College of Electronics Engineering

Systems & Control Engineering Department

Digital Communications (SCE3316)

Lectures 1
(Introduction to Digital Communications)
(Analog to Digital Conversion)

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REFERENCES

1.B. P. Lathi and Z. Ding ,"Modern Digital and Analog Communication Systems" fourth edition, published by Oxford University Press Inc., New York, 2010.

2.Leon W. Couch, "Digital and Analog Communication Systems" 8th Edition, Pearson, 2013.

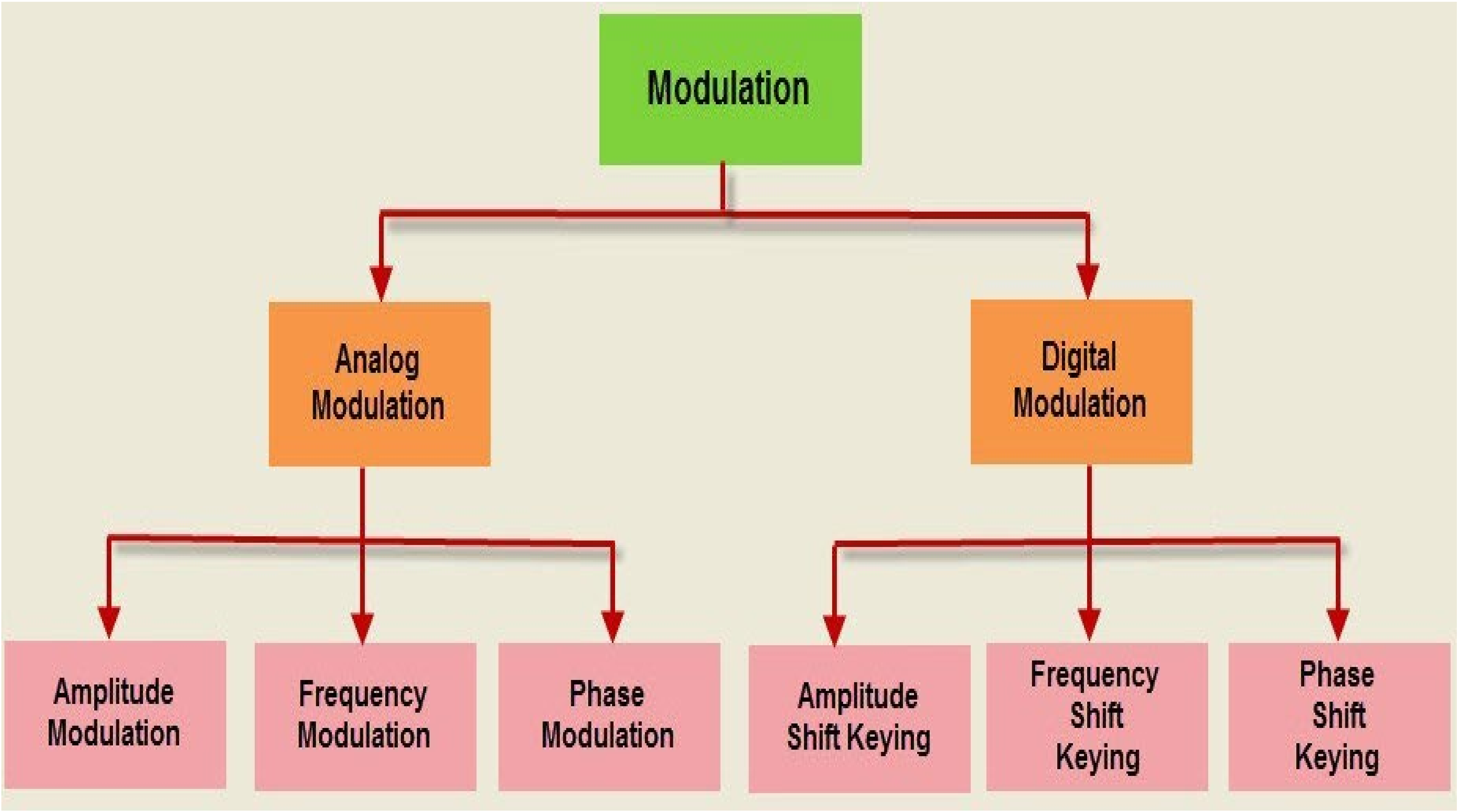
3.Digital Communications by Simon Haykins

4.Digital Communications by Bernard Sklar

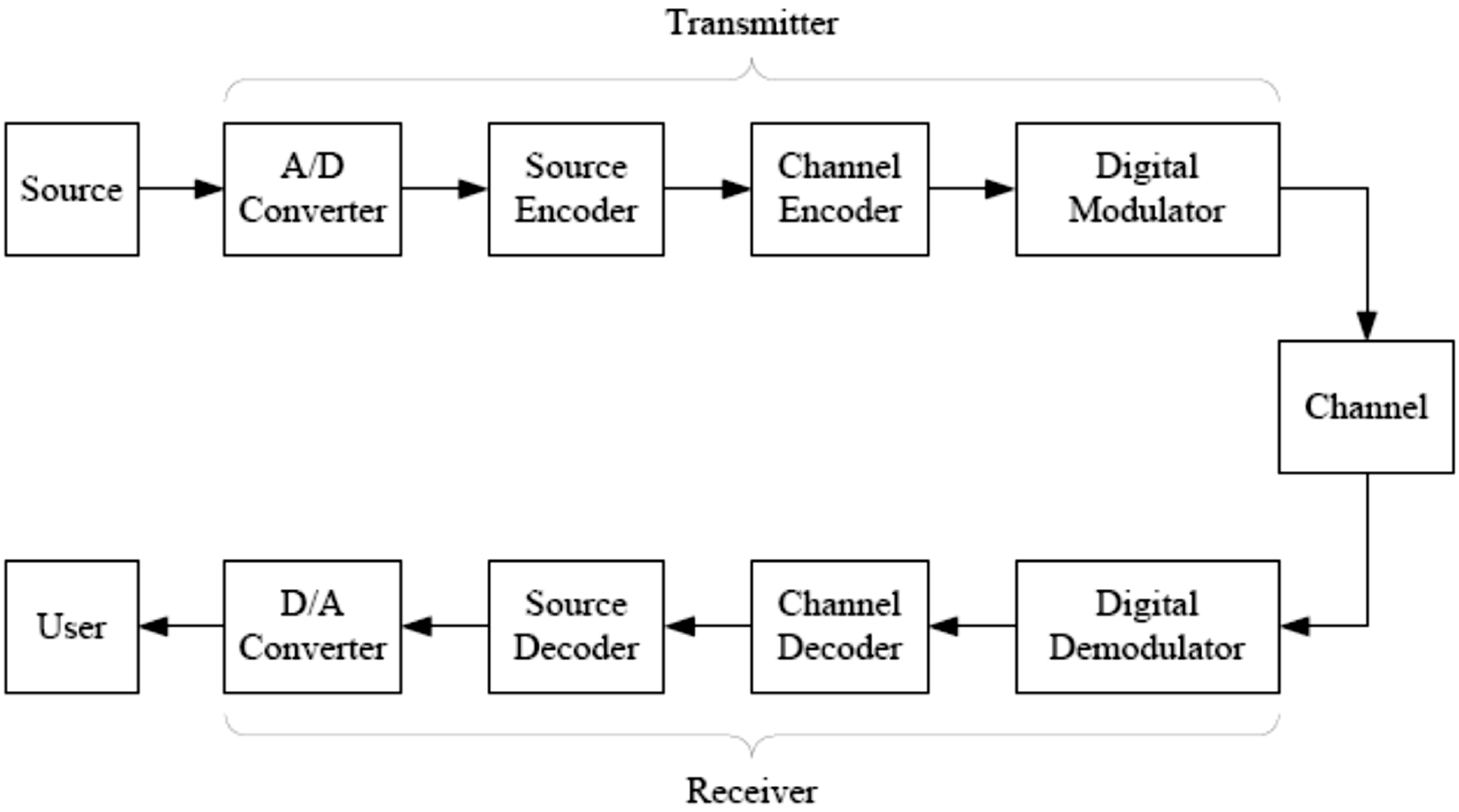
5.Analog & Digital Communications by T L Singal

OVERVIEW OF SYLLABUS

Nyquist sampling theorem, pulse modulation PAM. PWM, PPM, time division multiplexing (TDM), pulse code modulation PCM/TDM. Delta modulation (DM), quantization noise in PCM and DM. signaling format (unipolar, bipolar& split- phase Manchester) sinusoidal digital modulation ASK, PSK, FSK and M-ary. Noise in ASK, PSK FSK (error probability using coherent matched filter and no coherent detection).


Modulation
Analog Modulation
Digital Modulation
Amplitude Modulation
Frequency Modulation
Phase Modulation
Amplitude Shift Keying
Frequency Shift Keying
Phase Shift Keying

A TYPICAL DIGITAL COMMUNICATION LINK



WHY DIGITAL?

The shape of the waveform is affected by two basic mechanisms:

(1) As all transmission lines and circuits have some non-ideal frequency transfer function, there is a distorting effect on the ideal pulse; and

(2) Unwanted electrical noise or other interference further distorts the pulse waveform.

Before it is degraded to an ambiguous state, the pulse is amplified by a digital amplifier that recovers its original ideal shape. The pulse is thus reborn or regenerated. It is done by regenerative repeaters.

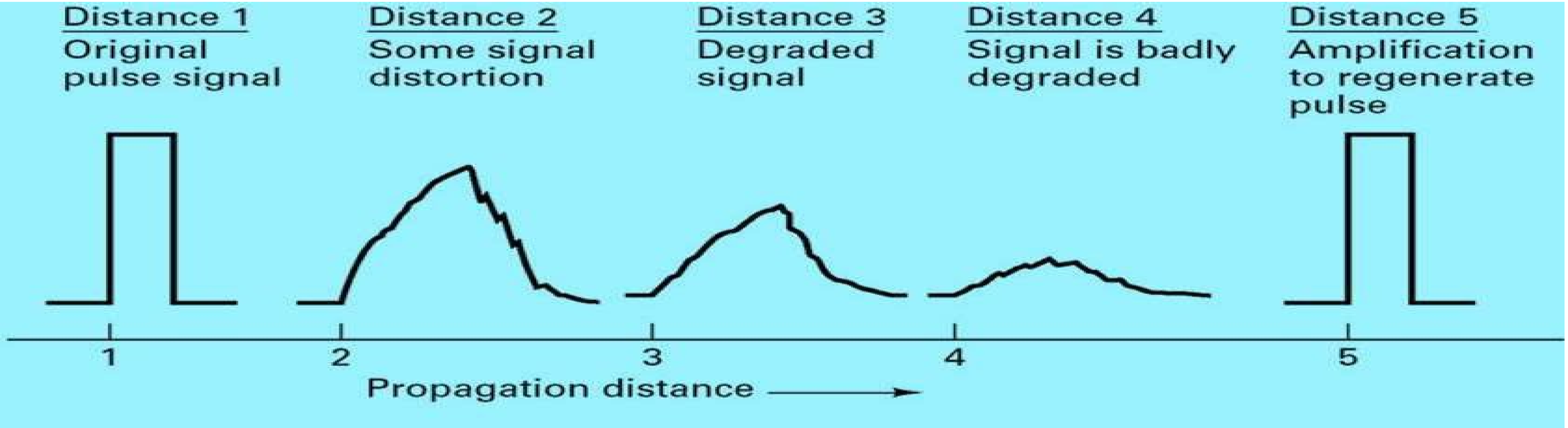


Figure 1.1 Pulse degradation and regeneration.

ADVANTAGES OF DIGITAL TRANSMISSION

- Digital communication, which can withstand channel noise and distortion much better than analog as long as the noise and the distortion are within limits, is more rugged than analog communication. With analog messages, on the other hand, any distortion or noise, no matter how small, will distort the received signal.

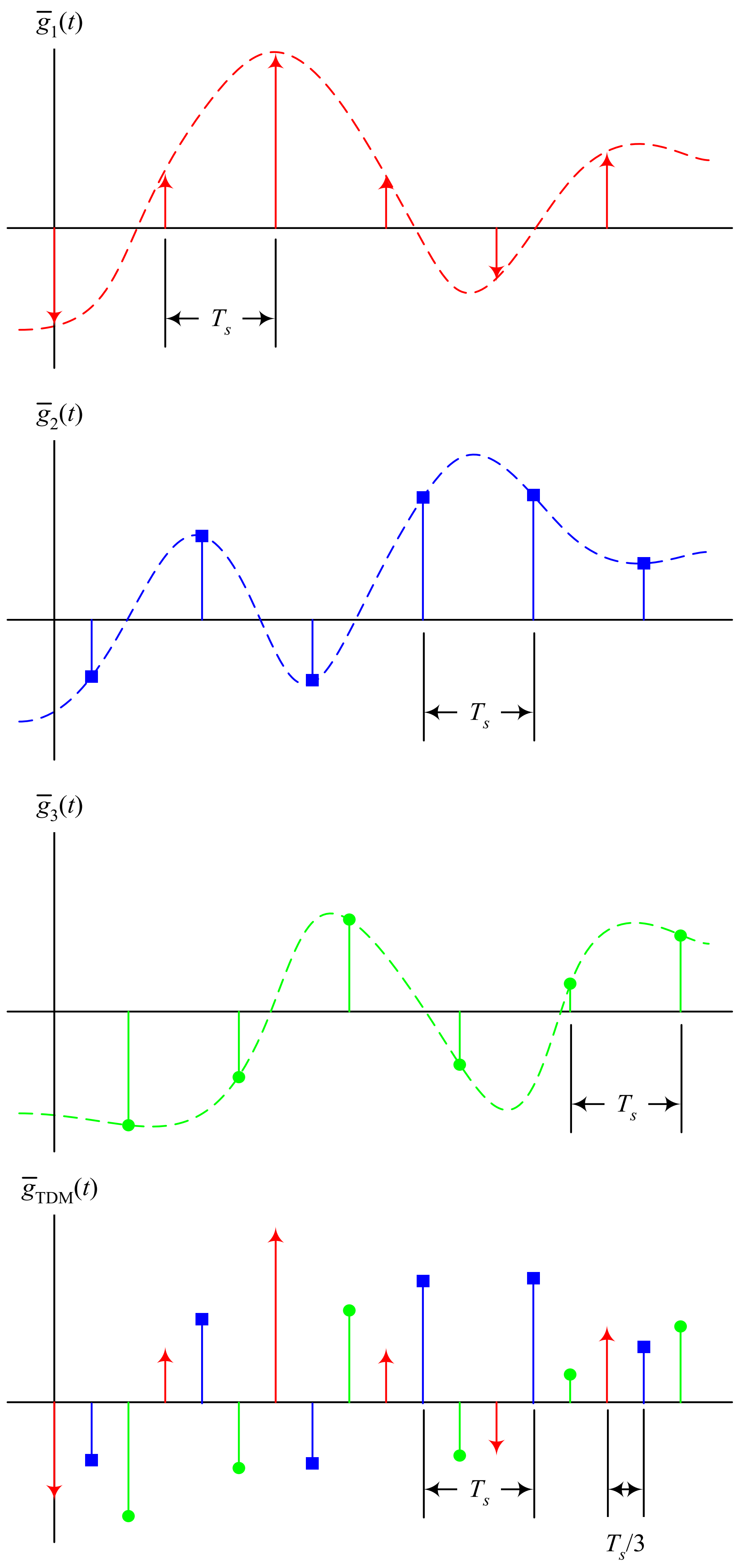
- The greatest advantage of digital over analog communication is the viability of regenerative repeaters in the former. If the noise and distortion are within limits (which is possible because of the closely spaced repeaters), pulses can be detected correctly. This way the digital messages can be transmitted over longer distances with greater reliability. The most significant error in PCM comes from quantizing. This error can be reduced by increasing the number of quantizing levels, the price of which is paid in an increased bandwidth of the transmission medium (channel).

ADVANTAGES OF DIGITAL TRANSMISSION

- Digital hardware implementation is flexible and permits the use of microprocessors, digital switching, and large-scale integrated circuits.
- Data security and privacy -due to encryption
- It is easier and more efficient to multiplex several digital signals. TDM is used.
- Digital communication is inherently more efficient than analog in exchanging SNR for bandwidth.
- Digital signal storage is relatively easy and inexpensive.
- The cost of digital hardware continues to halve every two or three years, while performance or capacity doubles over the same time period.

TIME DIVISION MULTIPLEXING (TDM)

- Multiplexing: The process of sending two or more signals together
  - FDM: Sending them together at the same time over different bands using carrier modulation (AM & FM broadcasting)
  - TDM: Sending them together over the same band by sampling the signals and sending the samples at different time instants (interleaved).



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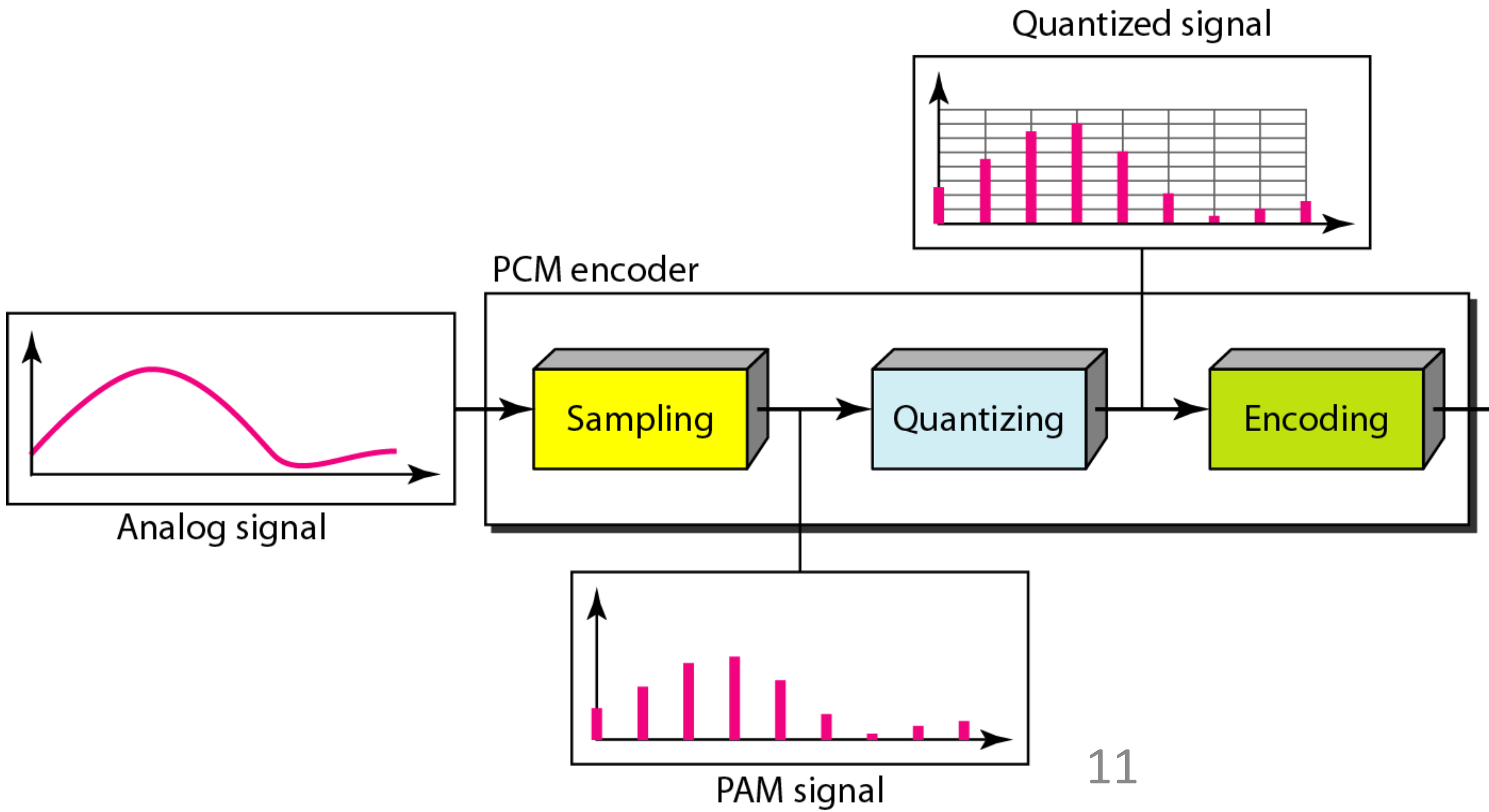
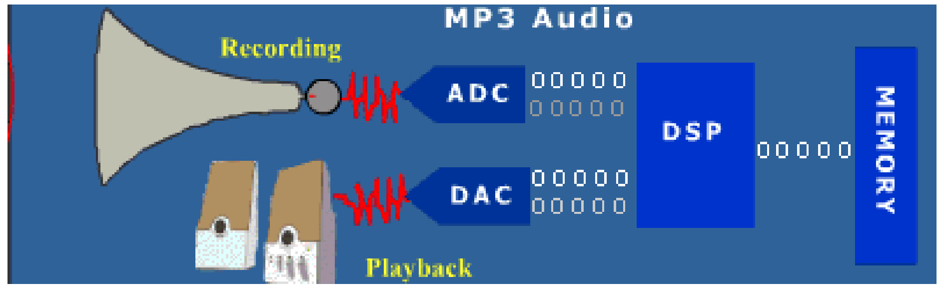
DISADVANTAGES OF DIGITAL TRANSMISSION

- Requires more bandwidth
- Requires precise time synchronization between the clocks used in transmitter and receiver
- Need for additional complex circuitry for encoding and decoding
- Quality of service (QoS) can degrade all of a sudden from very good to very poor if the SNR drops to a specified threshold level

DIGITAL TRANSMISSION

- Digital transmission refers to transmission of digital signals between two or more points in a communication system. If the original signal is in analog form, then it needs to be converted to digital pulses prior to transmission and converted back to analog signals in the receiver.
- The conversion of analog signal to digital pulses is known as waveform coding.
- The digitized signals may be in the form of binary or any other form of discrete level digital pulses.





PULSE CODE MODULATION (PCM)

The most common technique to change an analog signal to digital data (digitization) is called pulse code modulation (PCM).

A PCM encoder has three processes (SQE):

1. The analog signal is sampled.
2. The sampled signal is quantized.
3. The quantized values are encoded as streams of bits.

In converting an analog signal to an equivalent sequence of “0’s” and “1’s”, we go through three processes:

- **Sampling:**
  converting continuous–time analog signals to discrete–time analog signals.

- **Quantization:**
  Converting discrete–time analog signals to discrete–time digital signals (finite set of amplitude levels).

- **Coding**
  Mapping each amplitude level to a binary sequence.

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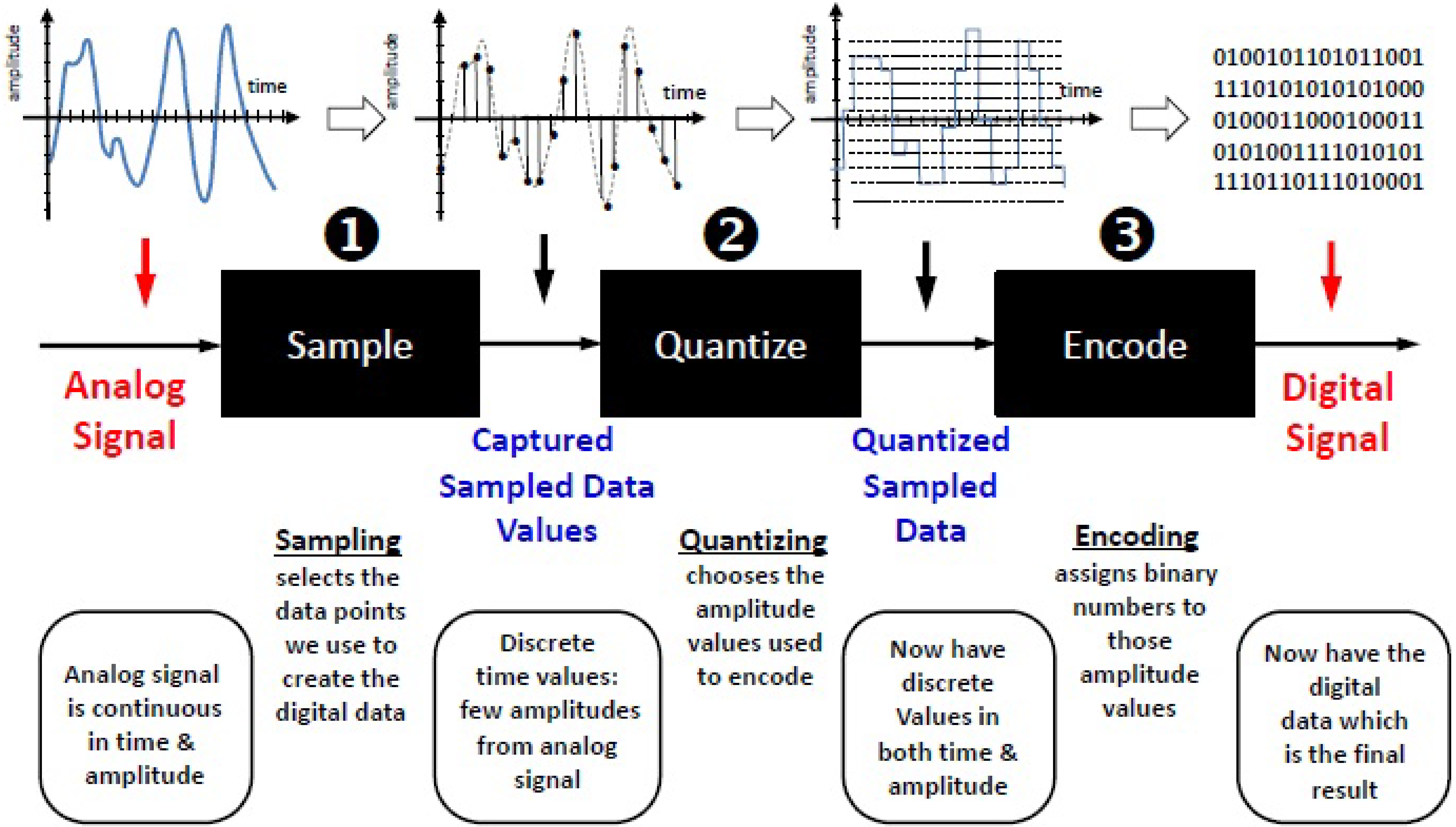
Analog to Digital Conversion Process (ADC)

Three Step Process



Note: "Discrete time" corresponds to the timing of the sampling.

SAMPLING

- **Sampling theorem**: A baseband signal whose spectrum is band-limited to B Hz can be reconstructed exactly (without any error) from its samples taken uniformly at a rate $f_s \geq 2B$.
- $f_s \geq 2B$ is called **Nyquist Criterion of sampling**.
- $f_s = 2B$ is called the **Nyquist rate of sampling**.
- $T_s = 1/f_s$ is called the **Nyquist interval**.
- Note that the larger the transmission rate (pulses/sec) the narrower the pulse, the wider its spectrum, the higher the channel bandwidth required for transmission.

