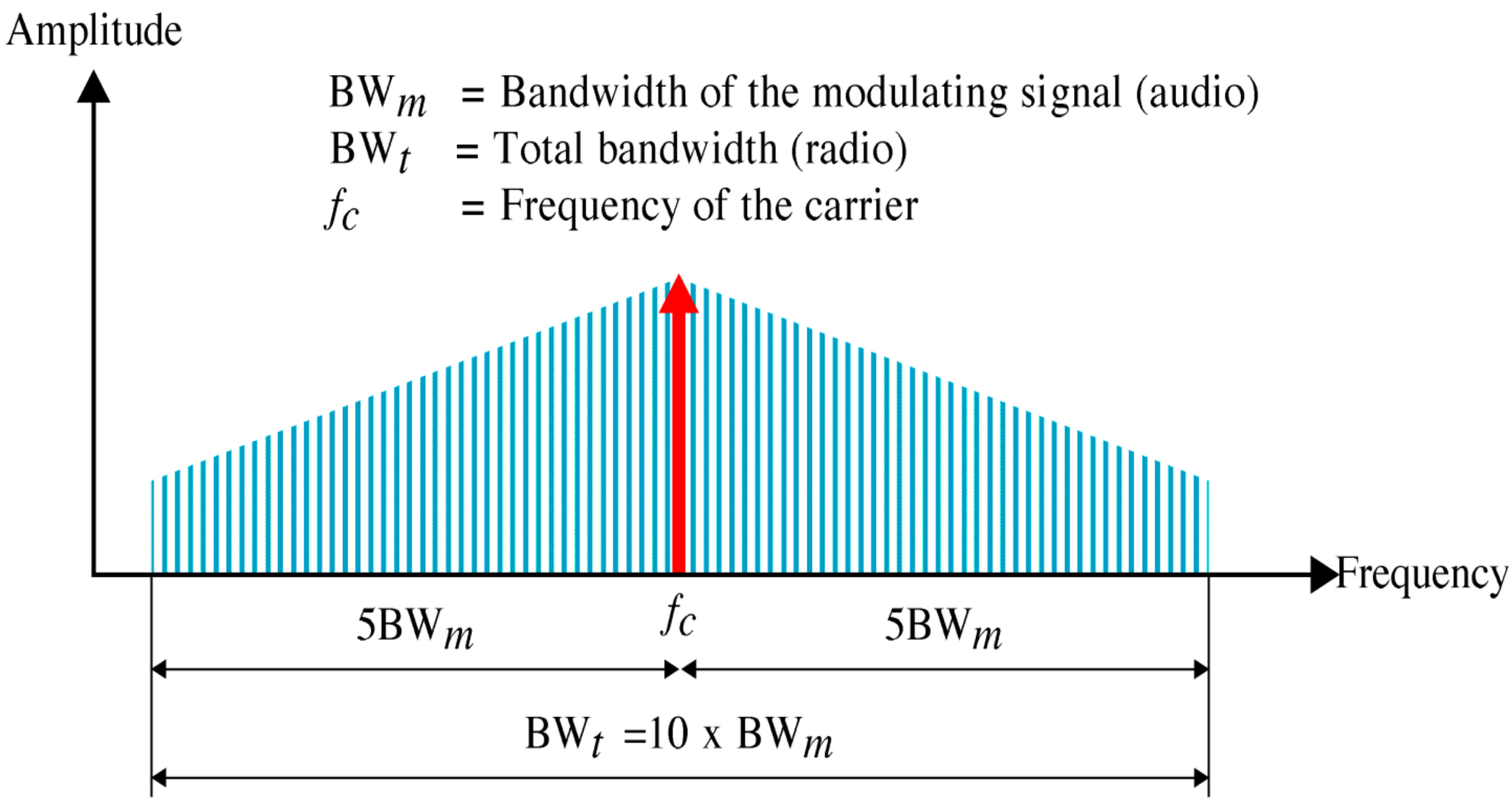
Frequency Modulation



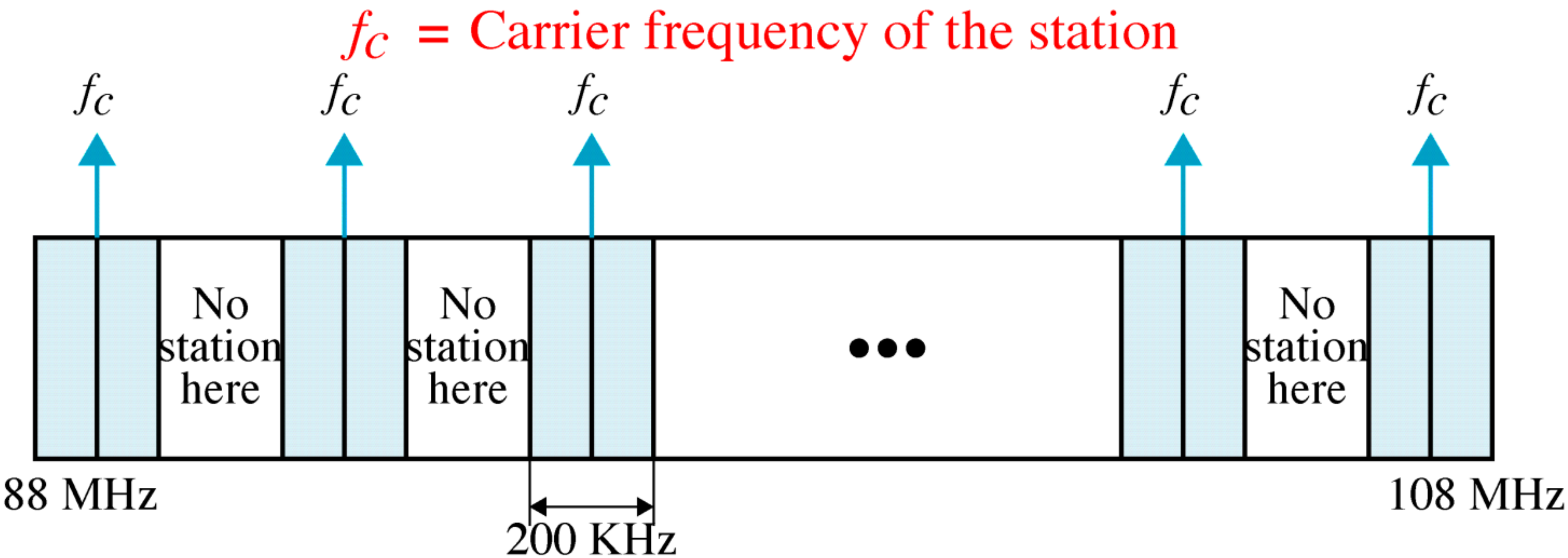
Frequency Modulation

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

FM Bandwidth



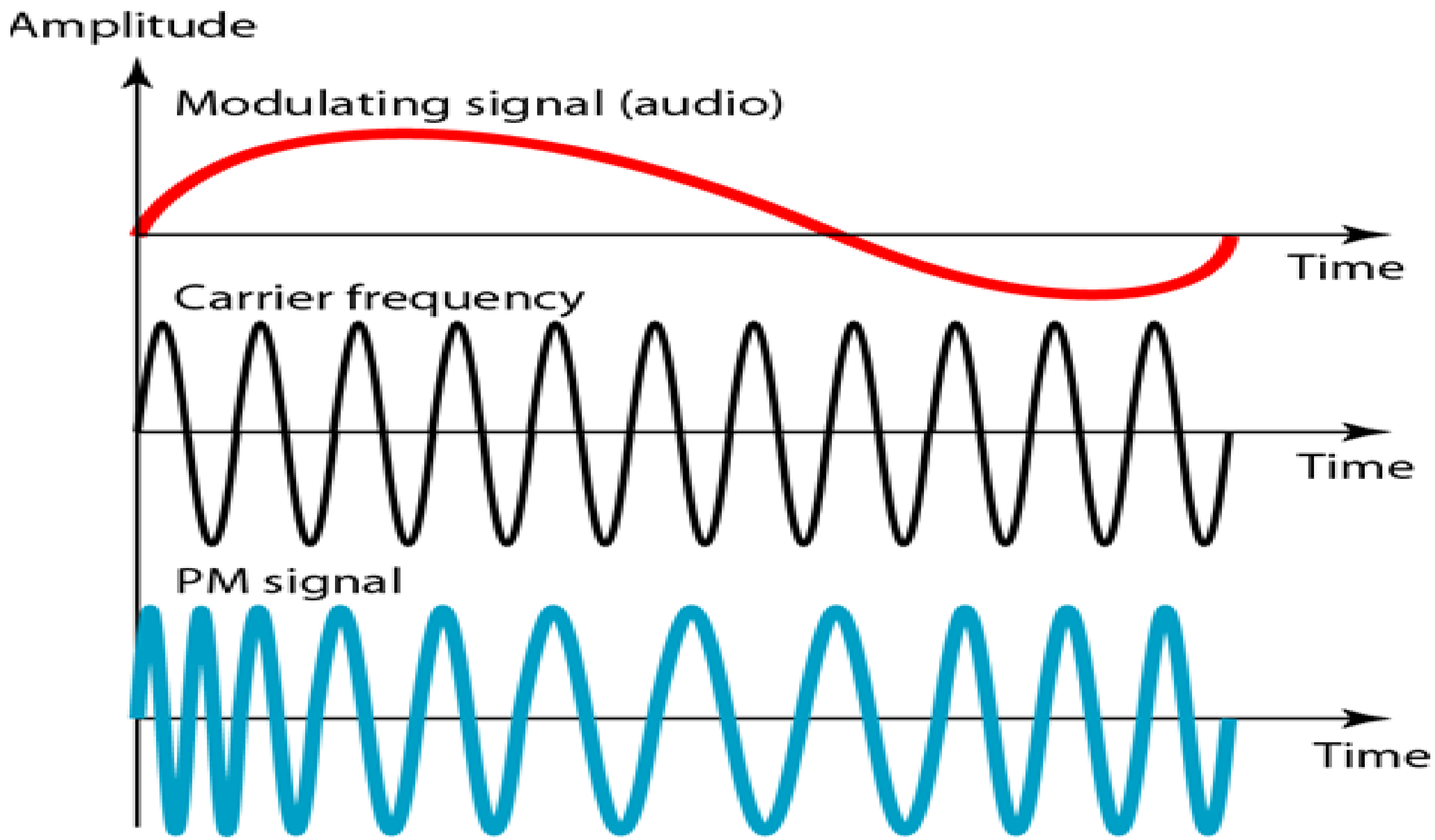
FM Band Allocation



Constellation Diagrams

- Constellation diagram helps us define the amplitude and phase of a signal element, particularly when we are using two carriers (one in-phase and one quadrature).
- In a constellation diagram, a signal element type is represented as a dot. The bit or combination of bits it can carry is often written next to it.
- The diagram has two axes.
- The horizontal X axis is related to the in-phase carrier;