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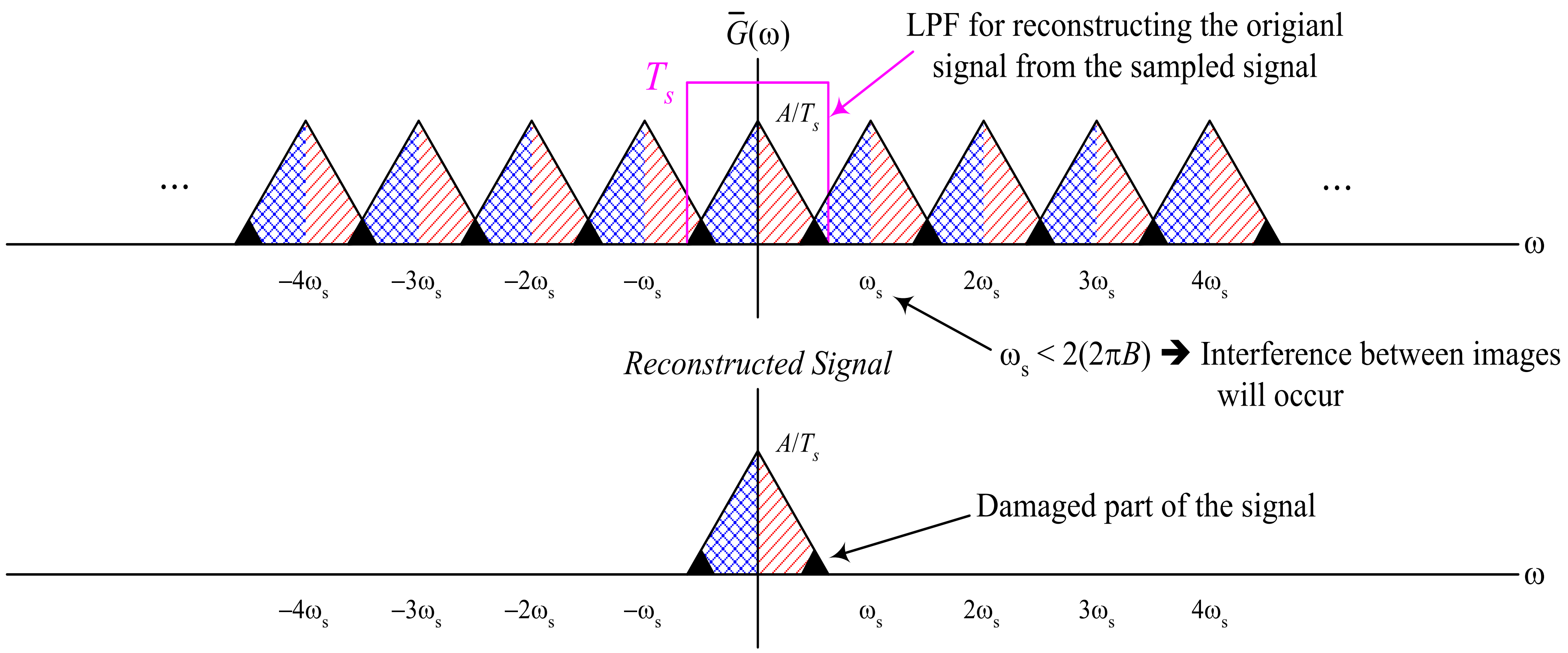
ALIASING



- **Sampling a signal at a rate less than the Nyquist rate results in Aliasing.**
- **In aliasing, the higher frequency components take the identity of lower frequencies.**
- **Real life example: Sampling a rotating wheel.**

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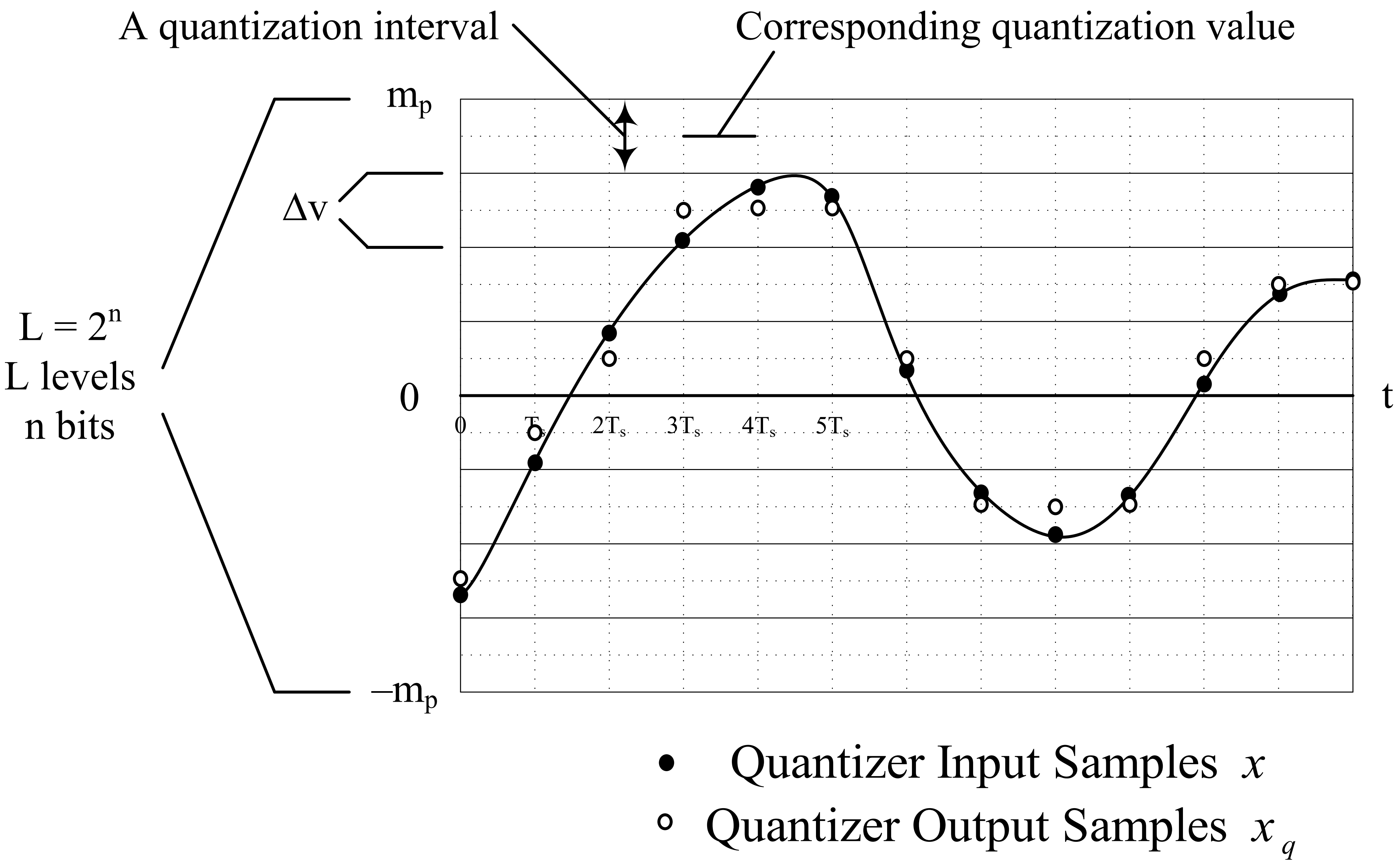
QUANTIZATION

- Analog samples with an amplitude that may take value in a specific range are converted to digital samples with an amplitude that takes one of a specific pre–defined set of values.
- The range of possible values of the analog samples is divided into L levels. L is usually taken to be a power of 2 (L = 2^n).
- The center value of each level is assigned to any sample that falls in that quantization interval.
- There are two sources of error in this scheme: quantization error and pulse detection error. In almost all practical schemes, the pulse detection error is quite small compared to the quantization error and can be ignored. In the present analysis, therefore, we shall assume that the error in the received signal is caused exclusively by quantization.

QUANTIZATION



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QUANTIZATION

Assuming that the error is equally likely to lie anywhere in the range $(-\Delta v/2, \; \Delta v/2)$, the mean square quantizing error $\widetilde{q^2}$ is given by*

$$\widetilde{q^2} = \frac{1}{\Delta v} \int_{-\Delta v/2}^{\Delta v/2} q^2 \, dq$$

$$= \frac{(\Delta v)^2}{12}$$

$$= \frac{m_p^2}{3L^2}$$

Because $\widetilde{q^2(t)}$ is the mean square value or power of the quantization noise, we shall denote it by N_q,

$$N_q = \widetilde{q^2(t)} = \frac{m_p^2}{3L^2}$$

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QUANTIZATION

Assuming that the pulse detection error at the receiver is negligible, the reconstructed signal $\hat{m}(t)$ at the receiver output is

$$\hat{m}(t) = m(t) + q(t)$$

The desired signal at the output is $m(t)$, and the (quantization) noise is $q(t)$. Since the power of the message signal $m(t)$ is $\widetilde{m^2(t)}$, then

$$S_o = \widetilde{m^2(t)}$$

$$N_o = N_q = \frac{m_p^2}{3L^2}$$

and

$$\frac{S_o}{N_o} = 3L^2 \frac{\widetilde{m^2(t)}}{m_p^2}$$

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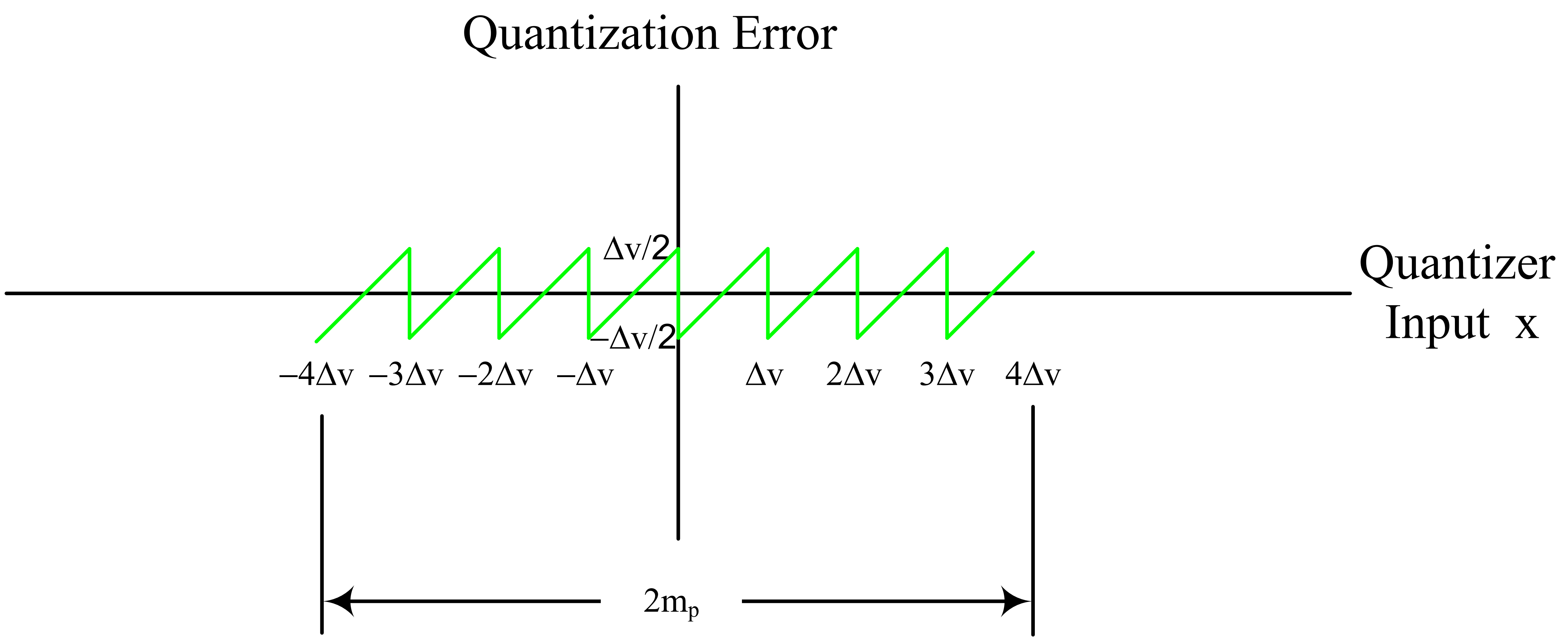
QUANTIZATION



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QUANTIZATION

Steps in quantization

1. We assume that the original analog signal has instantaneous amplitudes between V_{min} and V_{max}.
2. We divide the range into L zones, each of height Δ (delta).

$$\Delta = (V_{max} - V_{min}) / L \quad \text{or} \quad \Delta = 2m_p / L$$

3. We assign quantized values of 0 to L - 1 to the midpoint of each zone.
4. We approximate the value of the sample amplitude to the quantized values.

Example: Assume that we have a sampled signal and the sample amplitudes between -20 and +20 V. We decide to have eight levels (L = 8). What will be Δv value? The figures in the next slides show the details.

Δv= 5

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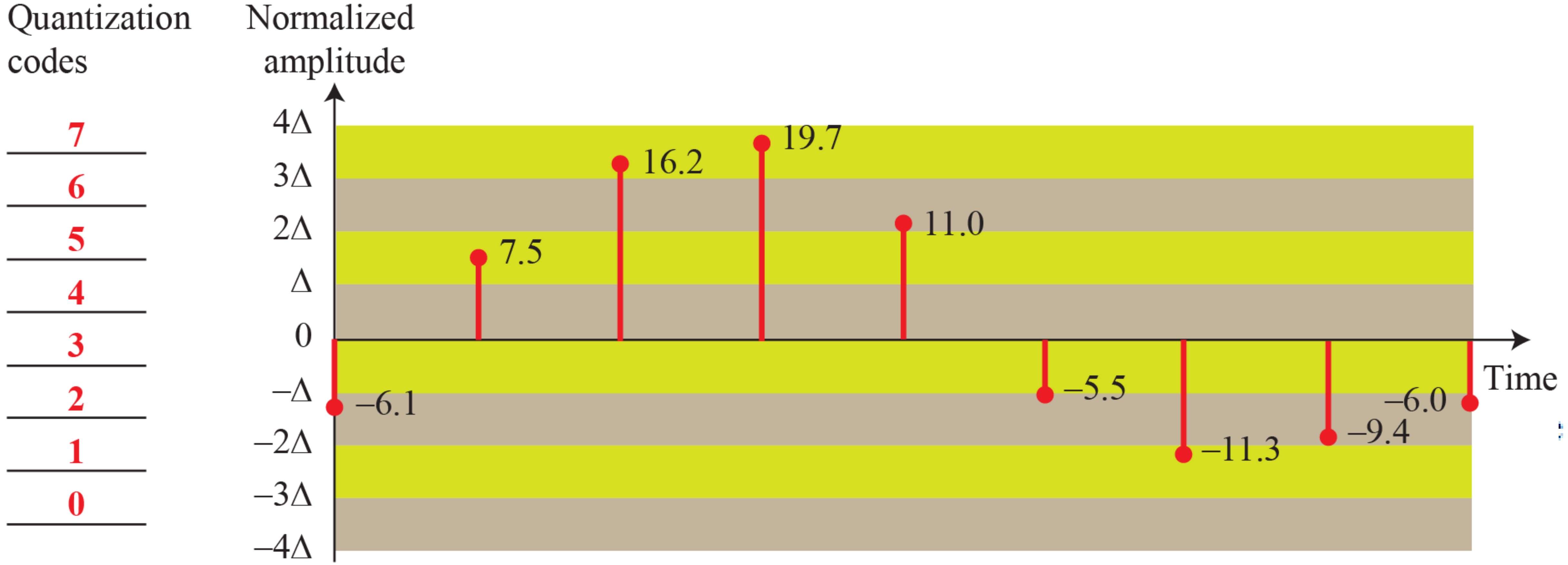

Normalized PAM values	–1.22	1.50	3.24	3.94	2.20	–1.10	–2.26	–1.88	–1.20
Normalized quantized values	–1.50	1.50	3.50	3.50	2.50	–1.50	–2.50	–1.50	–1.50
Normalized error	–0.28	0	+0.26	–0.44	+0.30	–0.40	–0.24	+0.38	–0.30
Quantization code	**2**	**5**	**7**	**7**	**6**	**2**	**1**	**2**	**2**
Encoded words	**010**	**101**	**111**	**111**	**110**	**010**	**001**	**010**	**010**

The value at the top of each sample in the graph shows the actual amplitude. The first row is the normalized value for each sample (actual amplitude/Δ). The quantization process selects the quantization value from the middle of each zone. This means that the normalized quantized values (second row) are different from the normalized amplitudes. The difference is called the *normalized error* (third row). The fourth row is the quantization code for each sample based on the quantization levels at the left of the graph. The encoded words (fifth row) are the final products of the conversion.

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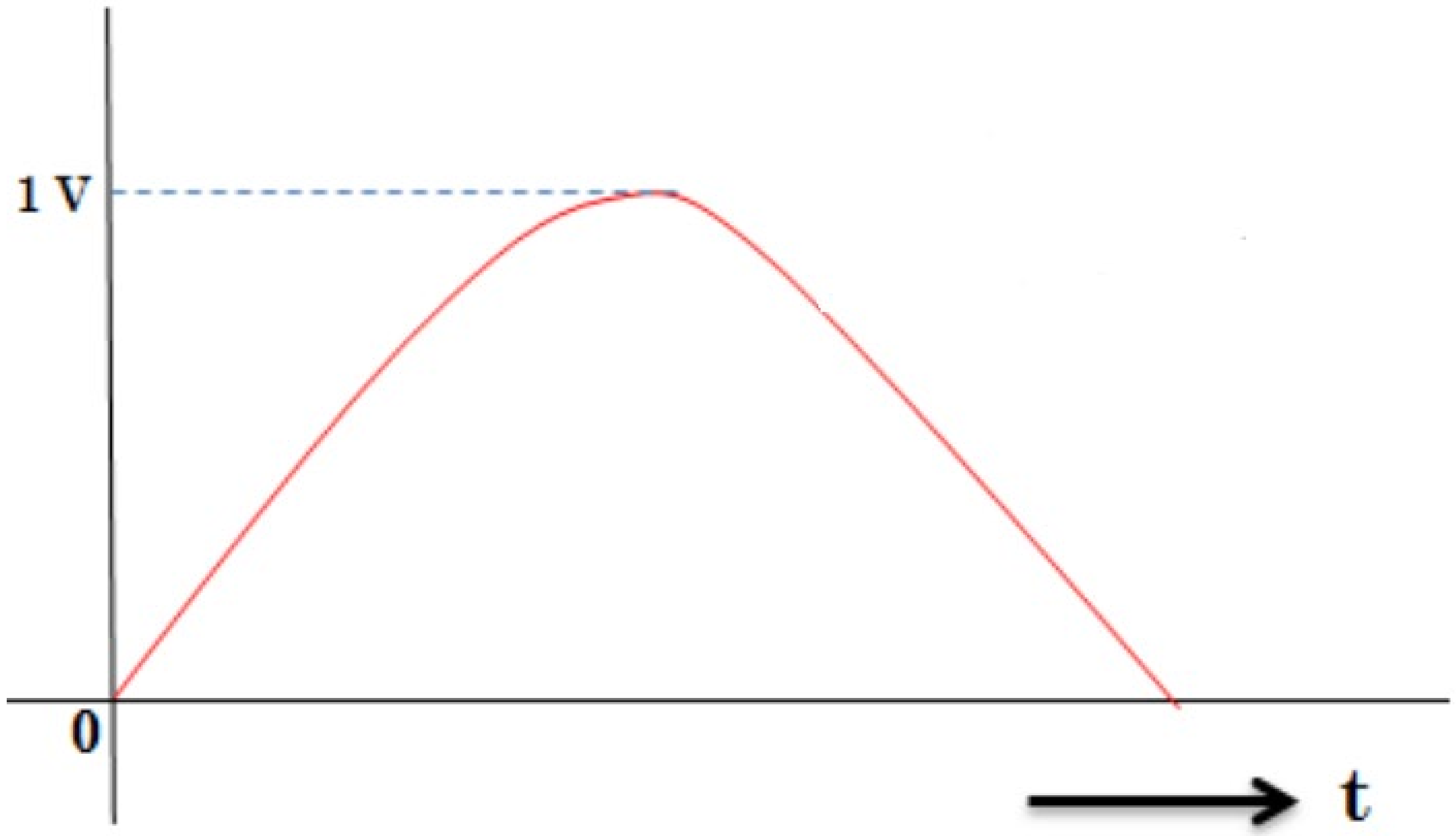
Example: Compute the binary code word values for the amplitudes 0.4 V and 0.78 V, then compute the quantization error for each. Suppose you have a 2bit ADC (PCM) modulator with 0 to 1 volt amplitude signal as shown below:



QUANTIZATION LEVELS:

- The choice of L, the number of levels, depends on the range of amplitudes of the analog signal and how accurately we need to recover the signal.
- If the amplitude of a signal fluctuates between two values only, we need only two levels.
- If the signal, like voice, has many amplitude values, we need more quantization levels.
  - In audio digitizing, L is normally chosen to be 256.
  - In video it is normally thousands.

- Choosing lower values of L increases the quantization error if there is a lot of fluctuation in the signal.

Increasing the number of levels increases the SNR. HOW??

- If the number of quantization levels is L, number of bits is $n_b = \log_2 L$.

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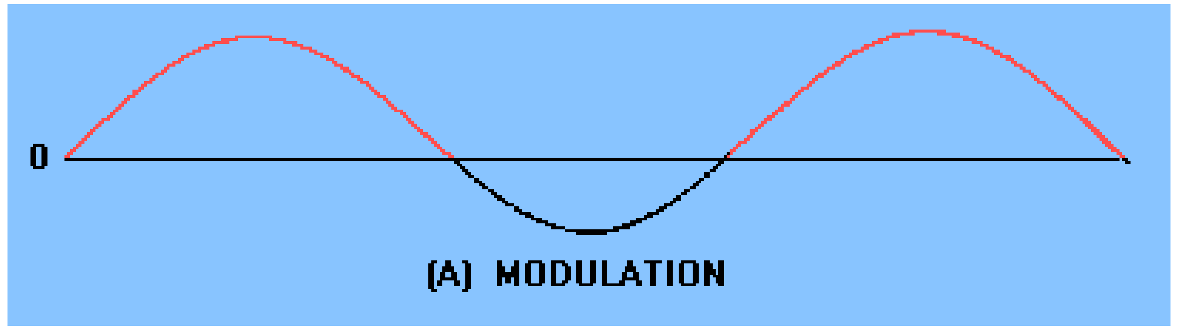
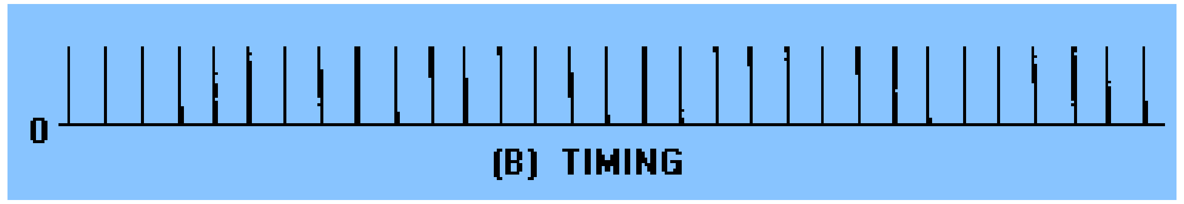
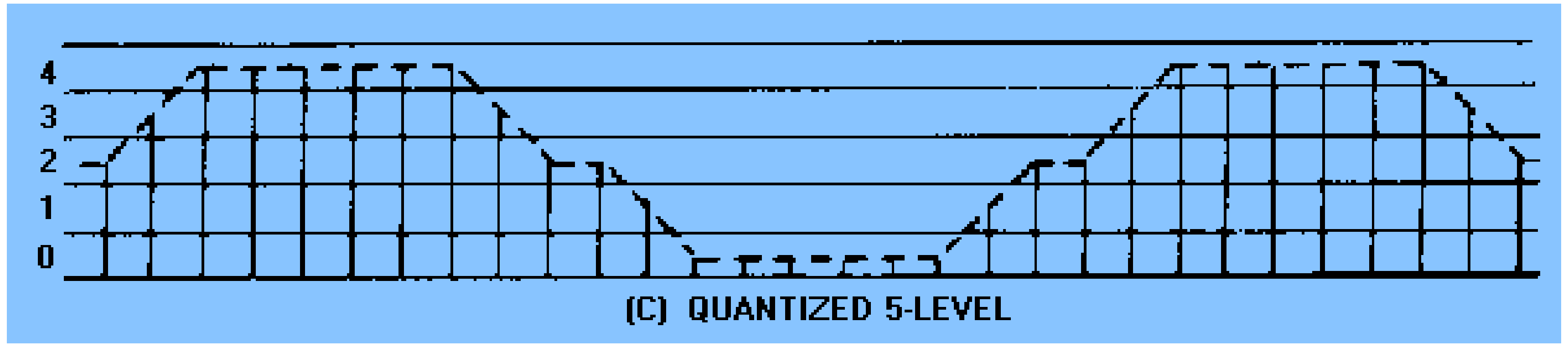
QUANTIZATION EXAMPLE



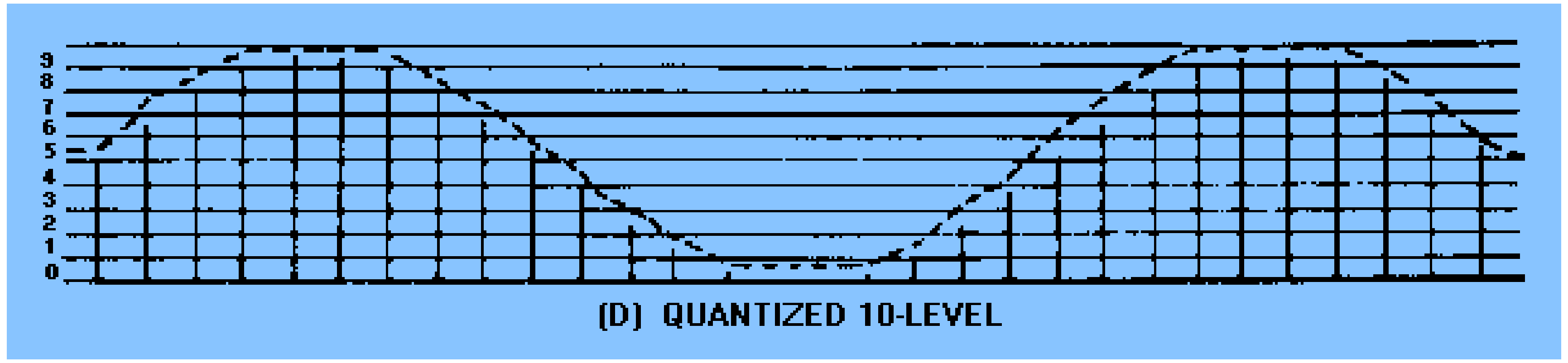
Analog signal



Sampling TIMING



Quantization levels.
Quantized to 5-levels



Quantization levels
Quantized 10-levels

QUANTIZATION ERROR:

- Quantization is an approximation process.
- The input values to the quantizer are the real values; the output values are the approximated values. The output values are chosen to be the middle value in the zone.
- If the input value is also at the middle of the zone, there is no quantization error; otherwise, there is an error.
- In the previous example, the normalized amplitude of the third sample is 3.24, but the normalized quantized value is 3.50. This means that there is an error of +0.26. The value of the error for any sample is less than $\Delta/2$.
- In other words, we have $-\Delta/2 \le \text{error} \le \Delta/2$.

Quantization error to the SNR_{dB}:

$$SNR_{dB} = 6.02n_b + 1.76 \text{ dB.}$$

where n_b = number of bits per sample

Example: What is the SNR_{dB} in the example of slide 22?

Solution: We have eight levels and 3 bits per sample, so $SNR_{dB} = 6.02(3) + 1.76 = 19.82$ dB.

Example: A telephone subscriber line must have an SNR_{dB} above 40. What is the minimum number of bits per sample?

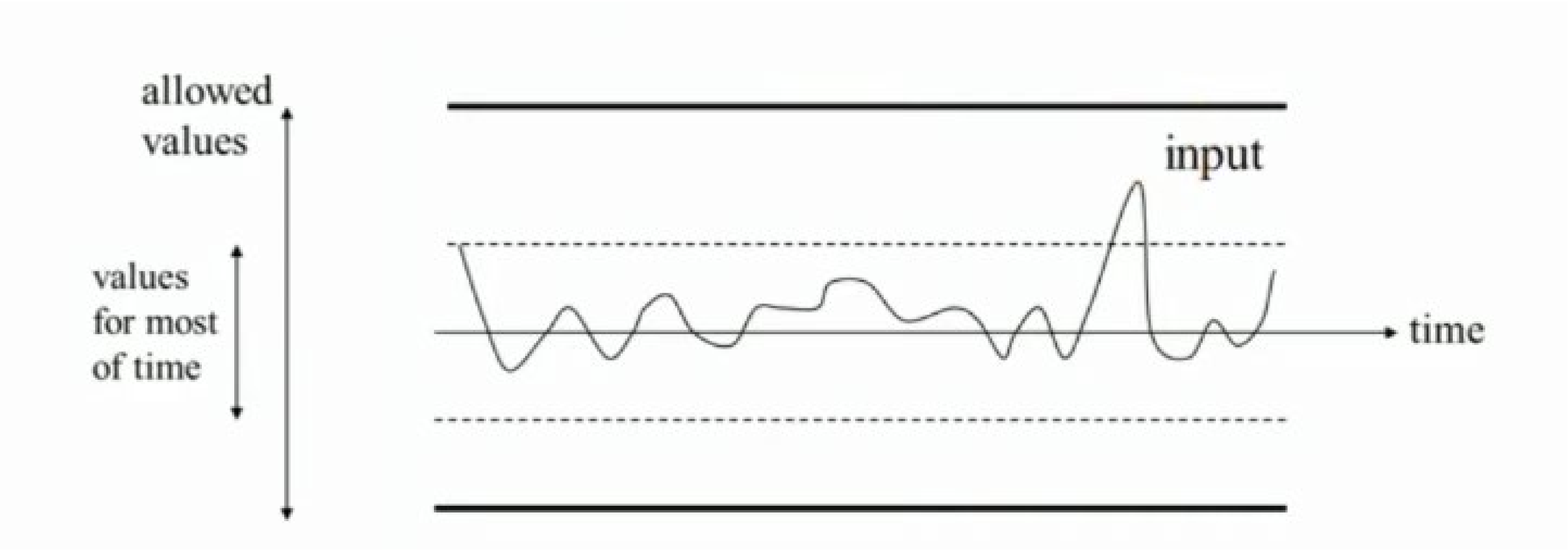
Solution: We can calculate the number of bits as

$$SNR_{dB} = 6.02(nb) + 1.76 = 40, \qquad nb = 6.35$$

Telephone companies usually assign 7 or 8 bits per sample.

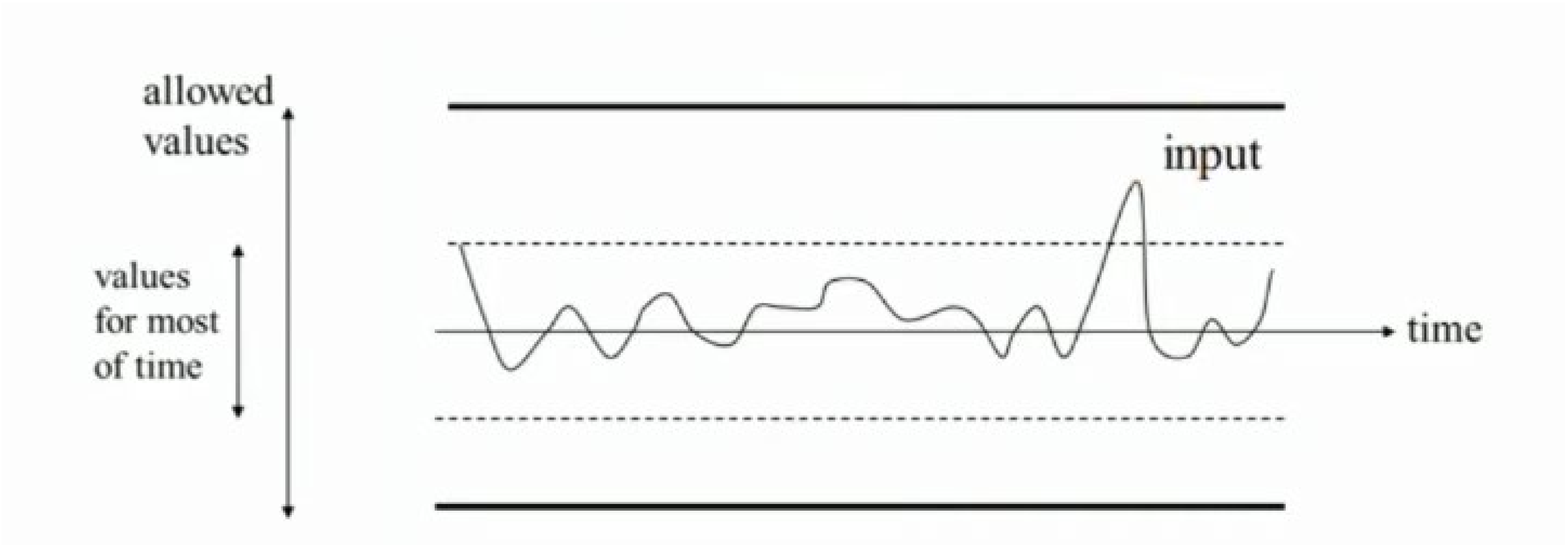
NON-UNIFORM QUANTIZATION

- The SNR, is an indication of the quality of the received signal. Ideally, we would like to have a constant SNR (the same quality) for all values of the message signal power $m(t)^2$.
- The SNR is directly proportional to the signal power, which varies from talker to talker.
- The signal power can also vary because of the connecting circuits.



NON-UNIFORM QUANTIZATION

- Statistically, it is found that smaller amplitudes predominate in speech and larger amplitudes are much less frequent. This means the SNR will be low most of the time.
- The quantization noise is proportional to the square of step size.



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NON-UNIFORM QUANTIZATION

- The problem can be solved by using smaller steps for smaller amplitudes (nonuniform quantizing), as shown in figure a.
- The horizontal axis is normalized input signal, and the vertical axis is the output signal y.
- The interval Δm contains large number of steps when m is small.
- The quantization noise is small for smaller input signal.

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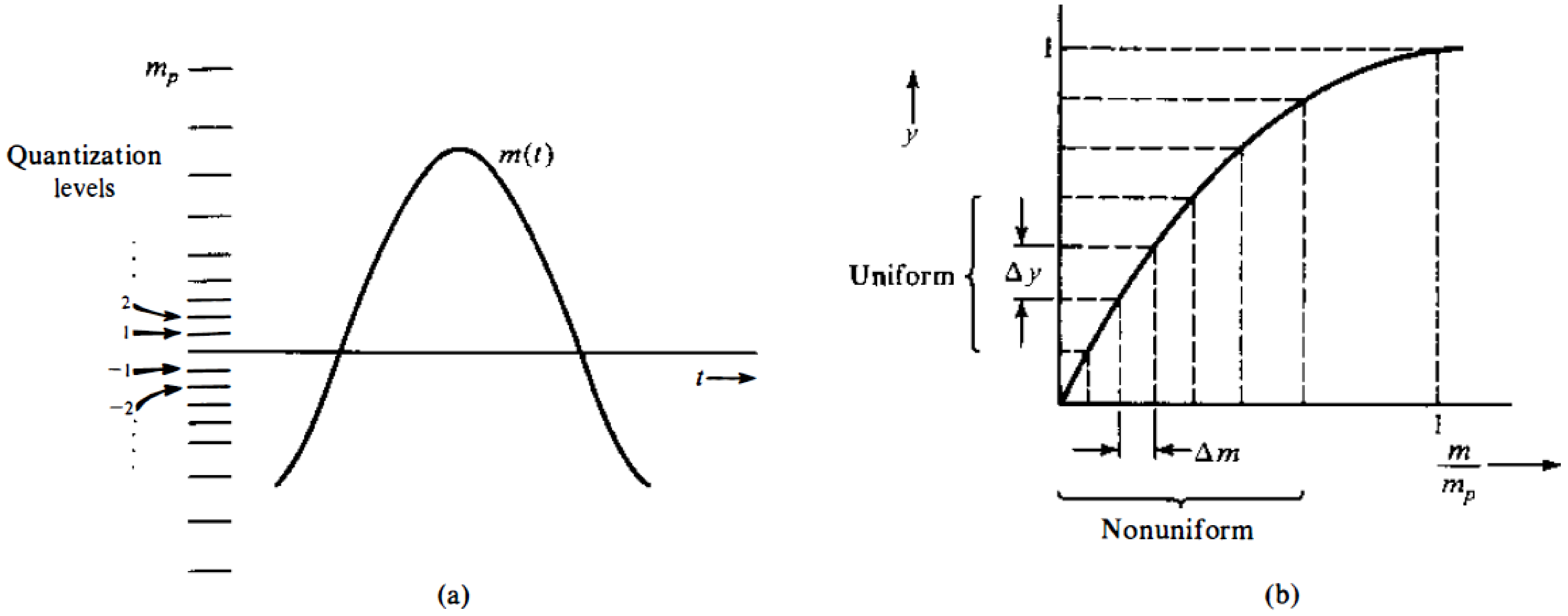
NON-UNIFORM QUANTIZATION

- The loud talkers and stronger signals are penalized with higher noise steps to compensate the soft talkers and weaker signals.



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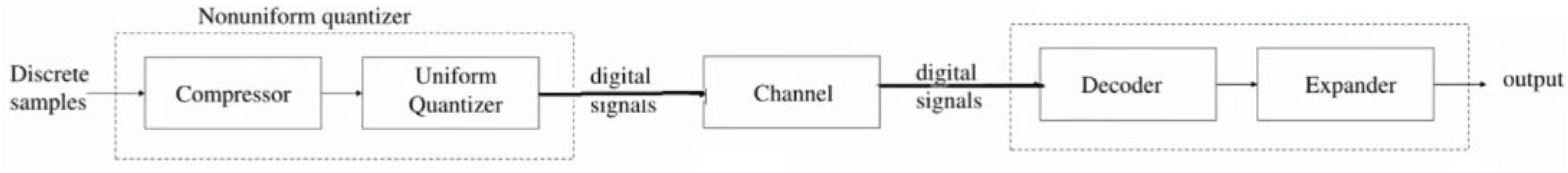
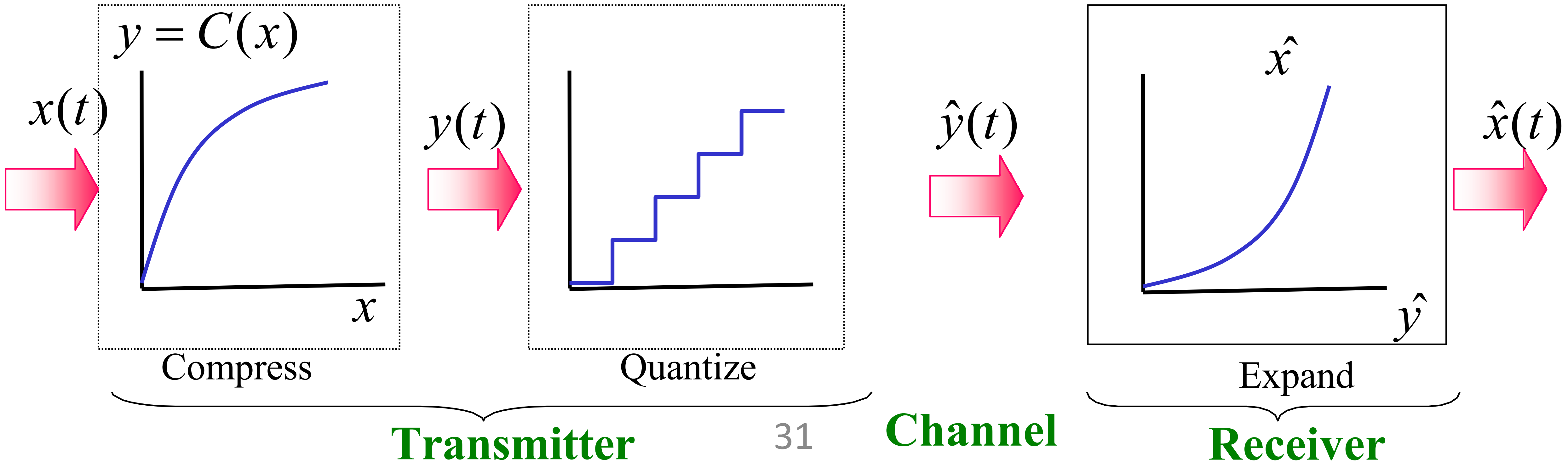
NON-UNIFORM QUANTIZATION

- It is achieved by uniformly quantizing the "compressed" signal.
- At the receiver, an inverse compression characteristic, called "expansion" is employed to avoid signal distortion.

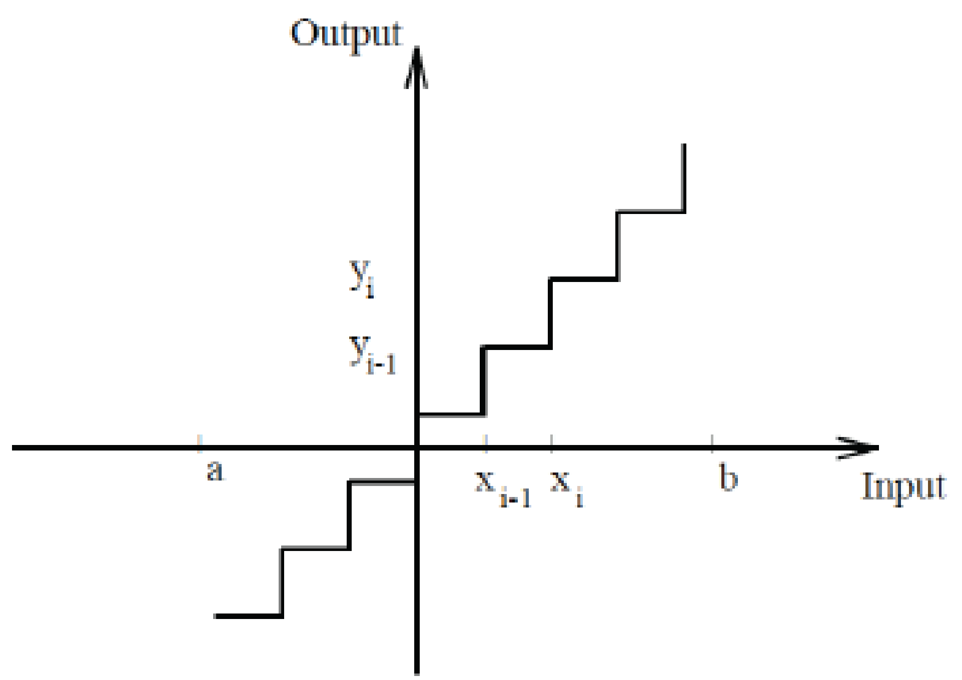


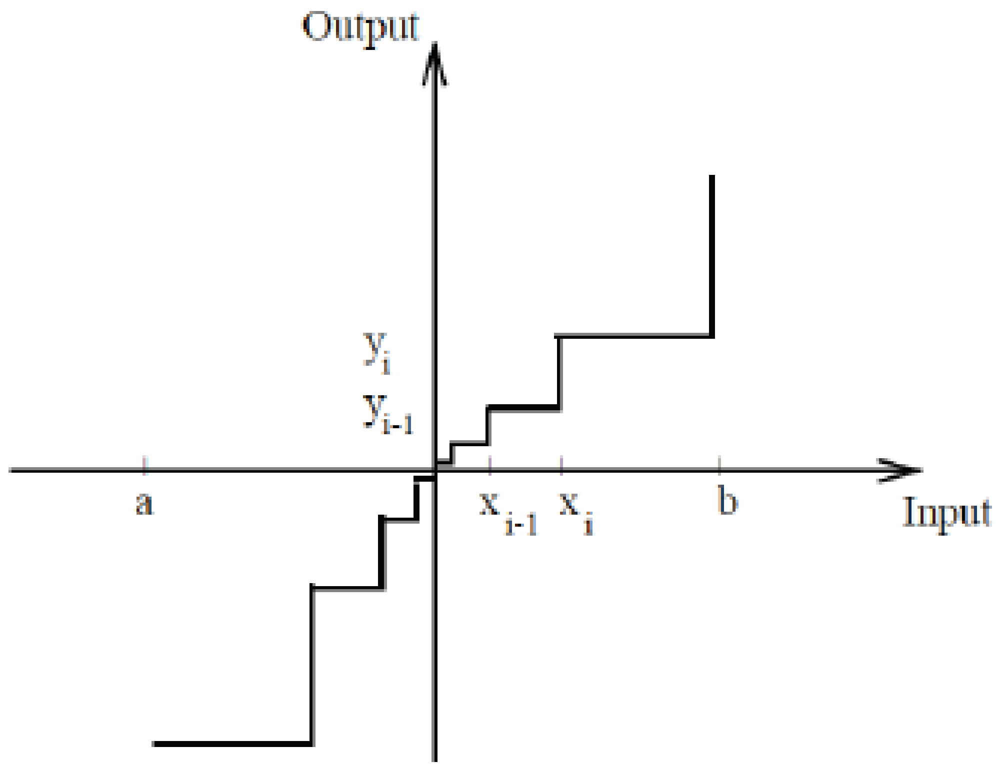
compression+expansion ⇨ companding



UNIFORM VS NON-UNIFORM QUANT.

- Uniform (linear) quantizing:
  - No assumption about amplitude statistics of the input.
  - Robust to small changes in input statistic by not finely tuned to a specific set of input parameters
  - Simple implementation
  - Application of linear quantizer:
    Signal processing, graphic and display applications, process control applications
- Non-uniform quantizing:
  - Using the input statistics to tune quantizer parameters
  - Larger SNR than uniform quantizing with same number of levels (HOW??)
  - Application of non-uniform quantizer:
    Commonly used for speech





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NON-UNIFORM QUANTIZATION

Among several choices, two compression laws have been accepted as desirable standards by the ITU-T:[6] the μ-law used in North America and Japan, and the A-law used in Europe and the rest of the world and on international routes. Both the μ-law and the A-law curves have odd symmetry about the vertical axis. The μ-law (for positive amplitudes) is given by

$$y = \frac{1}{\ln(1+\mu)} \ln\left(1 + \frac{\mu m}{m_p}\right) \qquad 0 \le \frac{m}{m_p} \le 1$$

The A-law (for positive amplitudes) is

$$y = \begin{cases} \dfrac{A}{1+\ln A}\left(\dfrac{m}{m_p}\right) & 0 \le \dfrac{m}{m_p} \le \dfrac{1}{A} \\[2ex] \dfrac{1}{1+\ln A}\left(1 + \ln \dfrac{A m}{m_p}\right) & \dfrac{1}{A} \le \dfrac{m}{m_p} \le 1 \end{cases}$$

EXPONENTIAL INCREASE OF THE OUTPUT SNR

$$\frac{S_o}{N_o} = c(2)^{2n}$$

where

$$c = \begin{cases} \dfrac{3\ \widetilde{m^2(t)}}{m_p^2} & \text{[uncompressed case]} \\[2ex] \dfrac{3}{[\ln(1+\mu)]^2} & \text{[compressed case]} \end{cases}$$

$$\frac{S_o}{N_o} = c(2)^{2B_T/B}$$

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EXPONENTIAL INCREASE OF THE OUTPUT SNR

Generally speaking, time compression of a signal increases its bandwidth. But in PCM, we are compressing not the signal $m(t)$ in time but its sample values. Because neither the time scale not the number of samples changes, the problem of bandwidth increase does not arise here. It happens that when a μ-law compandor is used, the output SNR is

$$\frac{S_o}{N_o} \simeq \frac{3L^2}{[\ln(1+\mu)]^2} \qquad \mu^2 \gg \frac{m_p^2}{\widetilde{m^2(t)}}$$

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TRANSMISSION BANDWIDTH AND THE OUTPUT SNR

For a binary PCM, we assign a distinct group of n binary digits (bits) to each of the L quantization levels. Because a sequence of n binary digits can be arranged in 2^n distinct patterns,

$$L = 2^n \qquad \text{or} \qquad n = \log_2 L$$

each quantized sample is, thus, encoded into n bits. Because a signal $m(t)$ band-limited to B Hz requires a minimum of $2B$ samples per second, we require a total of $2nB$ bit/s, that is, $2nB$ pieces of information per second. Because a unit bandwidth (1 Hz) can transmit a maximum of two pieces of information per second , we require a minimum channel of bandwidth B_T Hz, given by

$$B_T = nB \text{ Hz}$$

This is the theoretical minimum transmission bandwidth required to transmit the PCM signal.

TRANSMISSION BANDWIDTH AND THE OUTPUT SNR

Example A signal $m(t)$ band-limited to 3 kHz is sampled at a rate $33\frac{1}{3}\%$ higher than the Nyquist rate. The maximum acceptable error in the sample amplitude (the maximum quantization error) is 0.5% of the peak amplitude m_p. The quantized samples are binary coded. Find the minimum bandwidth of a channel required to transmit the encoded binary signal. If 24 such signals are time-division-multiplexed, determine the minimum transmission bandwidth required to transmit the multiplexed signal.

The Nyquist sampling rate is $R_N = 2 \times 3000 = 6000$ Hz (samples per second). The actual sampling rate is $R_A = 6000 \times (1\frac{1}{3}) = 8000$ Hz.

The quantization step is Δv, and the maximum quantization error is $\pm\Delta v/2$.

TRANSMISSION BANDWIDTH AND THE OUTPUT SNR

Example A signal $m(t)$ band-limited to 3 kHz is sampled at a rate $33\frac{1}{3}\%$ higher than the Nyquist rate. The maximum acceptable error in the sample amplitude (the maximum quantization error) is 0.5% of the peak amplitude m_p. The quantized samples are binary coded. Find the minimum bandwidth of a channel required to transmit the encoded binary signal. If 24 such signals are time-division-multiplexed, determine the minimum transmission bandwidth required to transmit the multiplexed signal.

$$\frac{\Delta v}{2} = \frac{m_p}{L} = \frac{0.5}{100} m_p \Longrightarrow L = 200$$

For binary coding, L must be a power of 2. Hence, the next higher value of L that is a power of 2 is $L = 256$.

we need $n = \log_2 256 = 8$ bits per sample. We require to transmit a total of $C = 8 \times 8000 = 64,000$ bit/s. Because we can transmit up to 2 bit/s per hertz of bandwidth, we require a minimum transmission bandwidth $B_T = C/2 = 32$ kHz.

The multiplexed signal has a total of $C_M = 24 \times 64,000 = 1.536$ Mbit/s, which requires a minimum of $1.536/2 = 0.768$ MHz of transmission bandwidth.

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NON-UNIFORM QUANTIZATION

A small increase in bandwidth yields a large benefit in terms of SNR. This relationship is clearly seen by using the decibel scale to rewrite Eq. (6.39) as

$$\begin{aligned}\left(\frac{S_o}{N_o}\right)_{\text{dB}} &= 10\log_{10}\left(\frac{S_o}{N_o}\right) \\ &= 10\log_{10}[c(2)^{2n}] \\ &= 10\log_{10} c + 2n\log_{10} 2 \\ &= (\alpha + 6n) \quad \text{dB}\end{aligned}$$

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NON-UNIFORM QUANTIZATION

Example A signal $m(t)$ of bandwidth $B = 4$ kHz is transmitted using a binary companded PCM with $\mu = 100$. Compare the case of $L = 64$ with the case of $L = 256$ from the point of view of transmission bandwidth and the output SNR.