- The vertical Y axis is related to the quadrature carrier.
- For each point on the diagram, four pieces of information can be deduced.
- Projection of the point on the X axis defines the peak amplitude of the in-phase component; the Projection of the point on the Y axis defines the peak amplitude of the quadrature component.
- Length of the line (vector) that connects the point to the origin is the peak amplitude of the signal element (combination of the X and Y components);
- The angle the line makes with the X axis is the phase of the signal element.

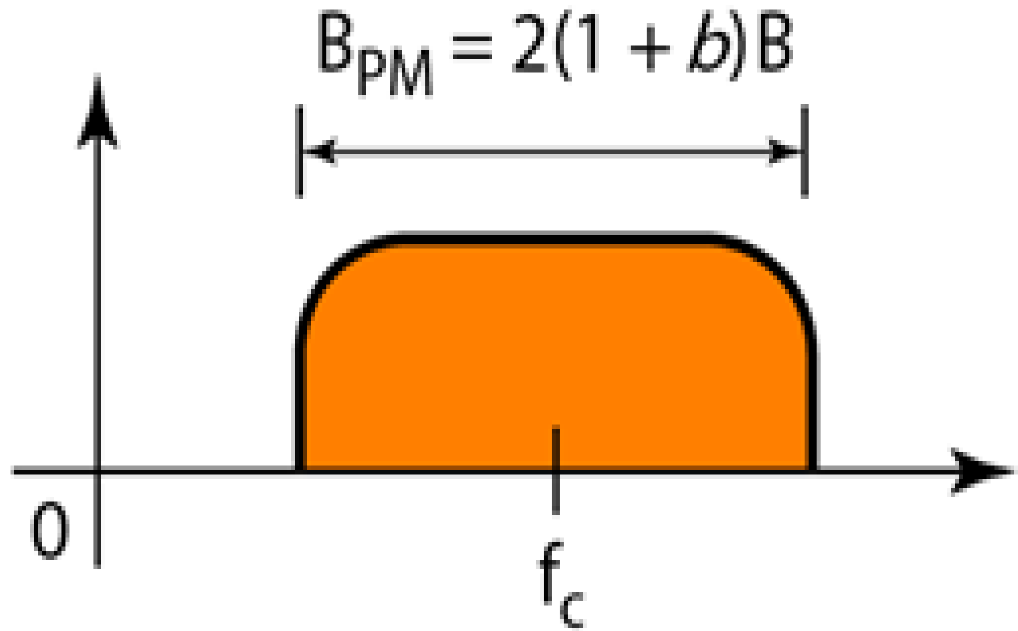
Phase Modulation



Phase Modulation

- 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.

PM Bandwidth



Value of β is lower in the case of PM (around 1 for narrowband and 3 for wideband).

Forms of Phase Modulation

Some of the forms of phase shift keying that are used:

- PM - Phase Modulation
- PSK - Phase Shift Keying
- BPSK - Binary Phase Shift Keying
- QPSK - Quadrature Phase Shift Keying
- 8 PSK - 8 Point Phase Shift Keying
- 16 PSK - 16 Point Phase Shift Keying
- QAM - Quadrature Amplitude Modulation
- 16 QAM - 16 Point Quadrature Amplitude Modulation
- 64 QAM - 64 Point Quadrature Amplitude Modulation
- MSK - Minimum Shift Keying
- GMSK - Gaussian filtered Minimum Shift Keying
- OPSK - Offset Phase Shift Keying

Combine phase shift keying and amplitude keying in a form of modulation known as quadrature amplitude modulation, QAM

End