NON-UNIFORM QUANTIZATION

For $L = 64$, $n = 6$, and the transmission bandwidth is $nB = 24$ kHz,

$$\frac{S_o}{N_o} = (\alpha + 36) \text{ dB}$$

$$\alpha = 10 \log \frac{3}{[\ln(101)]^2} = -8.51$$

Hence,

$$\frac{S_o}{N_o} = 27.49 \text{ dB}$$

For $L = 256$, $n = 8$, and the transmission bandwidth is 32 kHz,

$$\frac{S_o}{N_o} = \alpha + 6n = 39.49 \text{ dB}$$

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ENCODING

- The last step in PCM is encoding.
- After each sample is quantized and the number of bits per sample is decided, each sample can be changed to an nb-bit code word.
- In the previous figure in slide 22, the encoded words are shown in the last row.
- A quantization code of 2 is encoded as 010; 5 is encoded as 101; and so on.
- Note that the number of bits for each sample is determined from the number of quantization levels.
- The bit rate can be found from the formula:

$$\text{Bit rate} = \text{sampling rate x number of bits per sample} = fs \times nb$$

For no aliasing case (fs≥ 2B), the MINIMUM Bandwidth of PCM B_{pcm} is: BW_{pcm} = Bit rate/2 = fs x nb/2

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

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

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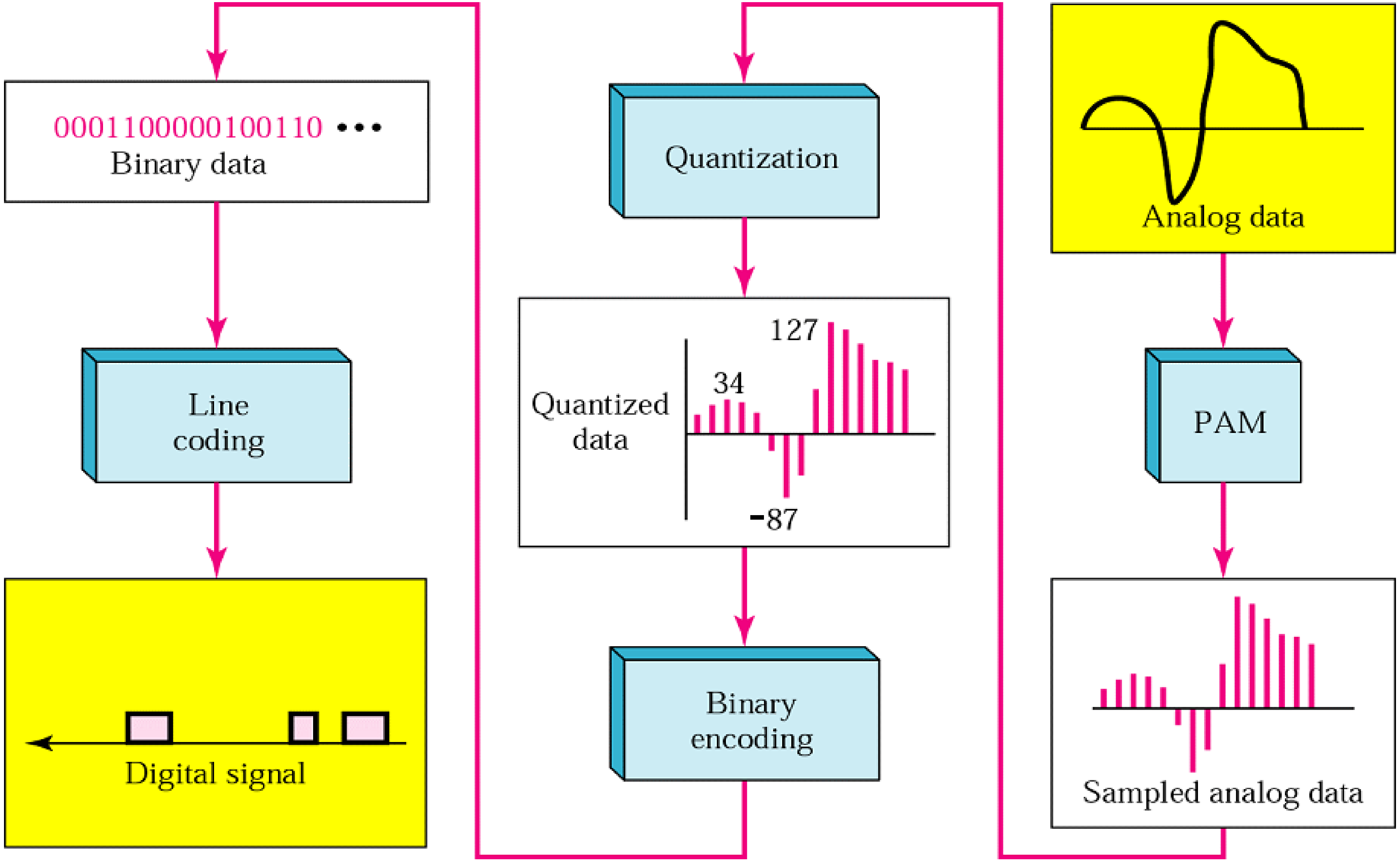
COMPLETE ANALOG TO DIGITAL CONVERSION



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DELTA AND ADAPTIVE DELTA MODULATION

Key Attributes About Delta Modulation

- Delta modulation (DM) is the simplest method for analog-to-digital conversion (ADC).
- DM uses 1-bit per sampling period (Ts) – it is a 1-bit ADC.
- DM requires a sampling rate much greater than the Nyquist rate (commonly four or five times the Nyquist rate).
- DM is closely related to DPCM.
- In DM we use a first-order predictor (one time delay TS is the predictor).
- DM uses very simple hardware and cost is low for that reason.

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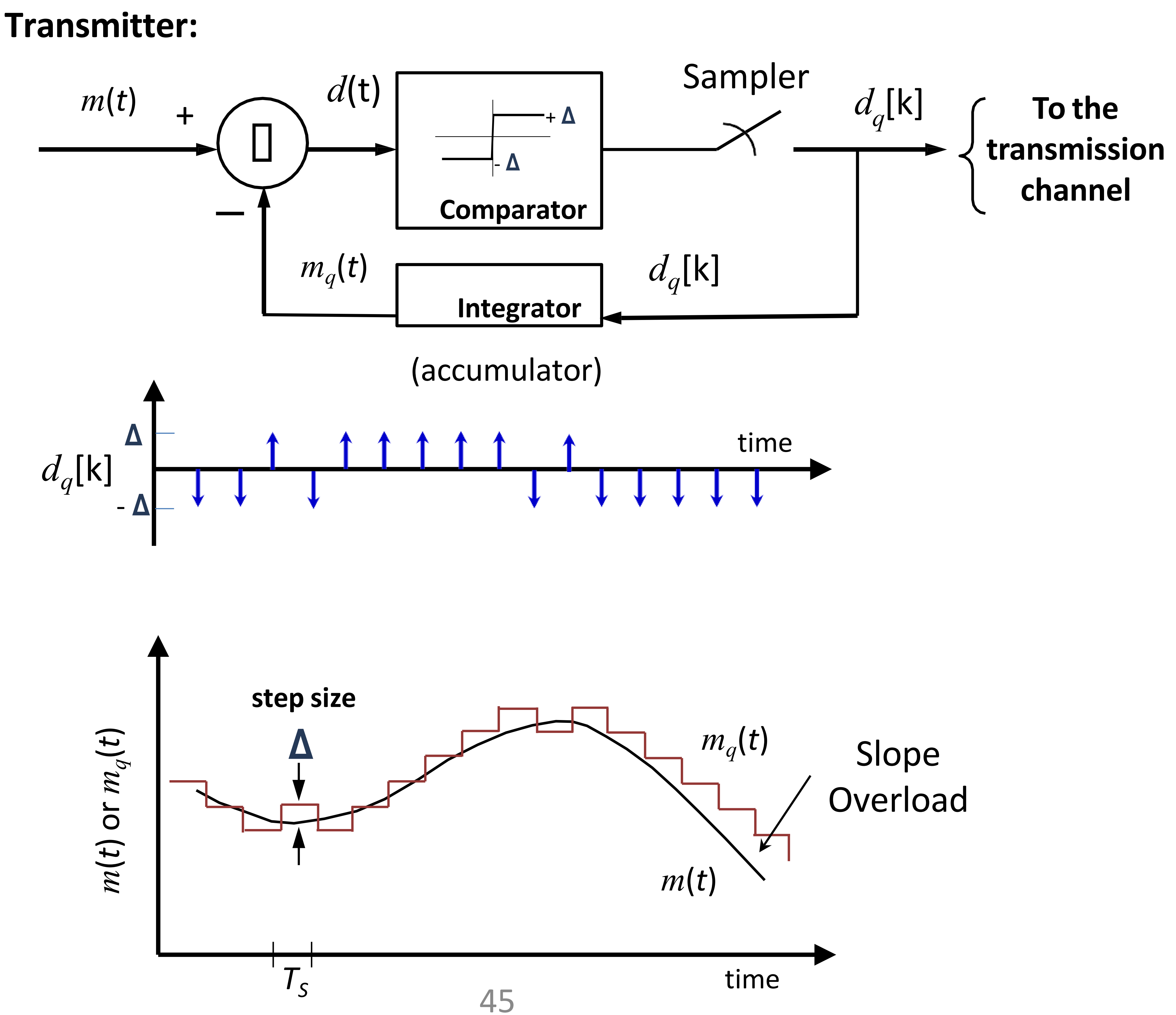
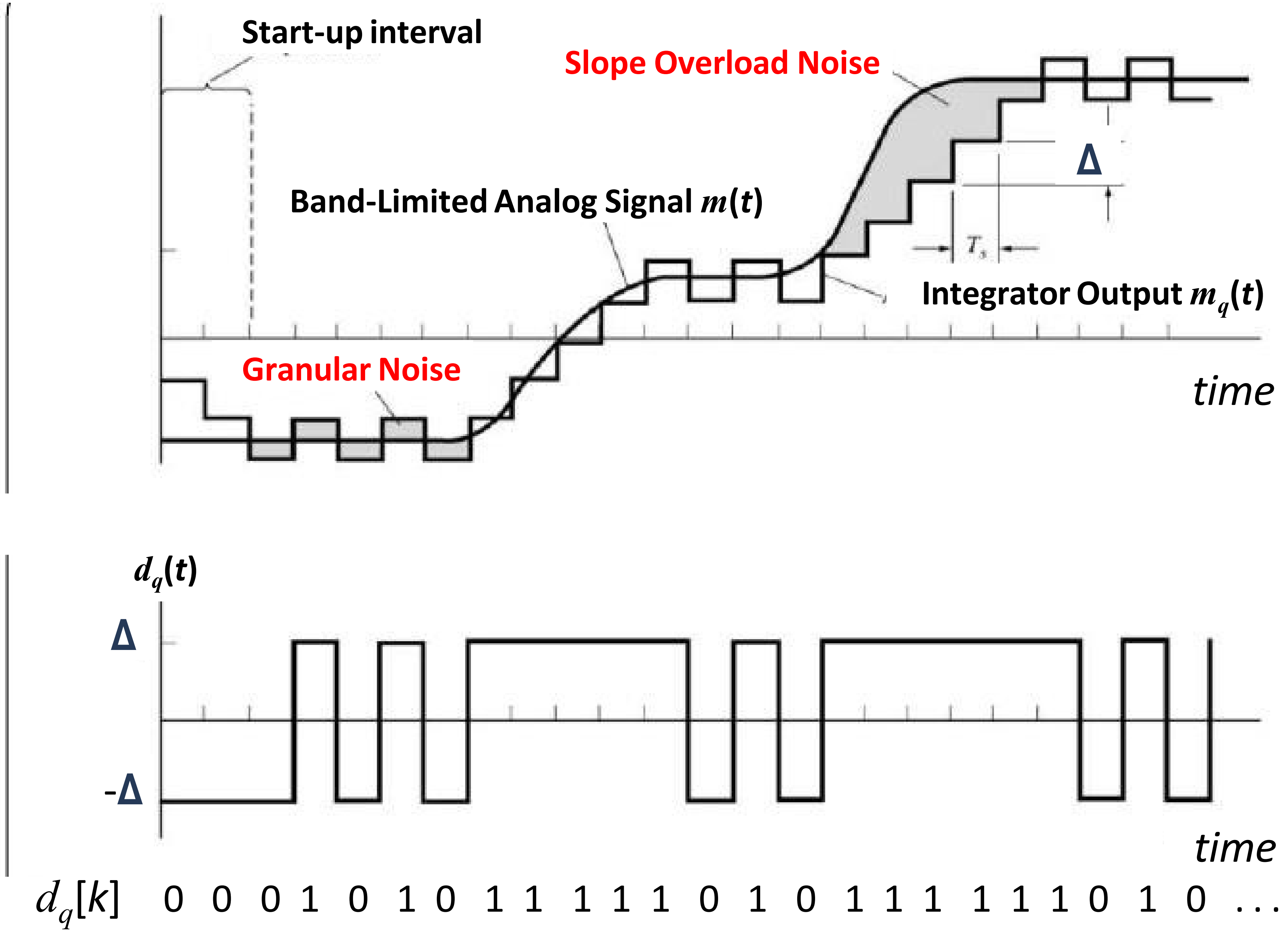
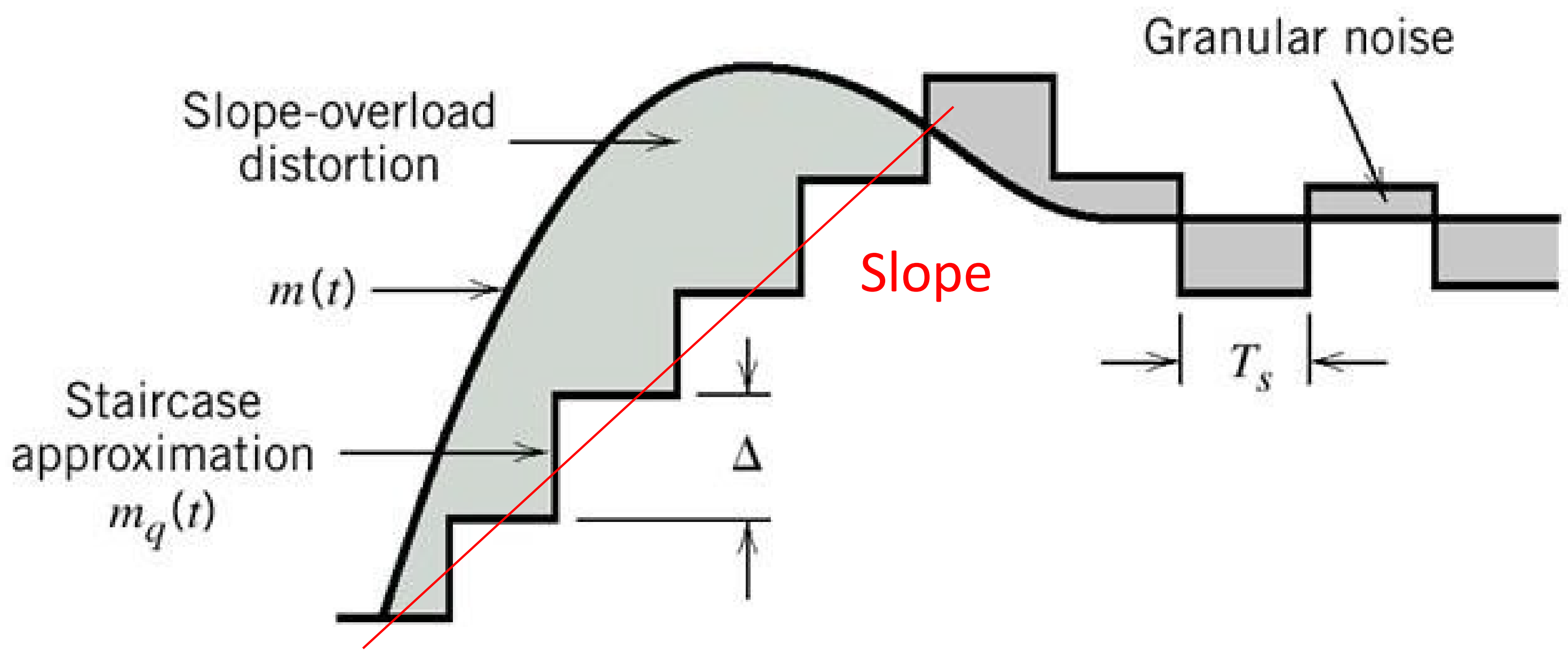
DELTA MODULATION



DELTA MODULATION



SLOPE-OVERLOAD DISTORTION AND GRANULAR NOISE



Suppose that the message signal is the sinusoidal signal $A_m \cos(2\pi f_m t)$, then the maximum amplitude Amax of this signal that can be tolerated without overload is given by:

$$Amax = \Delta fs/\omega$$

ADAPTIVE DELTA MODULATION THEORY

An Adaptive Delta Modulator (ADM) is basically used to quantize the difference between the current signal value and the predicted value of the following signal. It uses variable step height in order to predict the consequent values.

DIFFERENCES BETWEEN DELTA MODULATION & ADAPTIVE DELTA MODULATION

The differences between these two methods include the following.

•In delta modulation, the step size cannot be varied. It remains fixed for the entire signal. However, in the case of ADM, the step size can be varied according to the variation of the signal.

•In delta modulation, slope overload distortion and granular noise can be present but in ADM, quantization noise is present. The other types of errors are mainly avoided.

•This modulation utilizes the bandwidth a lot more efficiently as compared to that of delta modulation.

•It has a wider dynamic range than delta modulation.

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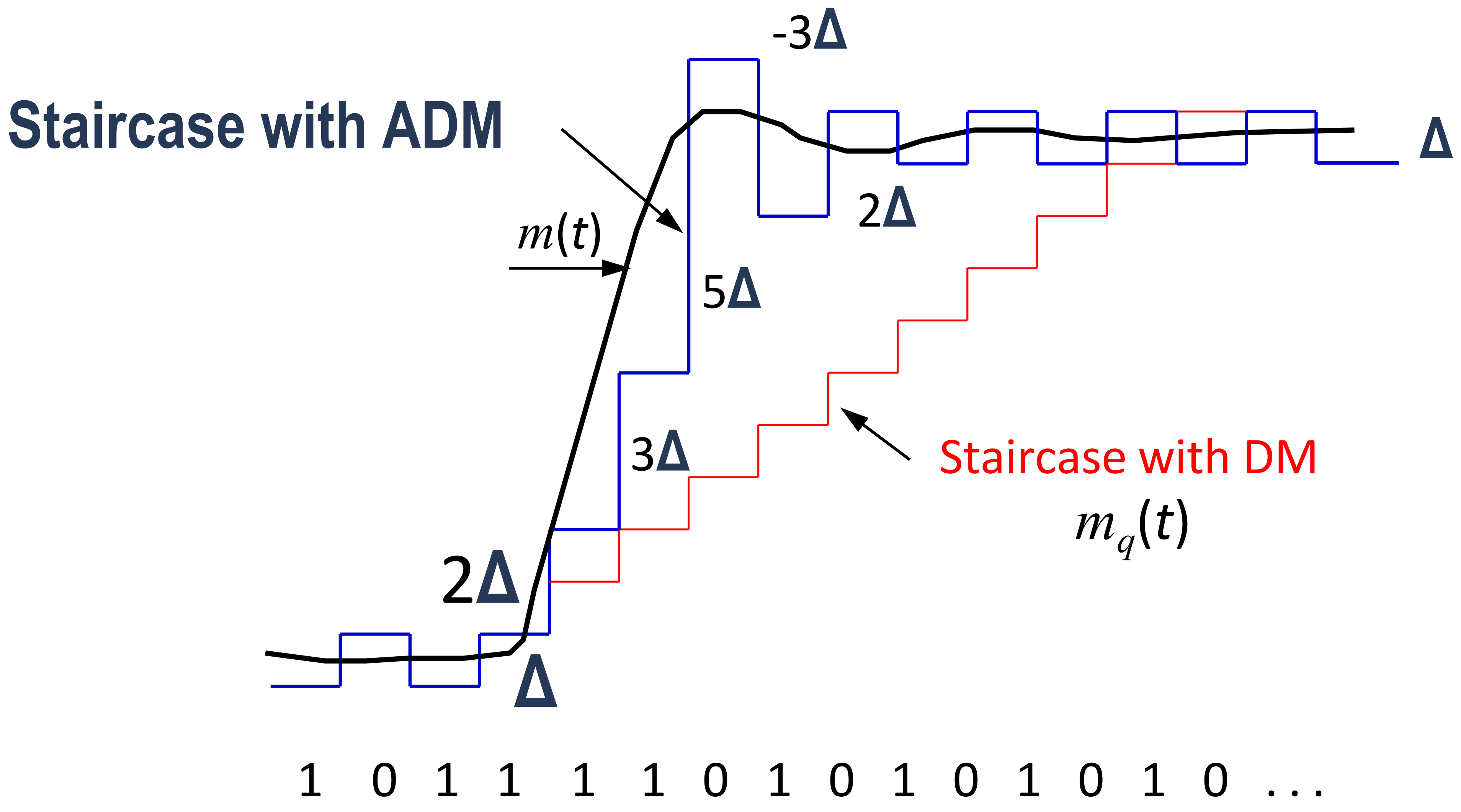
SPECIAL CASE: ADAPTIVE DELTA MODULATION

We can address the slope overload error problem with adaptive DM. Of course, it does add more complexity to its implementation.



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THE ADVANTAGES OF ADAPTIVE DELTA MODULATION INCLUDE THE FOLLOWING.

- This modulation can be used to reduce the slope error which is present in delta modulation.
- This can be used to remove the granular noises from the signal.
- It has an improved signal to noise ratio as compared to that of delta modulation.
- ADM has a low pass filter that can be used to remove the quantization noise.
- It has a higher dynamic range as compared to delta modulation.
- There is no need for error detection in the case of adaptive delta modulation.

THE APPLICATIONS OF ADAPTIVE DELTA MODULATION INCLUDE THE FOLLOWING.

- It is effectively used in audio communications.
- It can be used in systems that require improved voice quality.
- It is used in voice coding.
- It is used in television signal transmission.
- The SECURENET line of Motorola uses 12kbit/sec adaptive delta modulation.
- It is used by NASA for different communications between spacecraft and mission control.
- The US Army also uses adaptive delta modulation in order to conserve bandwidth over various tactical links.

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WORKED EXAMPLE FOR DM

Problem:

A Delta modulated system is designed to operate at five times the Nyquist Rate. The signal bandwidth B at its input port is 3 kHz and the quantized step Δ is 250 millivolts (0.25 volt). For this problem we assume a 2 kHz sinusoidal input – Find the maximum amplitude A_m of this 2 kHz tone that avoids slope overload.

Solution:

We know that B = 3 kHz, f_m = 2 kHz and Δ = 250 mV.

The Nyquist rate is 3,000 $\times$ 2 = 6,000 Hz. So five time the Nyquist rate is 30,000 Hz = f_S.

$$A_m \leq \frac{\Delta \cdot f_S}{2\pi f_m} = \frac{(0.25)30,000}{2\pi(2,000)} = 0.60 \text{ volt}$$

COMPARING PCM WITH DM

Problem:

One kilohertz (1 kHz) signal $m(t)$ is sampled at 8 kHz with 12-bit encoding for PCM transmission.

(a) How many bits are transmitted per second in in PCM? What is the bandwidth required in this case?

(b) Switch to using DM with 8 kHz sampling. How many bits are transmitted per second using DM? What is the bandwidth required in using DM?

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College of Electronics Engineering

Systems & Control Engineering Department

Digital Communications (SCE3316)

Lecture 3
(Analog Pulse Modulation)

Mr. Saadi Al-Saadoun
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INTRODUCTION

Continuous-Wave (CW) Modulation: (studied in the first semester)

- Some parameters of a sinusoidal carrier wave is varied continuously in accordance with the message signal
- Types: Amplitude Modulation (AM), Frequency Modulation (FM), and Phase Modulation (PM)

In Pulse Modulation: (to be studied in this semester)

- Some parameters of a pulse train is varied continuously in accordance with the message signal. We may distinguish two families of pulse modulation: **analog pulse modulation** and **digital pulse modulation.**

PULSE MODULATION TYPES

In analog pulse modulation, a periodic pulse train is used as the carrier wave, and some characteristic features of each pulse (e.g., amplitude, duration, or position) are varied continuously by the corresponding sample value of the message signal. Thus, in analog pulse modulation, information is transmitted in analog form, but the transmission occurs at discrete times.

In digital pulse modulation, on the other hand, the message signal is represented in a discrete form in both time and amplitude, thereby permitting its transmission in digital form as a sequence of coded pulses; this form of signal transmission has no CW counterpart.

Two potential advantages of pulse modulating over CW modulation:

Transmitted Power, can be concentrated into short bursts instead of being generated continuously.

Time Intervals, between pulses can be filled with sample values from other signals, [Time-division Multiplexing (TDM)]

The disadvantage of Pulse Modulation is that it requires a very large transmission bandwidth compared to the message bandwidth.

PULSE-MODULATION TYPES

- **The three main types of analog pulse modulation:**

    1. PAM: Pulse Amplitude Modulation.
    2. PWM: Pulse Width Modulation.
    3. PPM: Pulse Position Modulation.

- **The four main types of digital pulse modulation:**

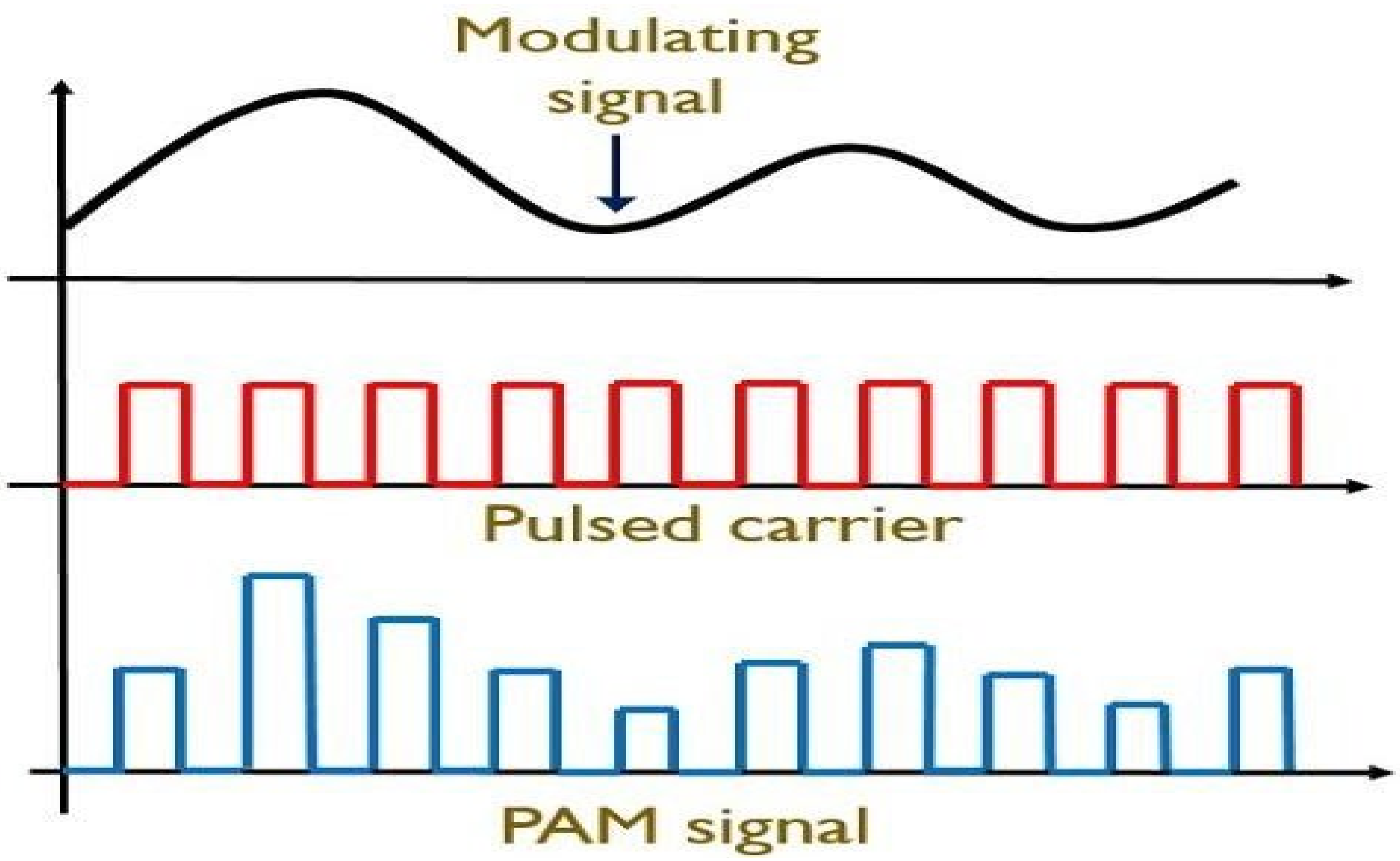
    1. PCM: Pulse Code Modulation.
    2. DPCM: Differential Pulse Code Modulation.
    3. DM: Delta Modulation.
    4. SDM: Sigma Delta Modulation.

(Pulse-Modulation Types)

Analog	Digital
(PAM, PWM, PPM)	(PCM, Diff. PCM, DM, ΣDM)

PULSE AMPLITUDE MODULATION

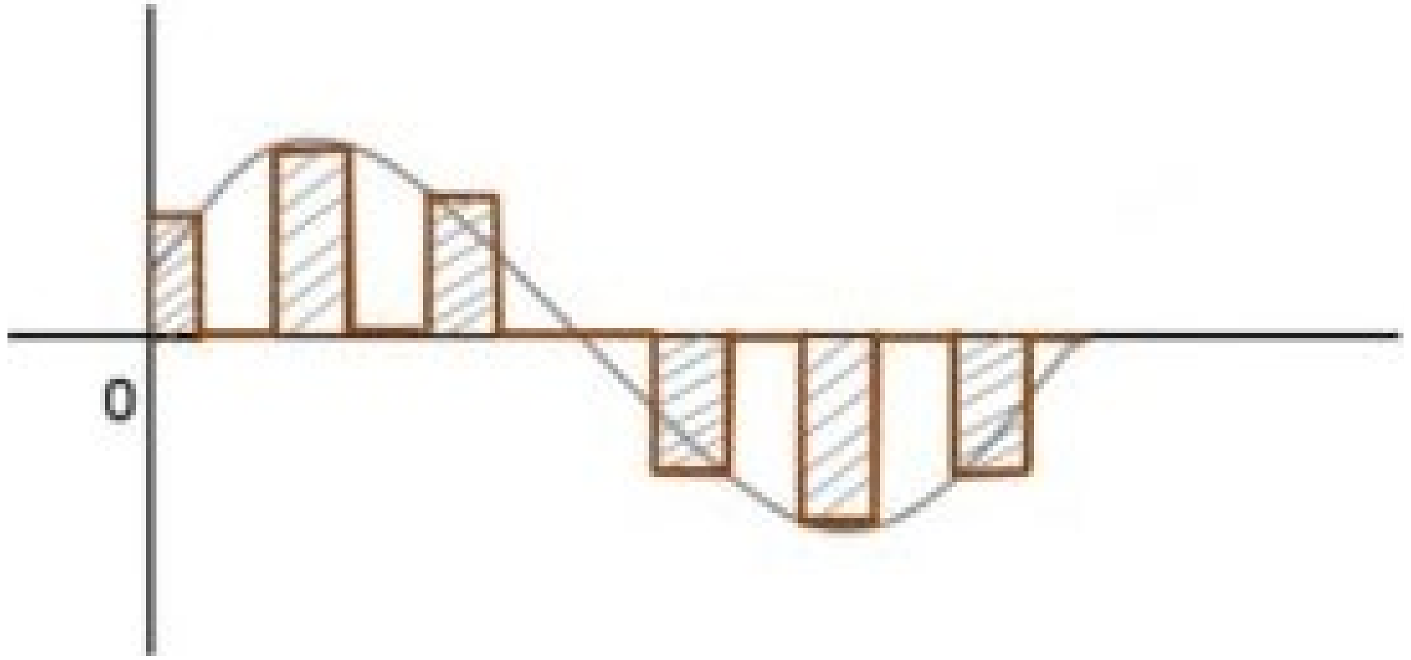
- The amplitude of the pulses of the carrier pulse train is varied in accordance with the modulating signal, that is amplitude of the pulses depends on the value of m(t) during the pulse time.



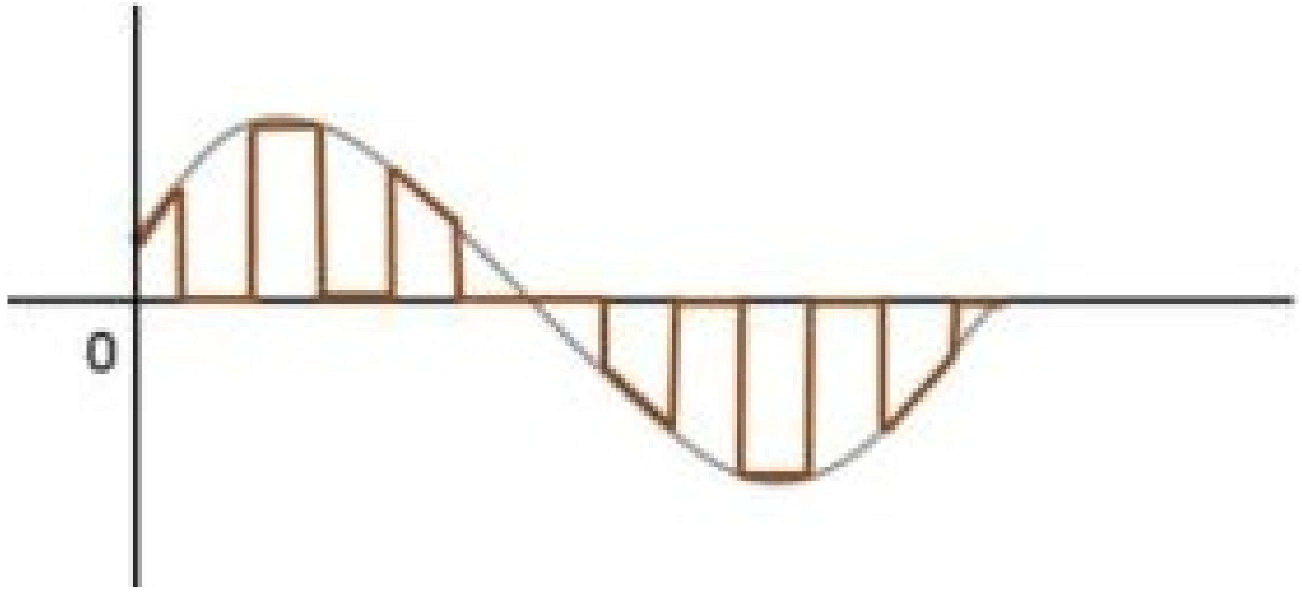
- There are two types of sampling techniques for transmitting a signal using PAM. They are:

(i) Flat Top PAM (ii) Natural PAM

Flat Top PAM: The amplitude of each pulse is directly proportional to the modulating signal amplitude at the time of pulse occurrence. The signal's amplitude cannot be changed with respect to the analog signal to be sampled. The tops of the amplitude remain flat.

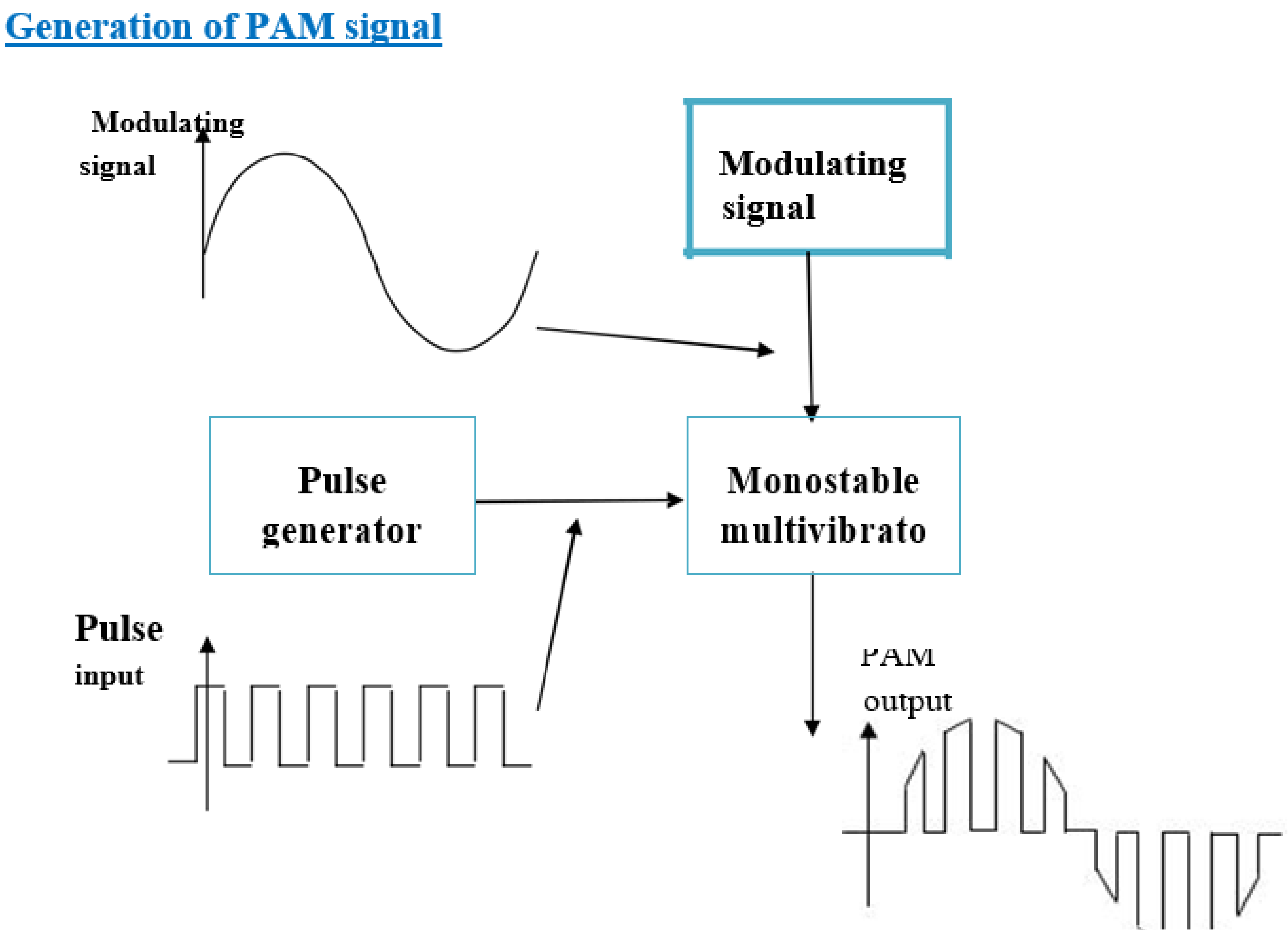


Natural PAM: The amplitude of each pulse is directly proportional to the modulating signal amplitude at the time of pulse occurrence and then the amplitude of the pulse for the rest of the half-cycle.



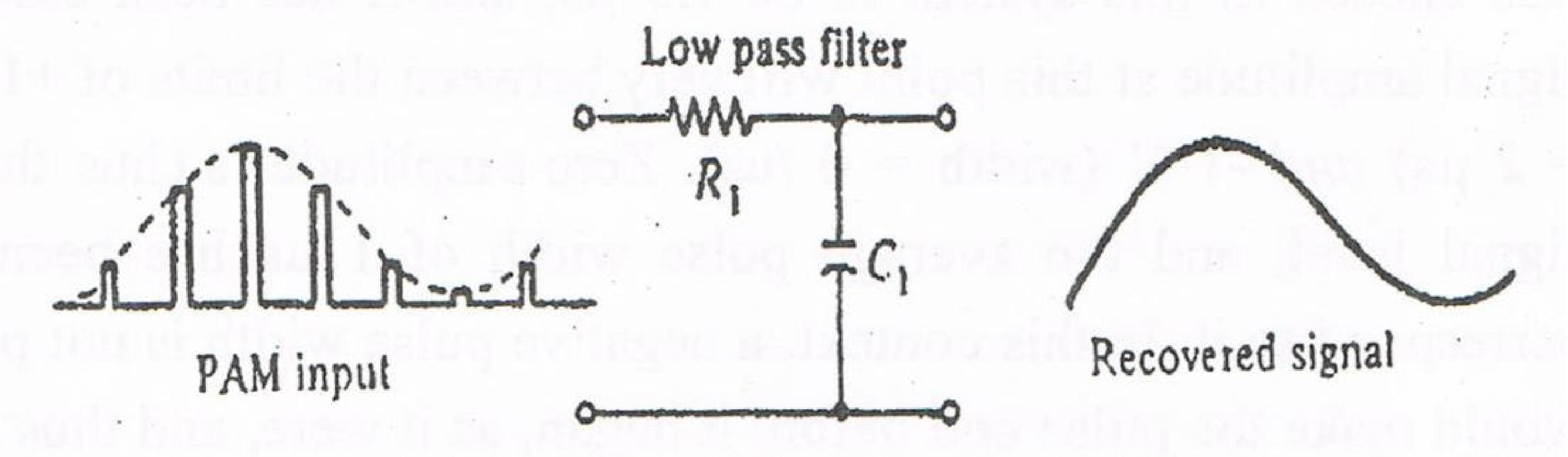
GENERATION OF PAM

- A PAM waveform production process is illustrated in the figure, in which a pulse generator triggers a monostable multivibrator at a sampling frequency.
- The output pulses from the multivibrator are made to increase and decrease in amplitude by the modulating signal.



DEMODULATION OF PAM SIGNAL

- This process is accomplished simply by passing the amplitude-modulated pulses through an LPF, where the PAM waveform consists of the fundamental modulating frequency and a number of high-frequency components which give the pulses their shape.
- The filter output is the low-frequency component corresponding to the original baseband signal.



Advantages of PAM

- The simplest pulse modulation technique
- The possibility of sending more than one signal on the same channel, each with a specific time slot (using TDM)

Disadvantages of PAM

- Less noise immunity than the other types of analog modulation
- Not power saving

PULSE TIME MODULATION (PTM)

- In pulse time modulation, pulse amplitude is held constant. In contrast, the position or width of the pulse is proportional to the amplitude of the signal at the sampling instant.
- There are two types of pulse time modulation.
  i. Pulse Width Modulation (PWM)
  ii. Pulse Position Modulation (PPM)

Pulse Width Modulation

- PWM is also called, Pulse Duration Modulation (PDM) or Pulse Length Modulation (PLM).
- In PWM, the width of pulses of the carrier pulse train varies according to the modulating signal.

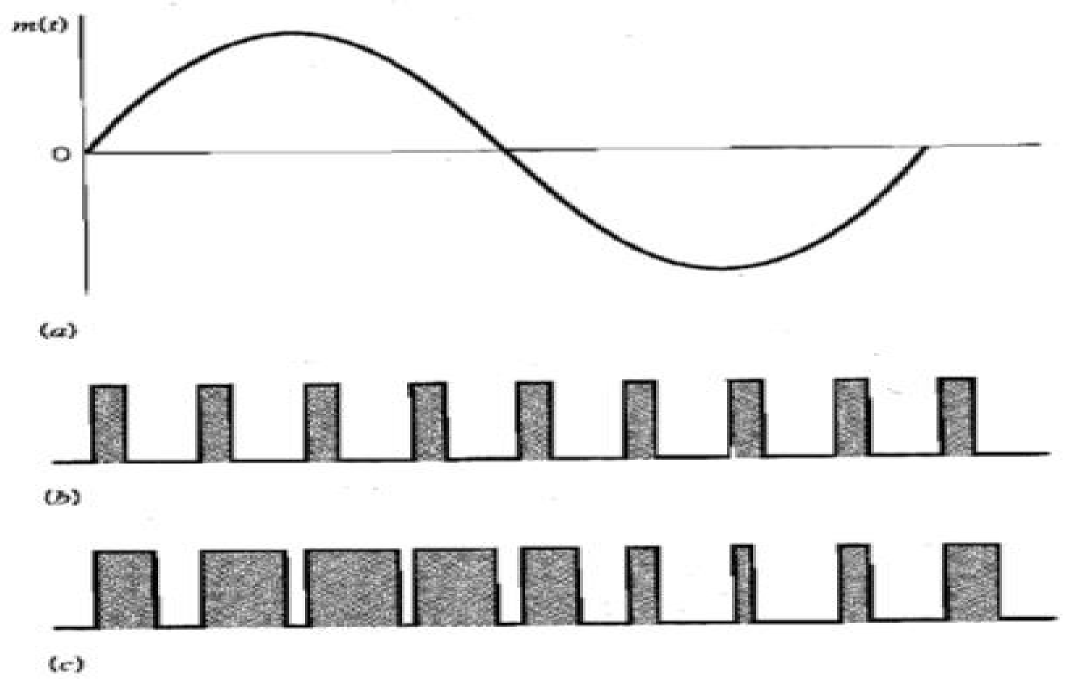


Figure: Illustration of PWM

(a)Modulating s signal

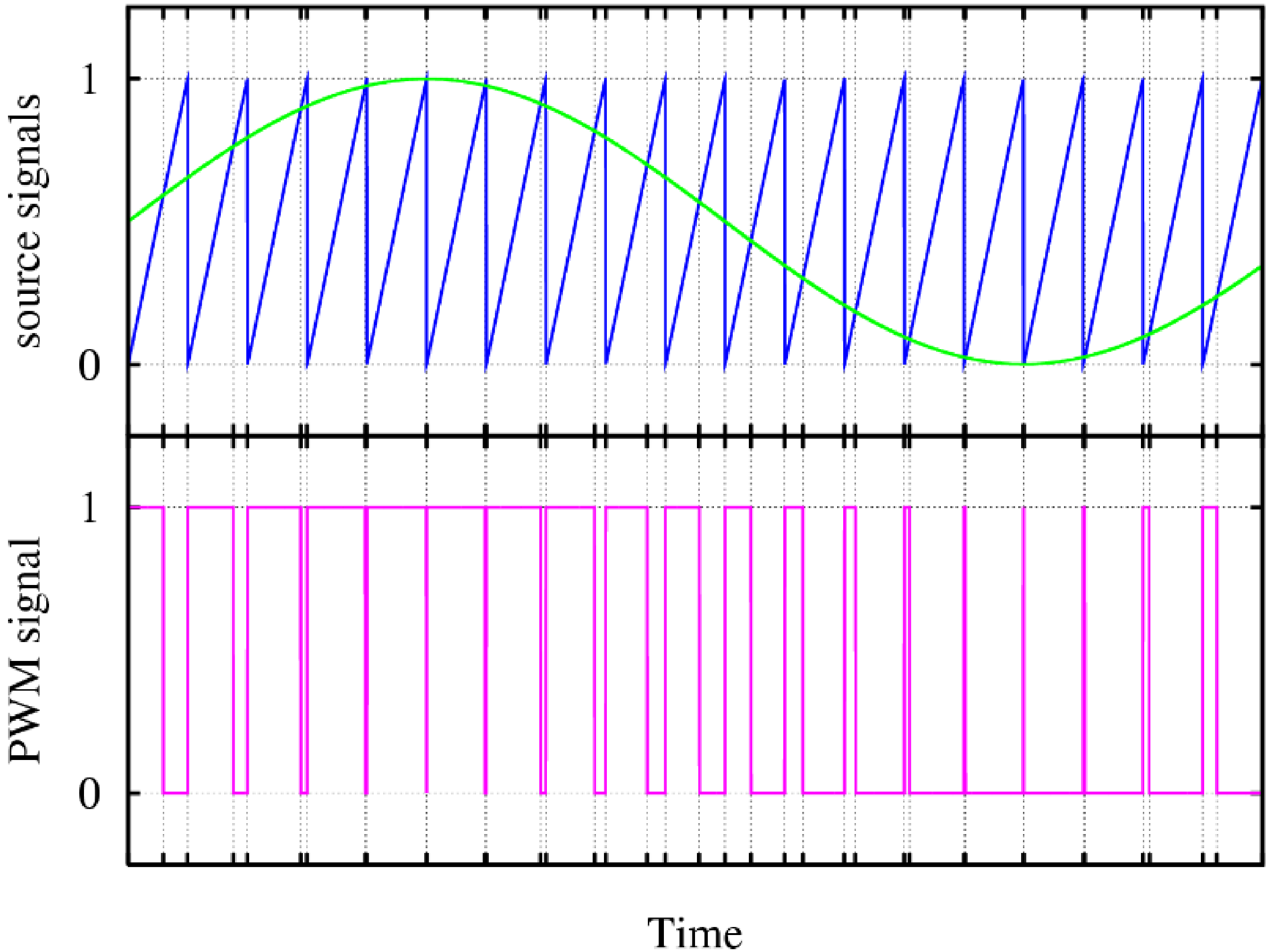
(b) Pulse Carrier

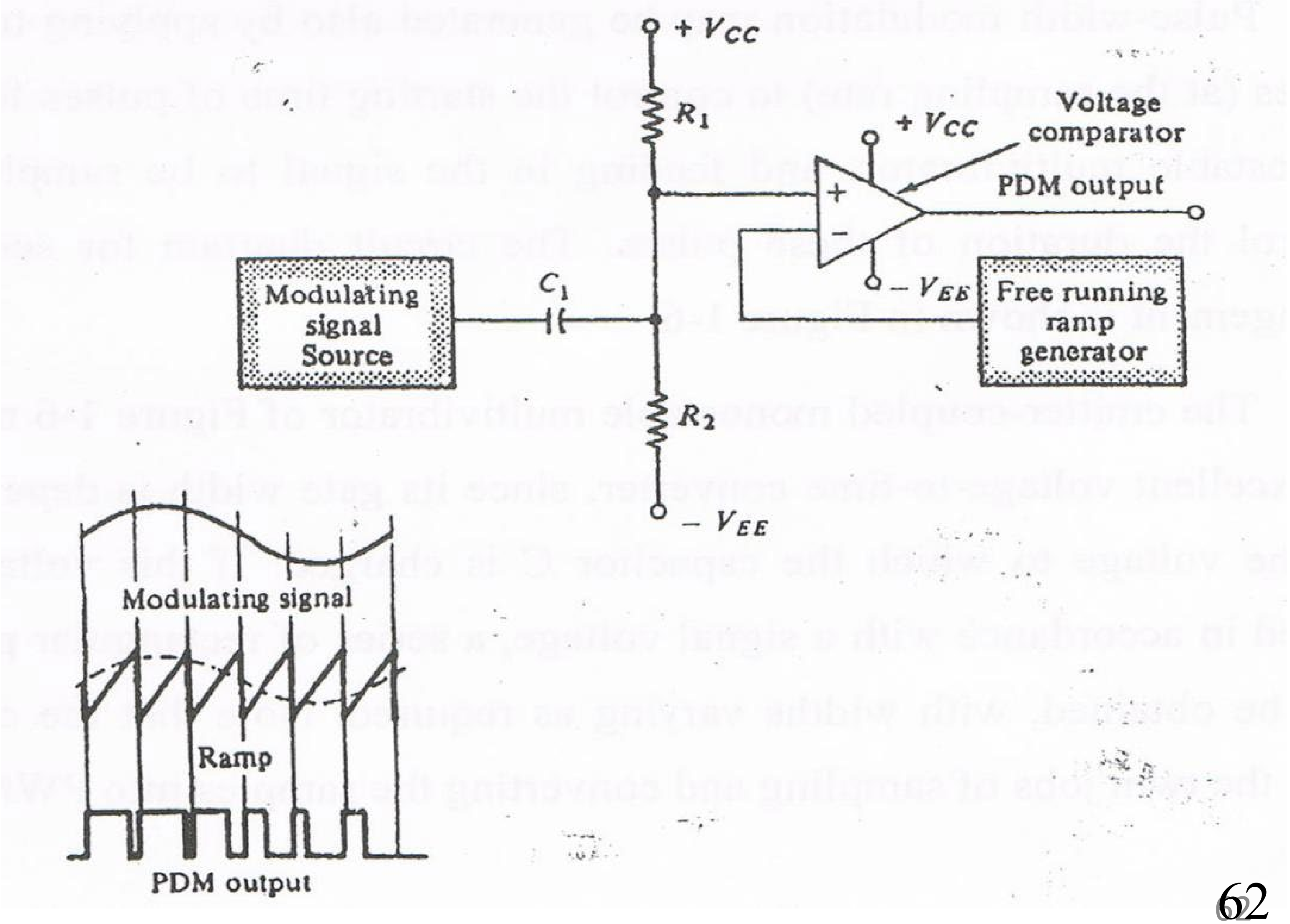
(c) PWM signal

PULSE WIDTH MODULATION (PWM)

- PWM is sometimes called pulse duration modulation (PDM) or pulse length modulation (PLM), as the width (an active portion of the duty cycle) of a constant amplitude pulse is varied proportional to the amplitude of the analog signal at the time the signal is sampled.
- The maximum analog signal amplitude produces the widest pulse, and the minimum analog signal amplitude produces the narrowest pulse. Note, however, that all pulses have the same amplitude.

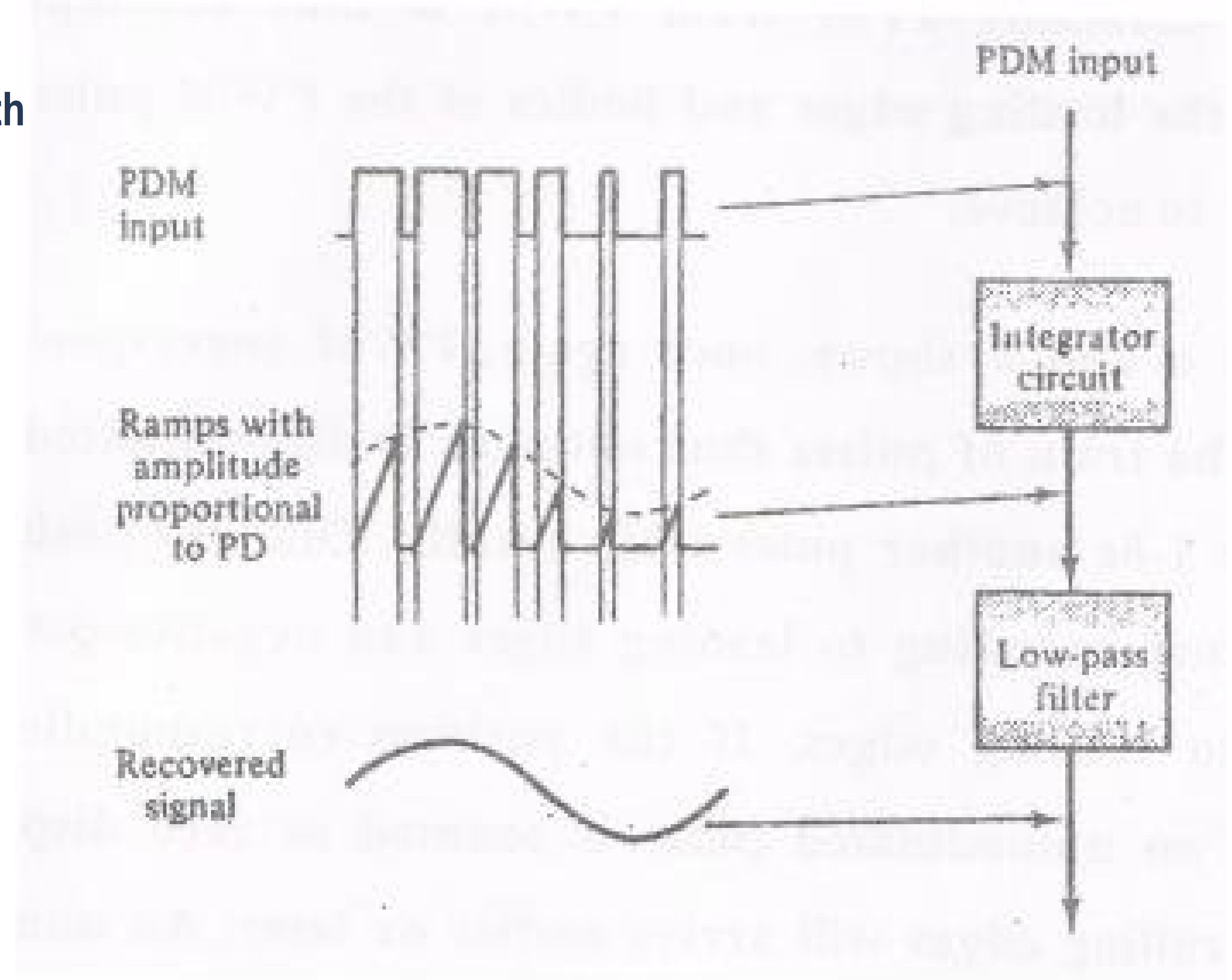
Generation of PWM signal





DEMODULATION OF PWM SIGNAL

- A miller integrator circuit is suitable for use with PWM demodulation
- The integrator circuit converts the PWM to PAM and using low-pass filter to recover the original signal

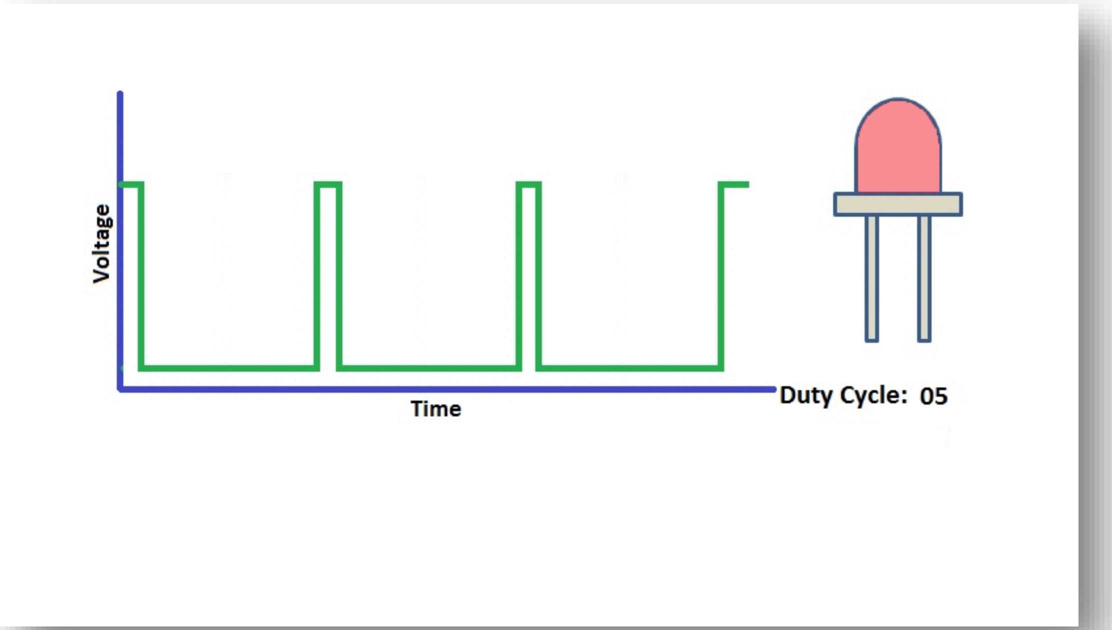


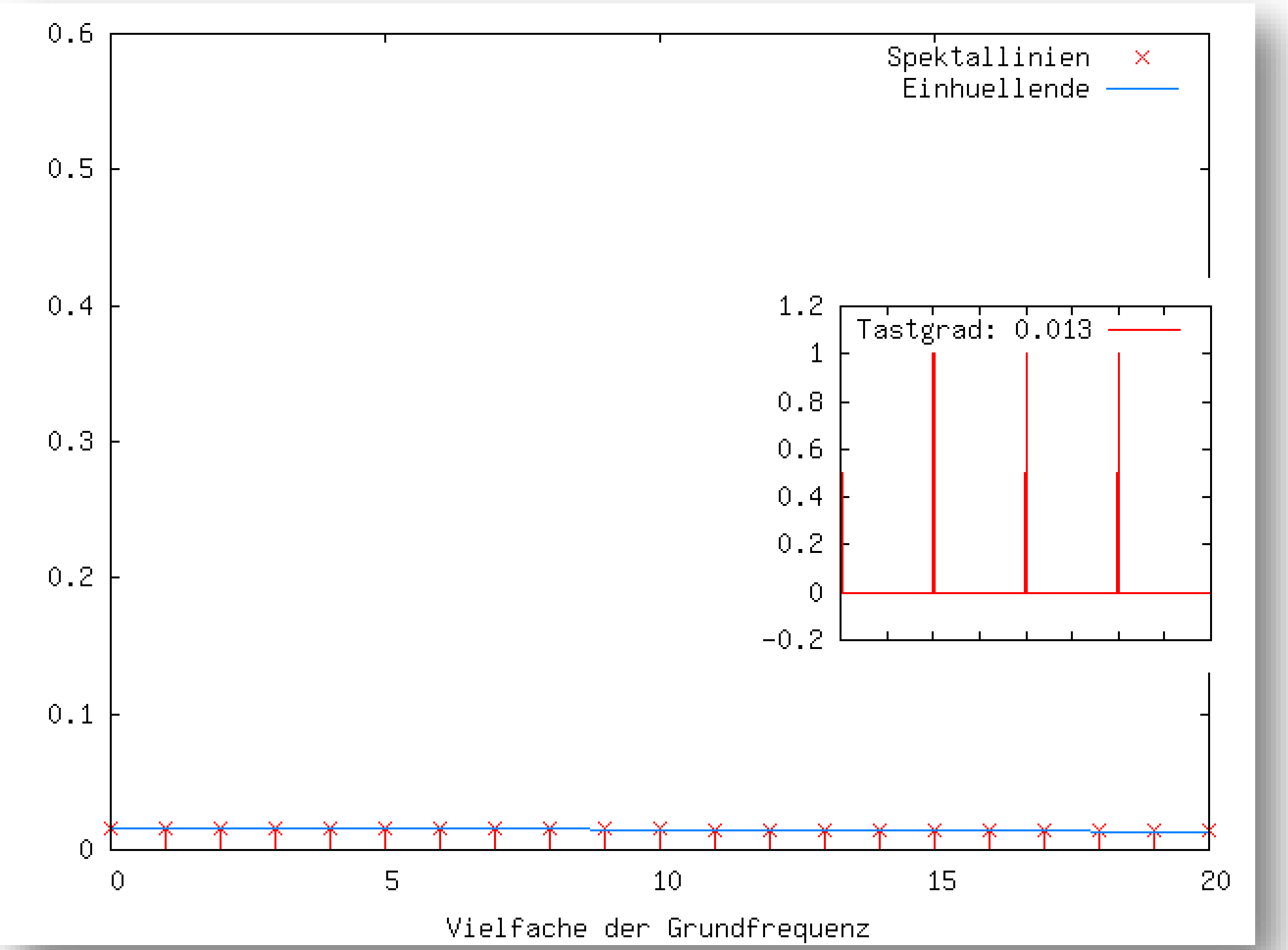
ADVANTAGES OF PWM

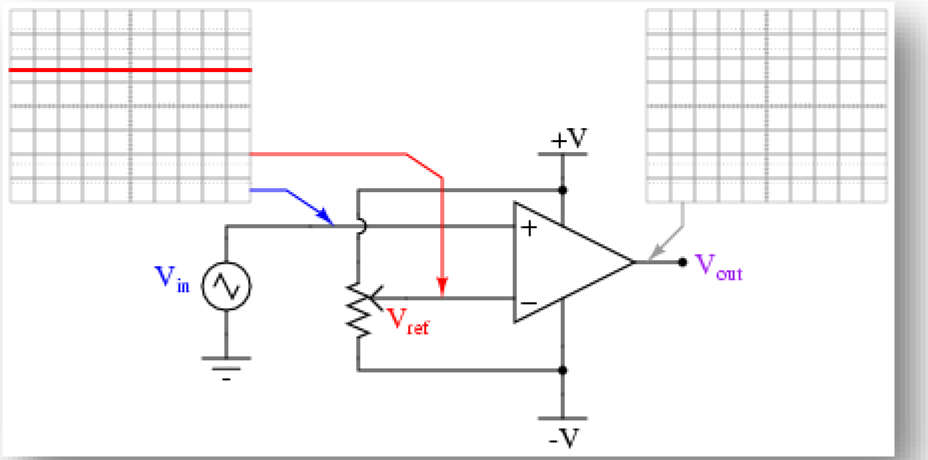
- Has high noise immunity where modulated pulses are constant in amplitude.

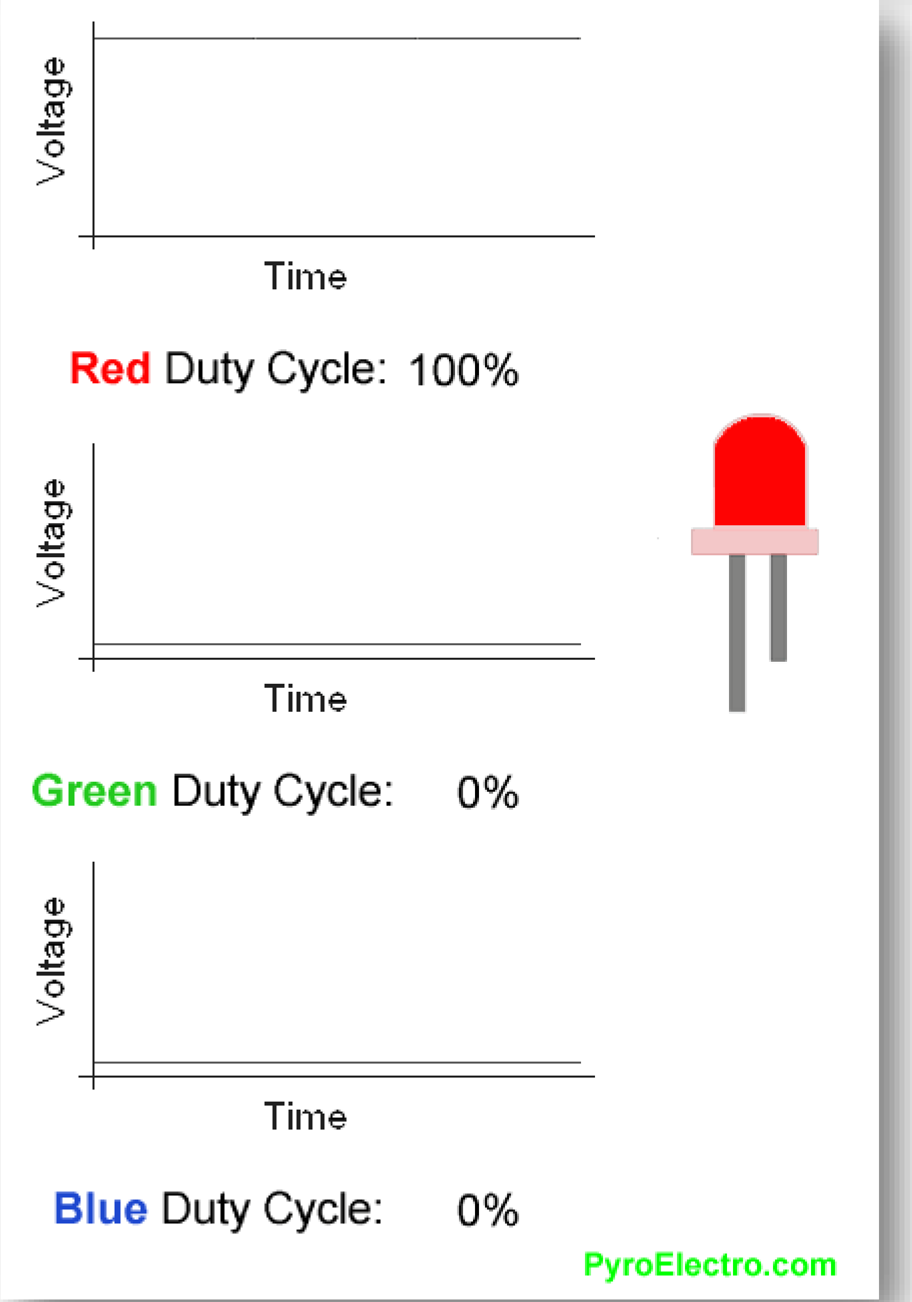
DISADVANTAGES OF PWM

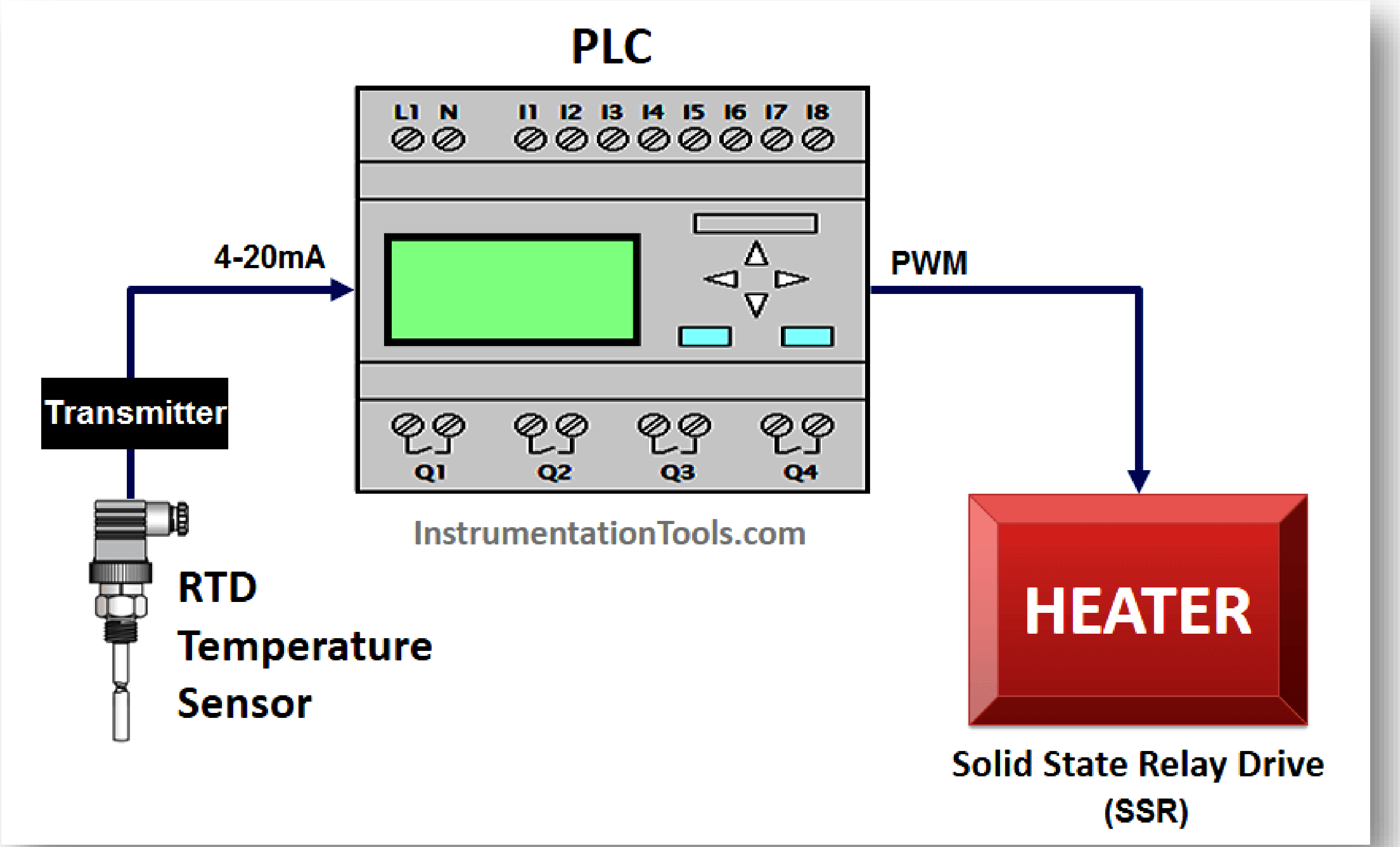
- Instability in the transmitter power system due to variable time intervals


Voltage
Time
Duty Cycle: 05


Spektallinien
Einhuellende
Tastgrad: 0.013
Vielfache der Grundfrequenz

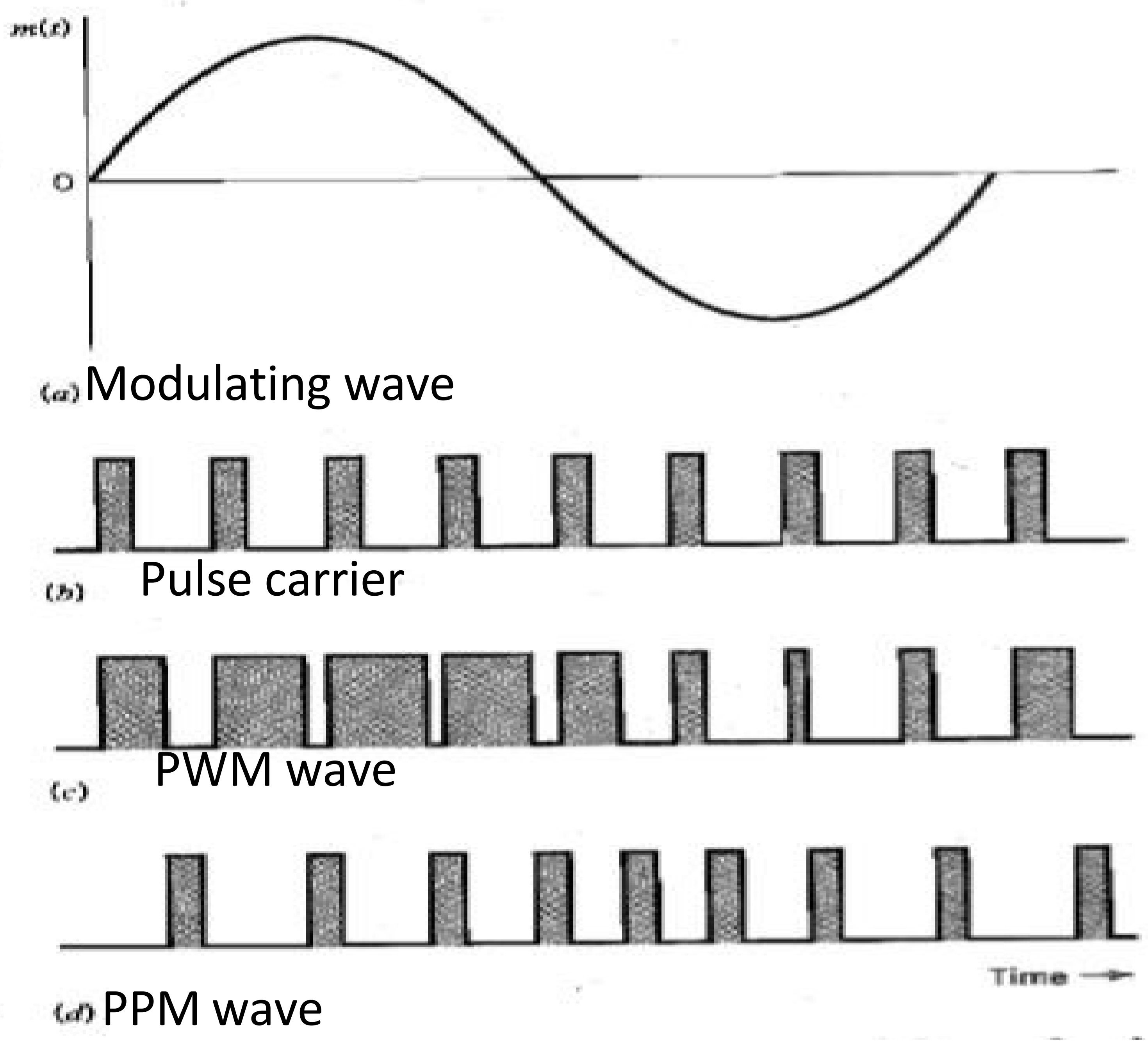

+V
-V
V_{in}
V_{ref}
V_{out}


Voltage
Time
Red Duty Cycle: 100%
Voltage
Time
Green Duty Cycle: 0%
Voltage
Time
Blue Duty Cycle: 0%
PyroElectro.com


PLC
L1 N I1 I2 I3 I4 I5 I6 I7 I8
Q1 Q2 Q3 Q4
4-20mA
PWM
Transmitter
RTD
Temperature
Sensor
InstrumentationTools.com
HEATER
Solid State Relay Drive
(SSR)

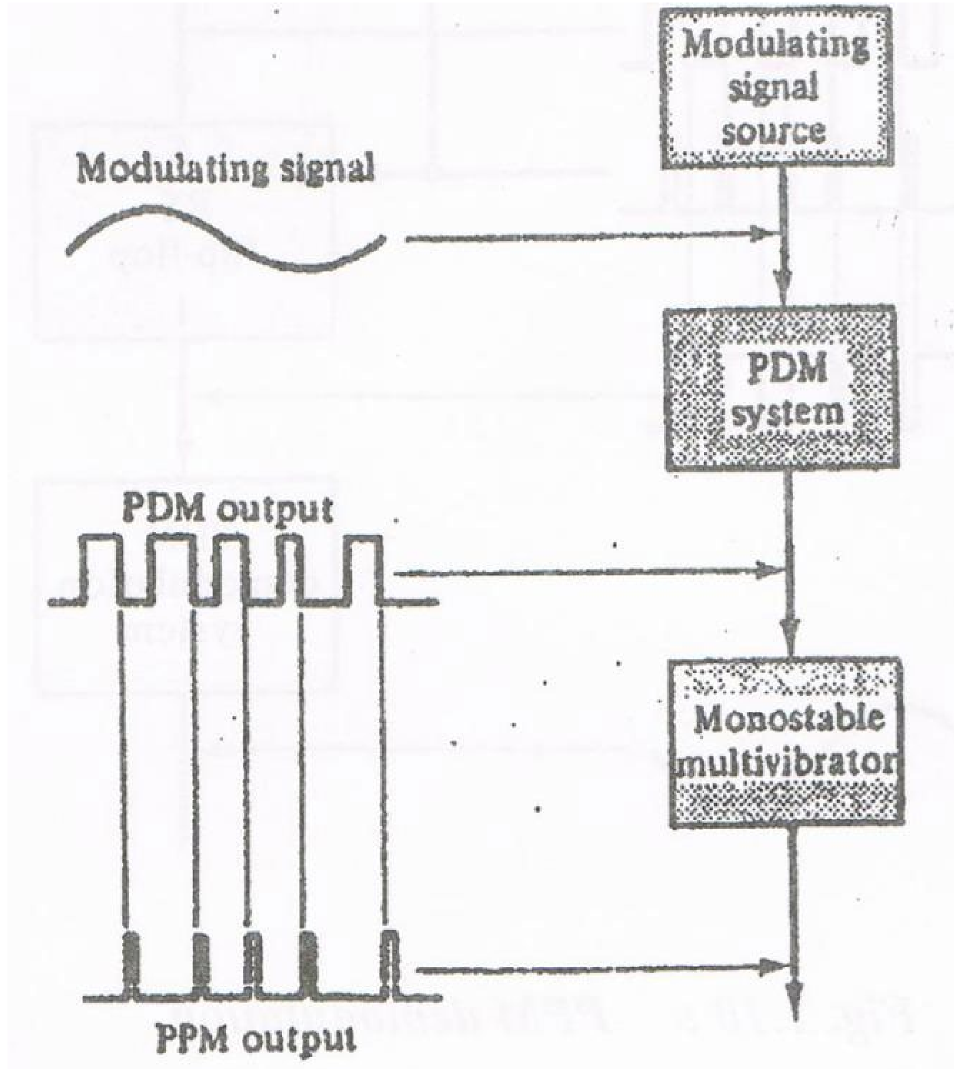
PULSE POSITION MODULATION (PPM)

In PPM, the position of the pulse relative to its un-modulated time occurrence is varied by the message signal.



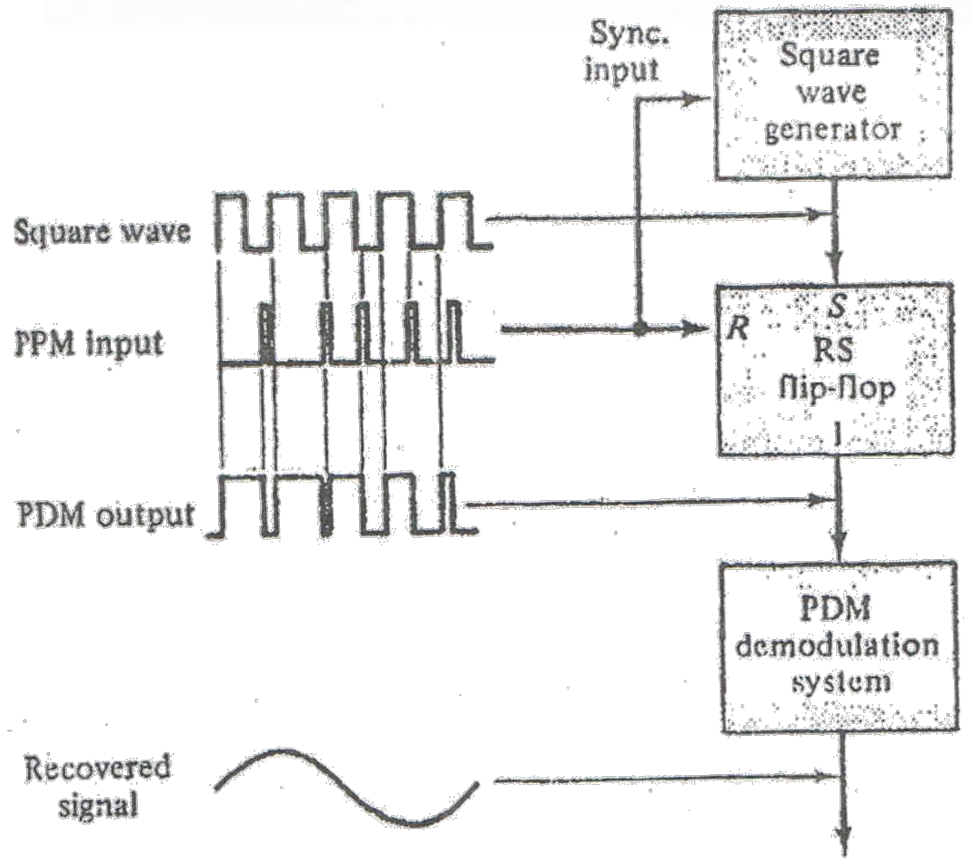
GENERATION OF PPM SIGNAL

- PPM generator consists of a differentiator and monostable multivibrator.
- The differentiator generates positive and negative spikes corresponding to the leading and trailing edges of the PWM waveform.



DEMODULATION OF PPM SIGNAL

First, PPM is converted to PWM by using an RS flip-flop. Then a PWM demodulator is used to recover the modulating signal.

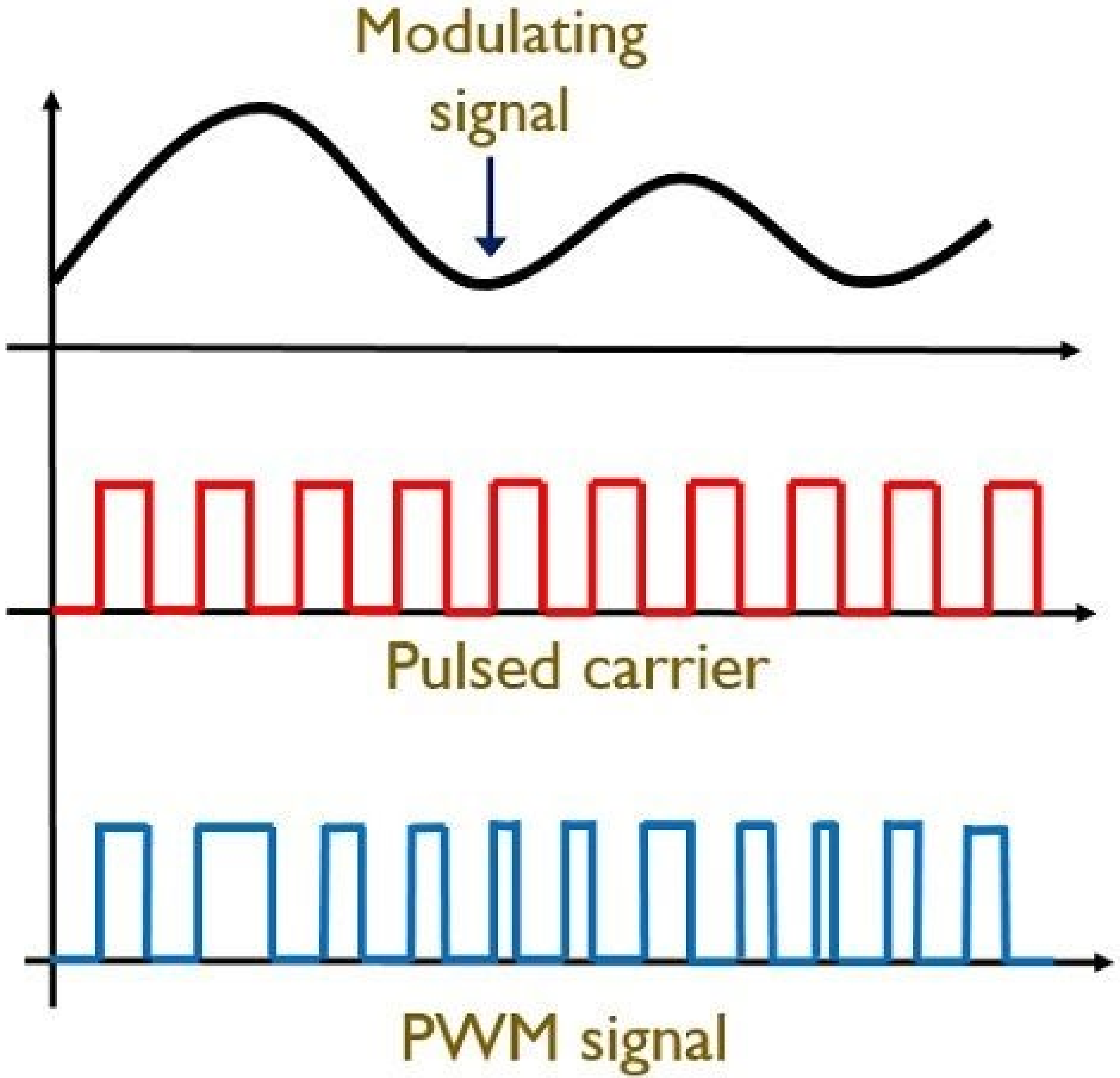


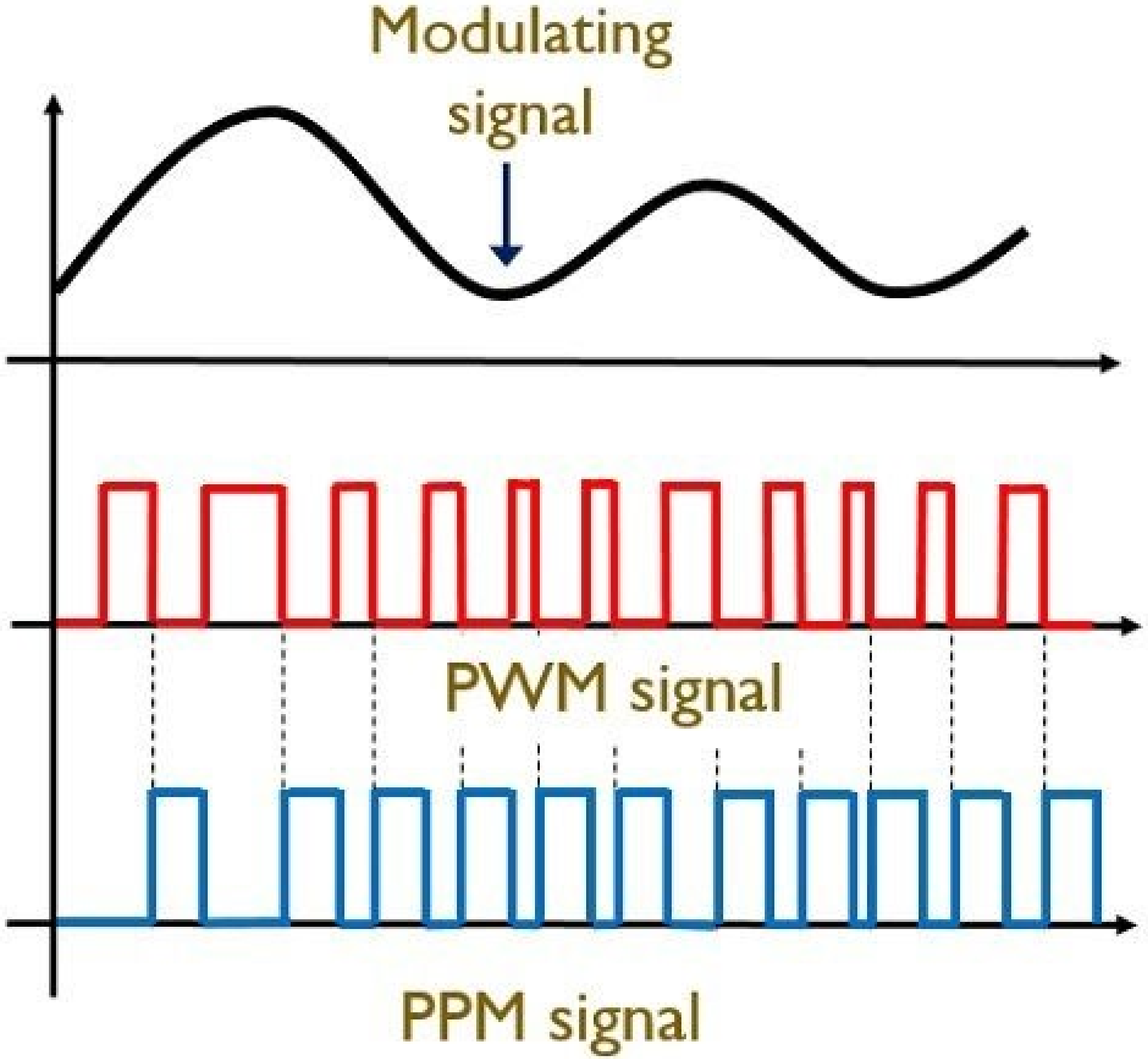
Advantages of PPM

- High noise immunity where modulated pulses are constant in amplitude.
- Stable in the transmitter power system.
- Optimum power saving, since modulated pulses have fixed amplitude and duration.

Disadvantages of PPM

- Required synchroniszation between modulator and demodulator.


Modulating
signal
Pulsed carrier
PWM signal


Modulating
signal
PWM signal
PPM signal

COMPARISON BETWEEN PAM, PWM & PPM

S. No	PAM	PWM/PDM	PPM
1	The amplitude of the pulse is proportional to the amplitude of the modulating signal	The width of the pulse is proportional to amplitude of the modulating signal.	The relative position of the pulse is proportional to the amplitude of the modulating signal.
2	The bandwidth of the transmission channel depends on width of the pulse	The bandwidth of transmission channel depends on rise time of the pulse.	The bandwidth of transmission channel depends on rise time of the pulse.
3	The instantaneous power of the transmitter varies with amplitude of pulses.	The instantaneous power of the transmitter varies with width of pulses	The instantaneous power of the transmitter remains constant with width of pulses.
4	Noise interference is high	Noise interference is minimum	Noise interference is minimum
5	System is complex	Implement is simple	Implement is simple
6	Similar to Amplitude modulation	Similar to Frequency modulation	Simple to Phase modulation

Problems

Example: For a pulse-amplitude modulated (PAM) transmission of voice signal having maximum frequency equal to fm = 3kHz, calculate the transmission bandwidth. It is given that the sampling frequency fs = 8 kHz and the pulse duration T = 0.1Ts.

Solution

Solution: The sampling period Ts is expressed as

$$T_s = \frac{1}{f_s} \quad \text{or} \quad T_s = \frac{1}{8\times 10^3}\text{seconds}$$

$$T_s = 0.125\times 10^{-3} \text{ seconds}$$

$$T_s = 125\,\mu\text{seconds}$$

$$\tau = 0.1\ T_s$$

$$\tau = 0.1 \times 125 = 12.5\,\mu \text{ seconds}$$

$$BW \geq \frac{1}{2\tau}$$

$$BW \geq \frac{1}{2 \times 12.5 \times 10^{-6}}$$

$$\geq \frac{1 \times 10^{6}}{25}$$

$$BW \geq 40\ kHz$$

Example: 5 message signals, each having a frequency of 2 KHz, are multiplexed using TDM. Number of quantization levels used are 256. Find the transmission bandwidth of the system.

Solution

Given 5 message signals (N=5), fm = 2 KHz,

L = 256 levels then nb = 8 We know that NR = 2fm

fs = NR = 2fm = 4 KHz

Bandwidth=Rb/2

=nbNfs/2

=8*5*4kHz/2

=80kHz

Example: 10 signals, each band limited to 2KHz are multiplexed using TDM. The time taken by the commutator to make one complete rotation is 125 microseconds. Find the bit rate of the transmitter if 5 bits encoder is used.

Solution

Given that

N = 10, fm = 2 KHz, Ts = 125 microseconds

nb = 5 bits/sample

fs=1/Ts=1/125micro=8kHz

Rb=nbNfs=5*10*8kHz=400kbps

College of Electronics Engineering

Systems & Control Engineering Department

Digital Communications (SCE3316)

Lecture 4
(Line Coding)

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INTRODUCTION

Binary data can be transmitted using a number of different types of pulses. The choice of a particular pair of pulses to represent the symbols 1 and 0 is called Line Coding and the choice is generally made on the grounds of one or more of the following considerations:

- ***Presence or absence of a DC level.***
- ***Power Spectral Density.***
- ***Bandwidth.***
- ***BER performance.***

- ***Transparency (i.e. the property that any arbitrary symbol, or bit pattern can be transmitted and received).***
- ***Ease of clock signal recovery for symbol synchronisation.***
- ***Presence or absence of inherent error detection properties.***

LINE CODING

***Line coding* is the process of converting digital data to digital signals.**

We assume that data, in the form of text, numbers, graphical images, audio, or video, are stored in computer memory as sequences of bits.

Line coding converts a sequence of bits to a digital signal. At the sender, digital data are encoded into a digital signal; at the receiver, the digital data are recreated by decoding the digital signal.

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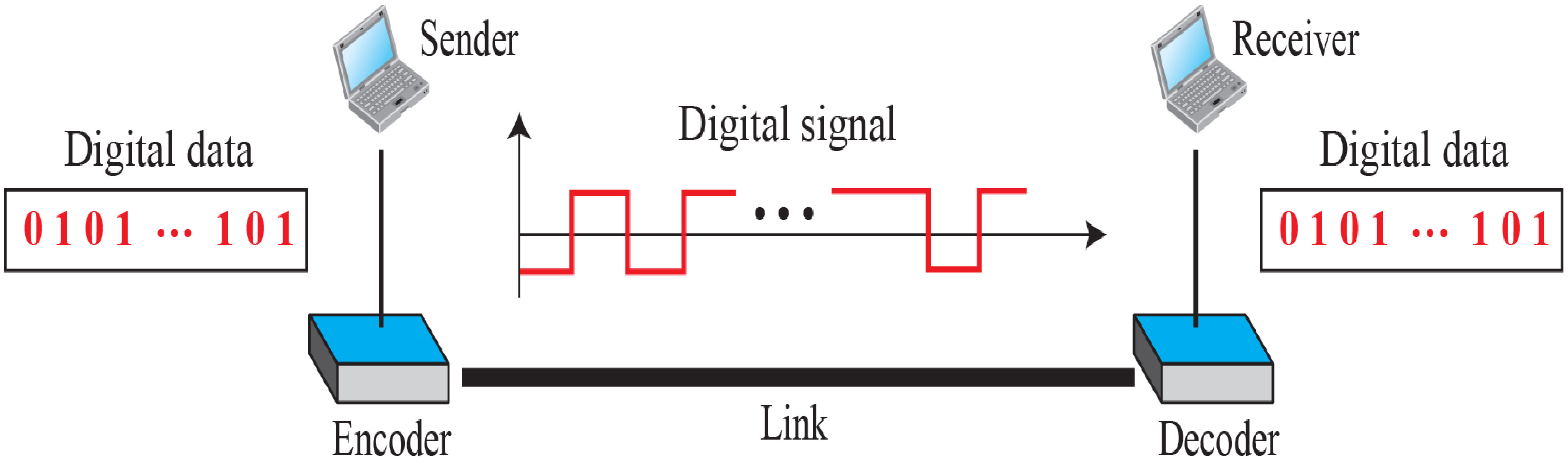
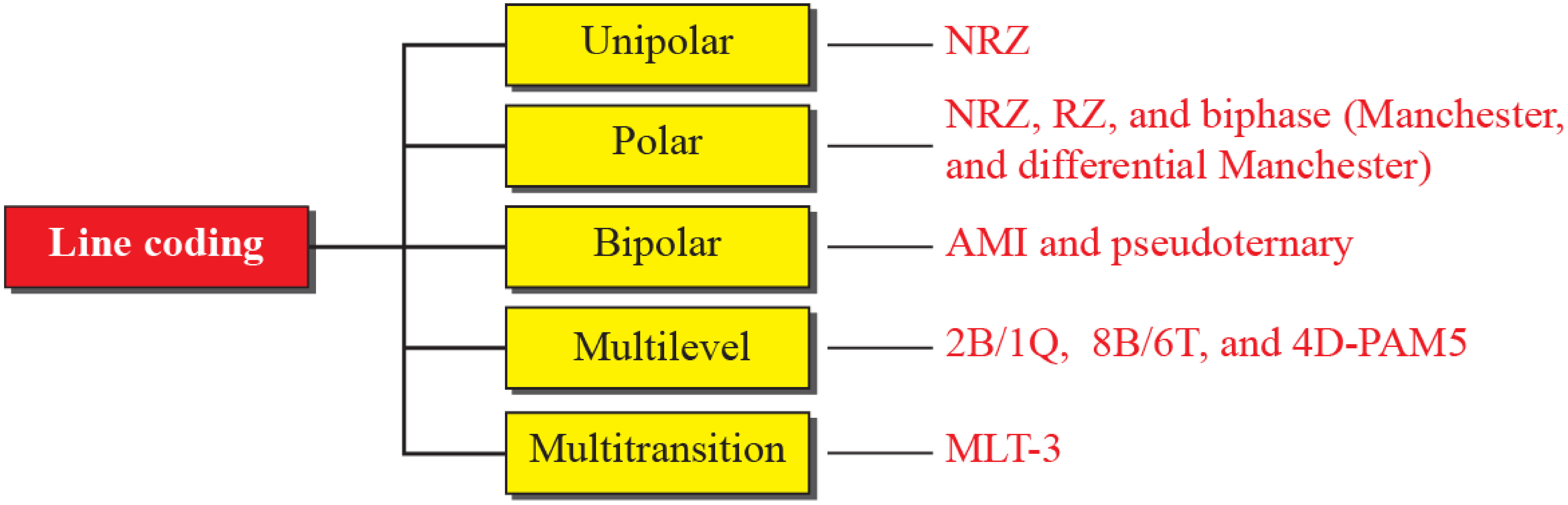

Sender
Receiver
Digital signal
Digital data
0 1 0 1 … 1 0 1
Digital data
0 1 0 1 … 1 0 1
Encoder
Link
Decoder


Line coding
Unipolar
Polar
Bipolar
Multilevel
Multitransition
NRZ
NRZ, RZ, and biphase (Manchester, and differential Manchester)
AMI and pseudoternary
2B/1Q, 8B/6T, and 4D-PAM5
MLT-3

In digital data communications, a signal element carries data elements.

***Data rate** defines the number of data elements (bits) sent in 1s, which is called **bit rate**. The unit is bits per second (bps).*

***Signal rate** is the number of signal elements sent in 1s, that is called also **baud rate, pulse rate or modulation rate**. The unit is (baud).*

***The ratio r** defines number of bits per baud **(data rate/signal rate) (r=N/S)**.*

Bit rate= baud rate x number of bits per sample

- *One goal in data communications is to increase the data rate while decreasing the signal rate. Increasing the data rate increases the speed of transmission; decreasing the signal rate decreases the bandwidth requirement.*
- $S_{average} = c * N * (\frac{1}{r})$, *where N is the data rate(bps); c is the case factor; which varies for each case; S is the signal rate, r is the previously defined factor.*

EXAMPLE

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 if c is between 0 and 1?

Solution

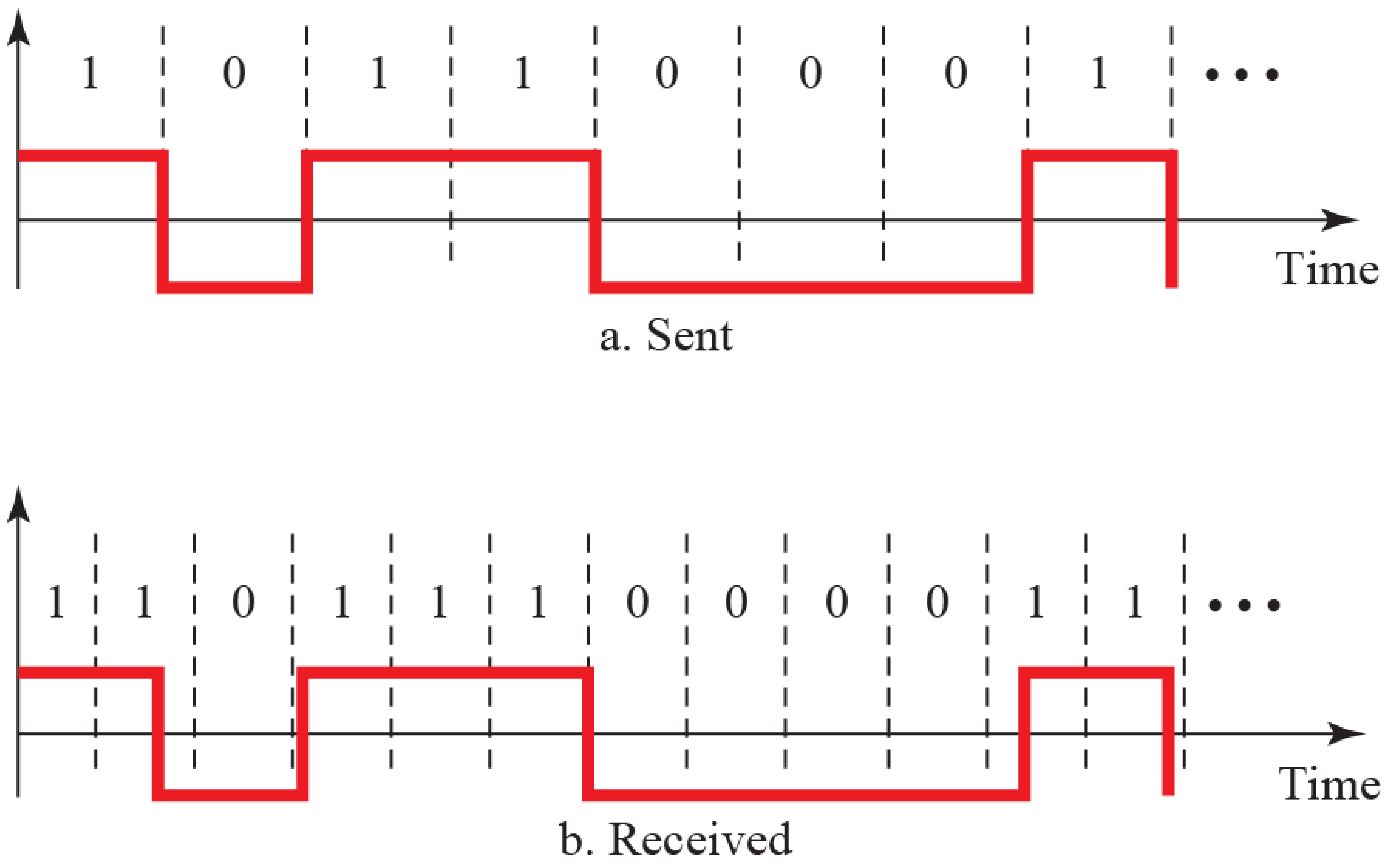
We assume that the average value of c is 1/2. The baud rate is then

$$S = c \times N \times (1 / r) = 1/2 \times 100{,}000 \times (1/1) = 50{,}000 = 50 \text{ kbaud}$$

SELF SYNCHRONIZATION

- ***To correctly interpret the signals received from the sender, the receiver's bit intervals must correspond exactly to the sender's bit intervals.***

- ***The next figure shows a situation in which the receiver has a shorter bit duration. The sender sends 10110001, while the receiver receives 110111000011. This is called lack of synchronization.***



Effect of lack of synchronization

DC COMPONENT

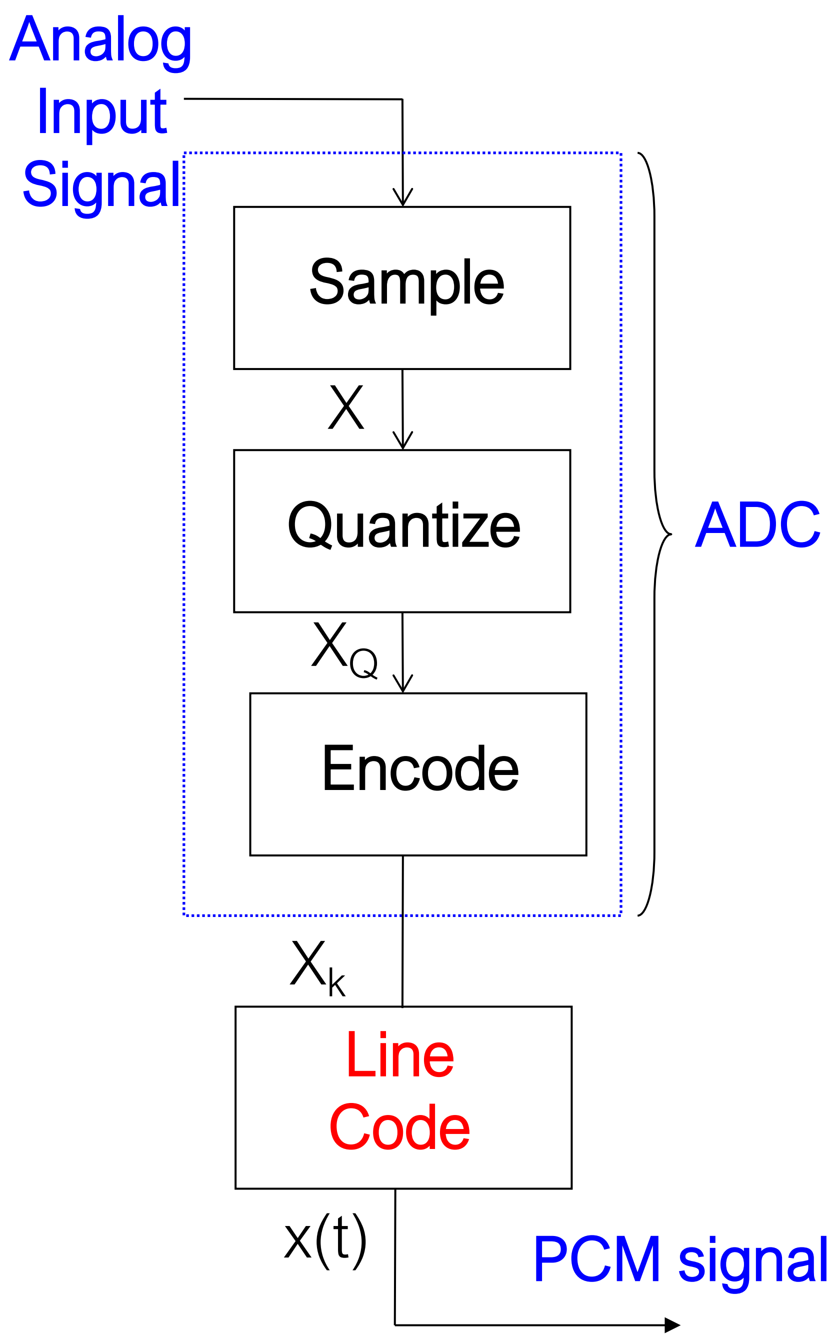
- ***Over one cycle period of a waveform, if all the positive voltages are canceled by the negative voltages, then DC component of the waveform is zero.***

- ***In line coding, the signal with non-zero DC component is treated as distorted one and it can create error in received signal.***

LINE CODES IN PCM

The output of an ADC can be transmitted over a baseband channel.

- The digital information must first be converted into a physical signal.
- The physical signal is called a **line code**.

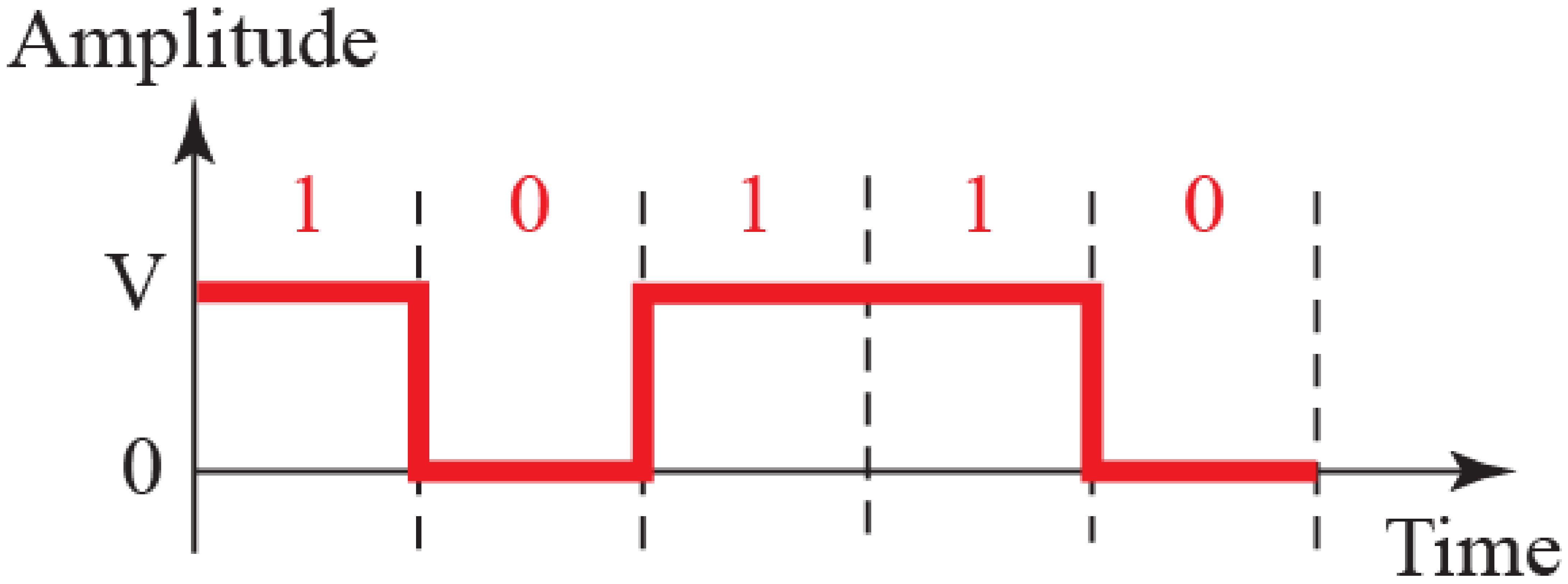


UNIPOLAR SIGNALLING

- Unipolar signalling (also called on-off keying, OOK) is the type of line coding in which one binary symbol (representing a 0 for example) is represented by the absence of a pulse (i.e. a SPACE) and the other binary symbol (denoting as 1) is represented by the presence of a pulse (i.e. a MARK).
- There are two common variations of unipolar signalling: Non-Return to Zero (NRZ) and Return to Zero (RZ).

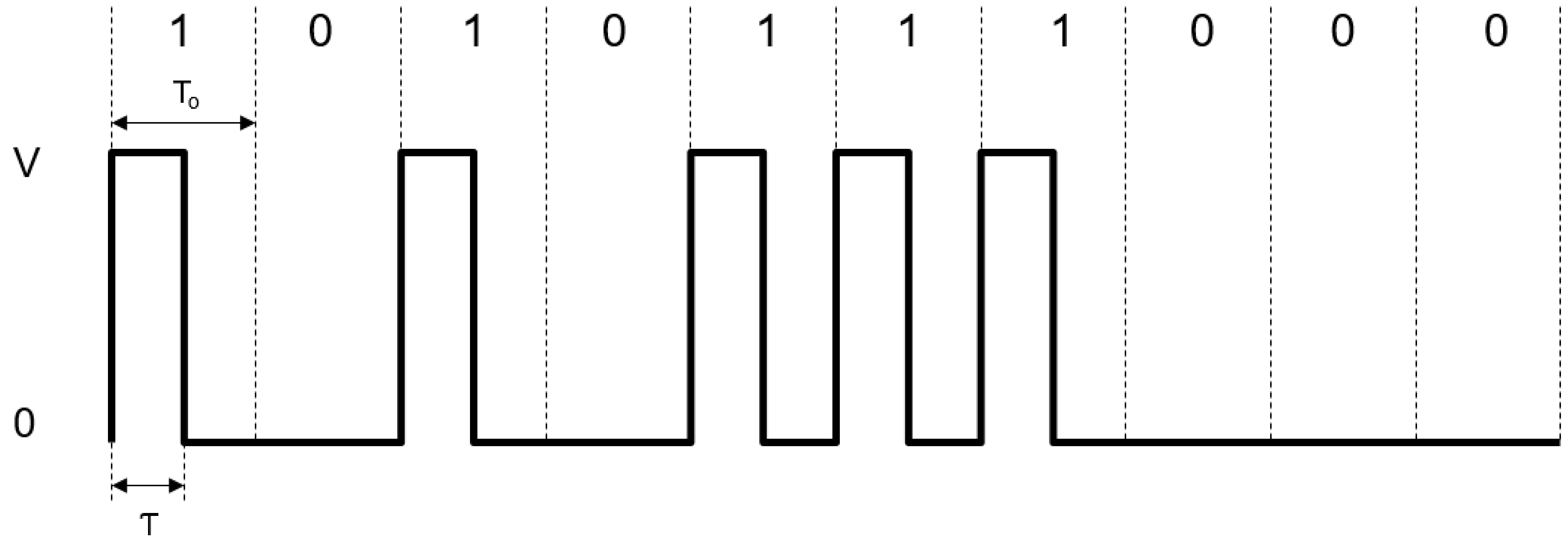
Unipolar Non-Return to Zero (NRZ):

- In unipolar NRZ, the duration of the MARK pulse (τ) is equal to the duration (To) of the symbol slot.



Return to Zero (RZ):

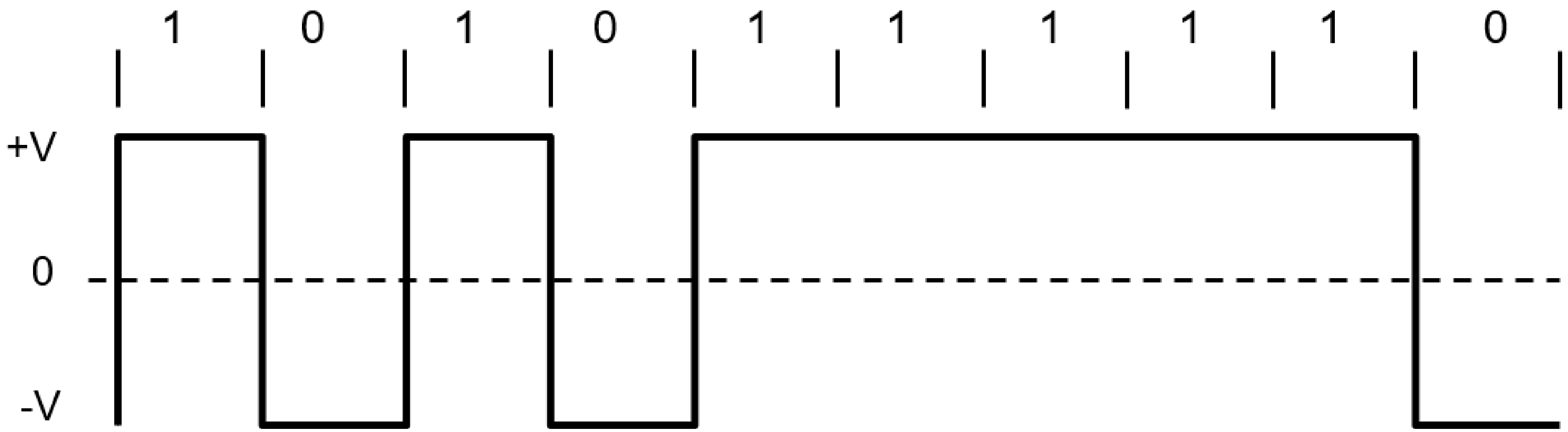
- In unipolar RZ, the duration of the MARK pulse (ꚍ) is less than the duration (To) of the symbol slot. Typically, RZ pulses fill only the first half of the time slot, returning to zero for the second half.



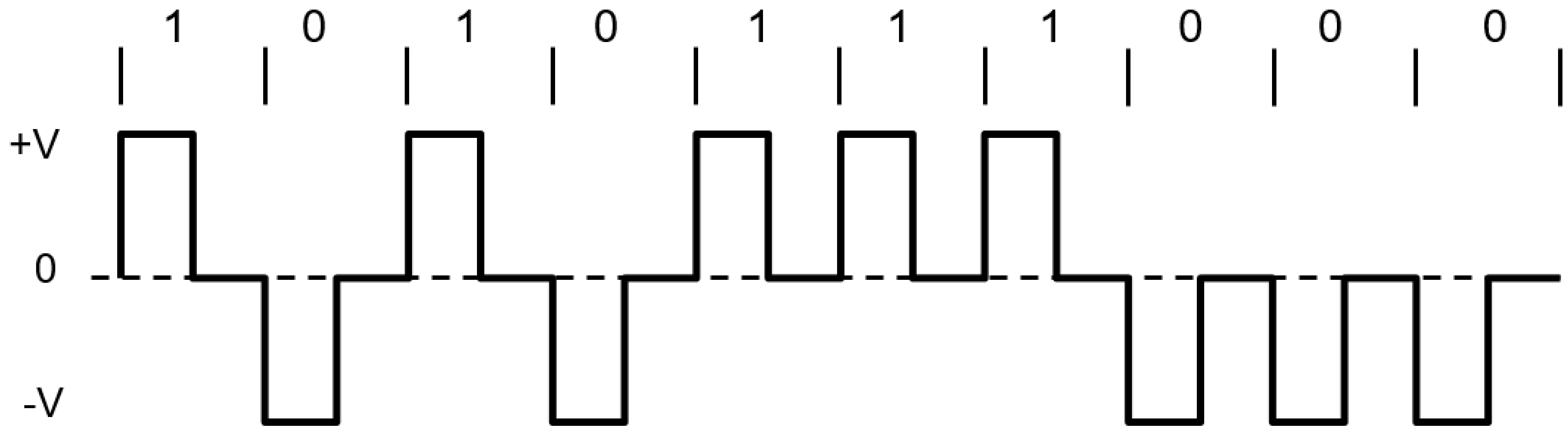
POLAR SIGNALLING

- In polar signalling, a binary 1 is represented by a pulse g1(t) and a binary 0 by the opposite (or antipodal) pulse g0(t) = -g1(t). Polar signalling also has NRZ and RZ forms.

Polar Non-Return to Zero (NRZ):



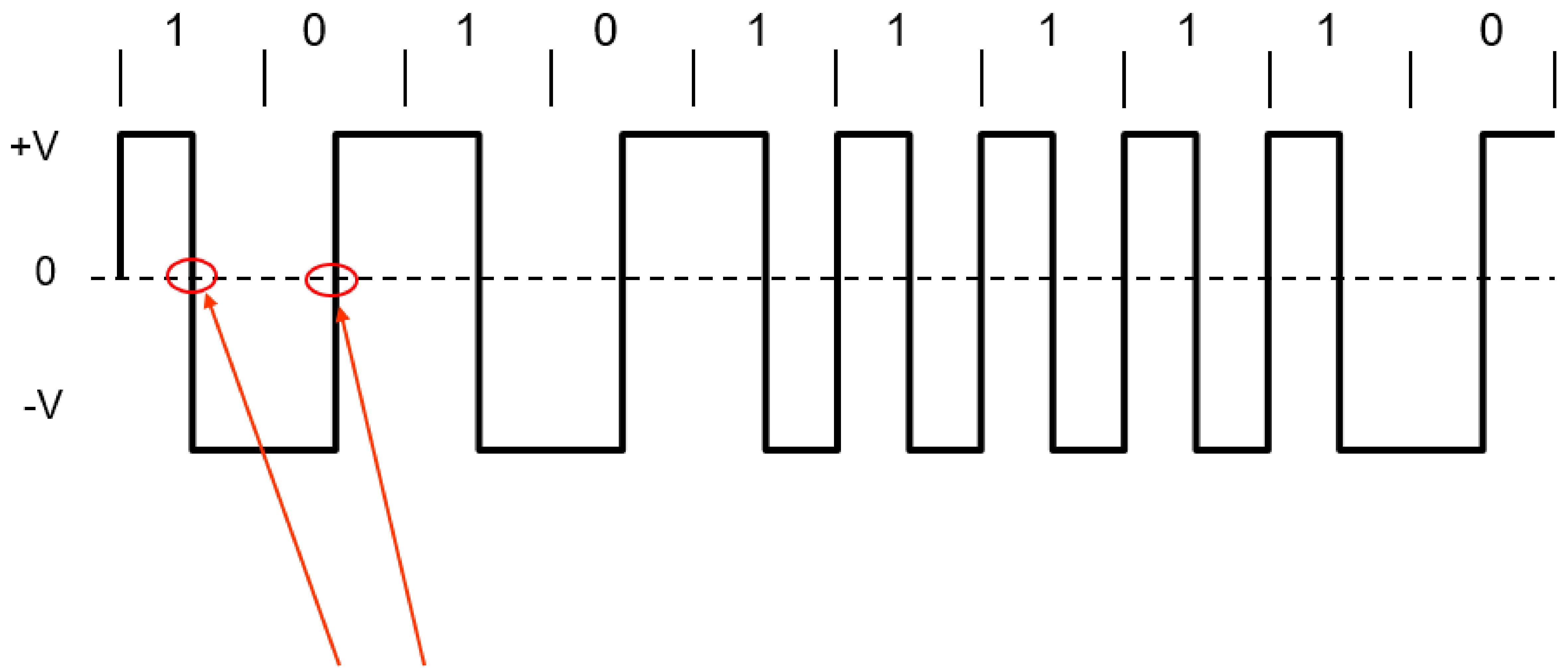
Polar Return to Zero (RZ):



Polar Manchester:

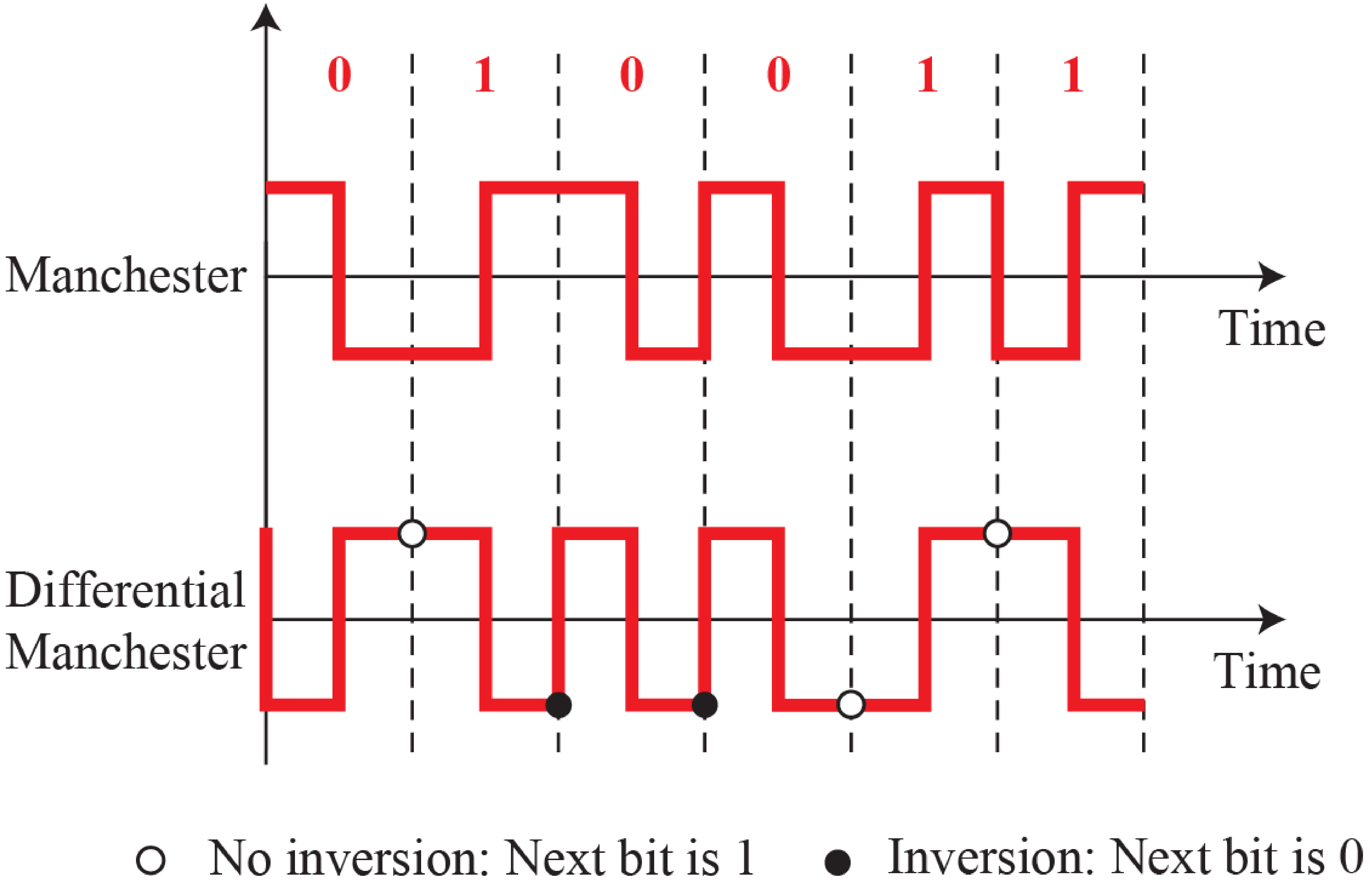
In Manchester encoding , the duration of the bit is divided into two halves. The voltage remains at one level during the first half and moves to the other level during the second half.

- ‘One’ is +ve in 1st half and -ve in 2nd half.
- ‘Zero’ is -ve in 1st half and +ve in 2nd half.



Note: There is always a transition at the centre of bit duration.

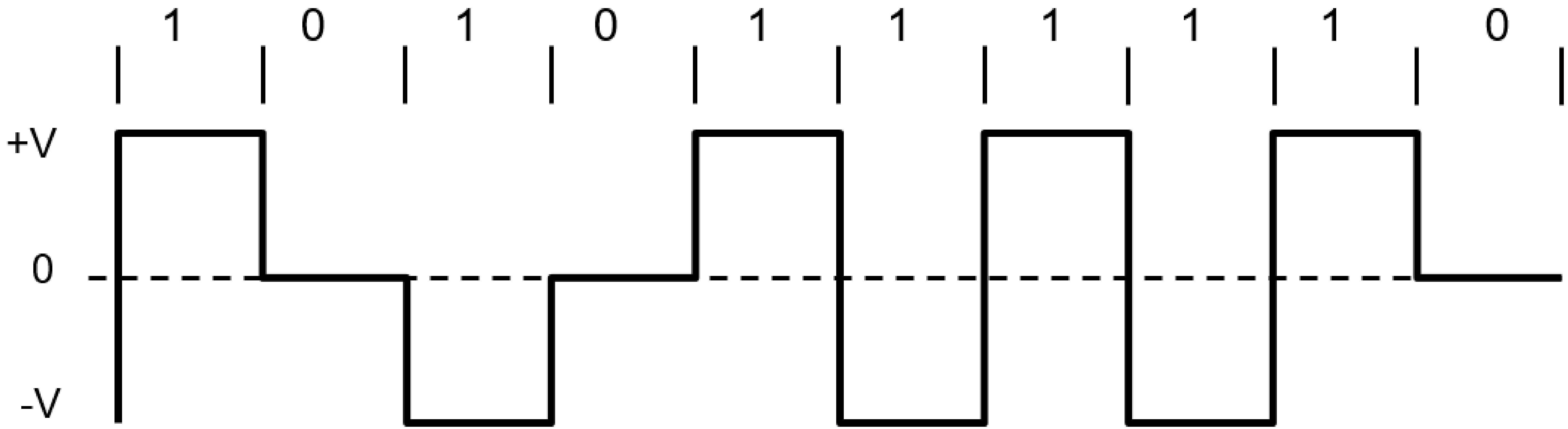
- **The transition at the centre of every bit interval is used for synchronization at the receiver.**
- **Manchester encoding is called self-synchronizing. Synchronization at the receiving end can be achieved by locking on to the transitions, which indicate the middle of the bits.**


0
1
0
0
1
1
Manchester
Time
Differential Manchester
Time
No inversion: Next bit is 1
Inversion: Next bit is 0

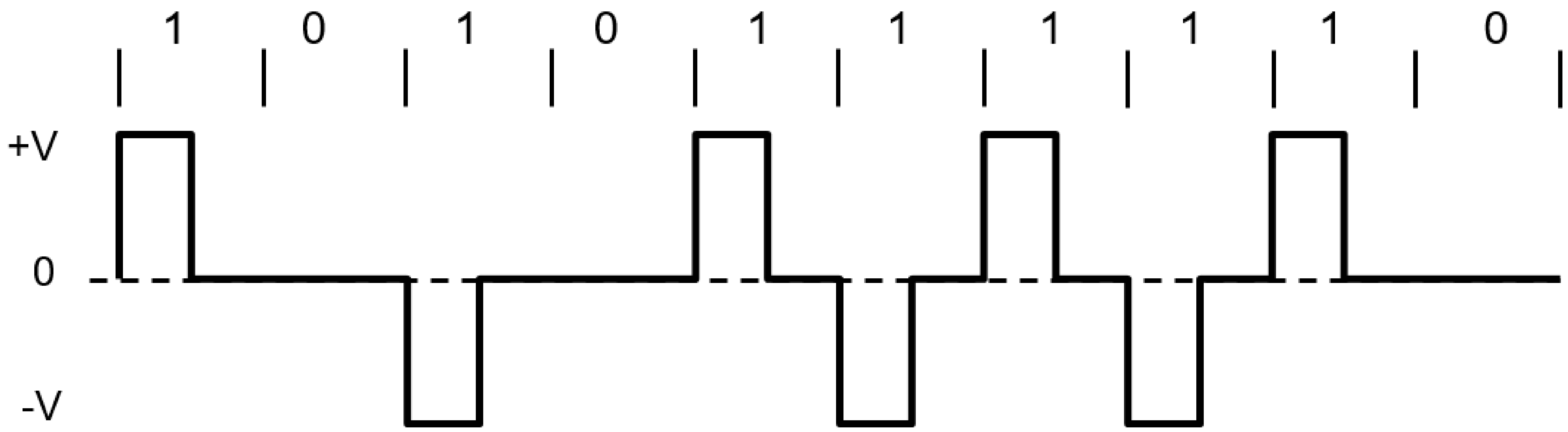
BIPOLAR SIGNALLING

- Bipolar Signalling is also called “alternate mark inversion” (AMI) uses three voltage levels (+V, 0, -V) to represent two binary symbols. Zeros, as in unipolar, are represented by the absence of a pulse and ones (or marks) are represented by alternating voltage levels of +V and –V.
- Alternating the mark level voltage ensures that the bipolar spectrum has a null at DC.
- Like the Unipolar and Polar cases, Bipolar also has NRZ and RZ variations.

Bipolar Non-Return to Zero (NRZ):



Bipolar Return to Zero (RZ):



College of Electronics Engineering

Systems & Control Engineering Department

Digital Communications (SCE3316)

Lectures 5 and 6
(Digital to Analog Conversion)

INTRODUCTION

- Digital modulation is used to transfer a digital bit stream over an analog channel at a high frequency.

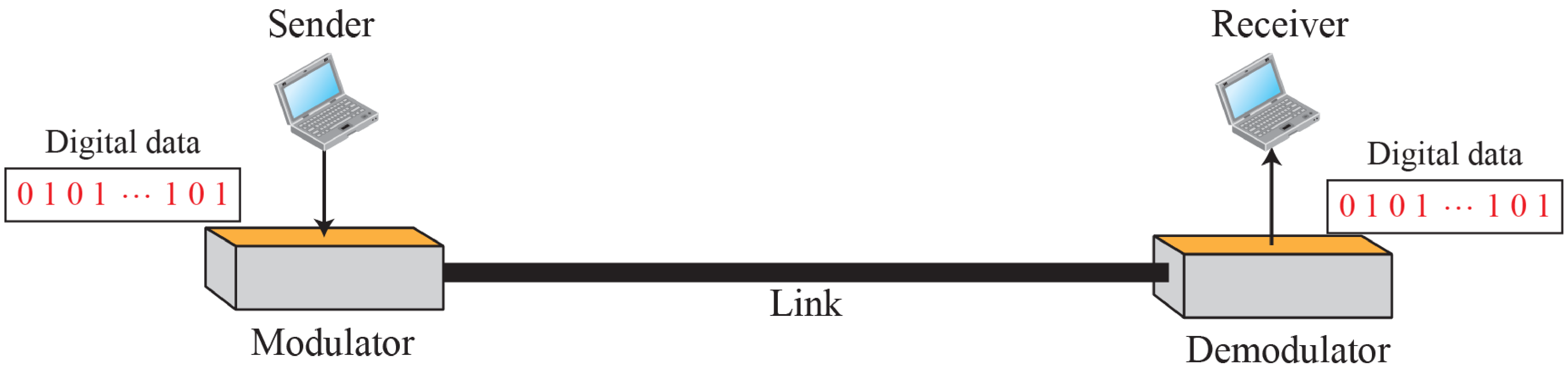
- This enables us to transmit signals generated in a digital circuit across a physical medium. This is because digital signals can be handled with higher security and digital systems are readily and widely available.

  Depending on which parameter of the carrier signal is varied in accordance with the digital message signal, we obtain three main variants of digital modulation called Amplitude Shift Keying (ASK), Frequency Shift Keying (FSK) and Phase Shift Keying (PSK).

INTRODUCTION

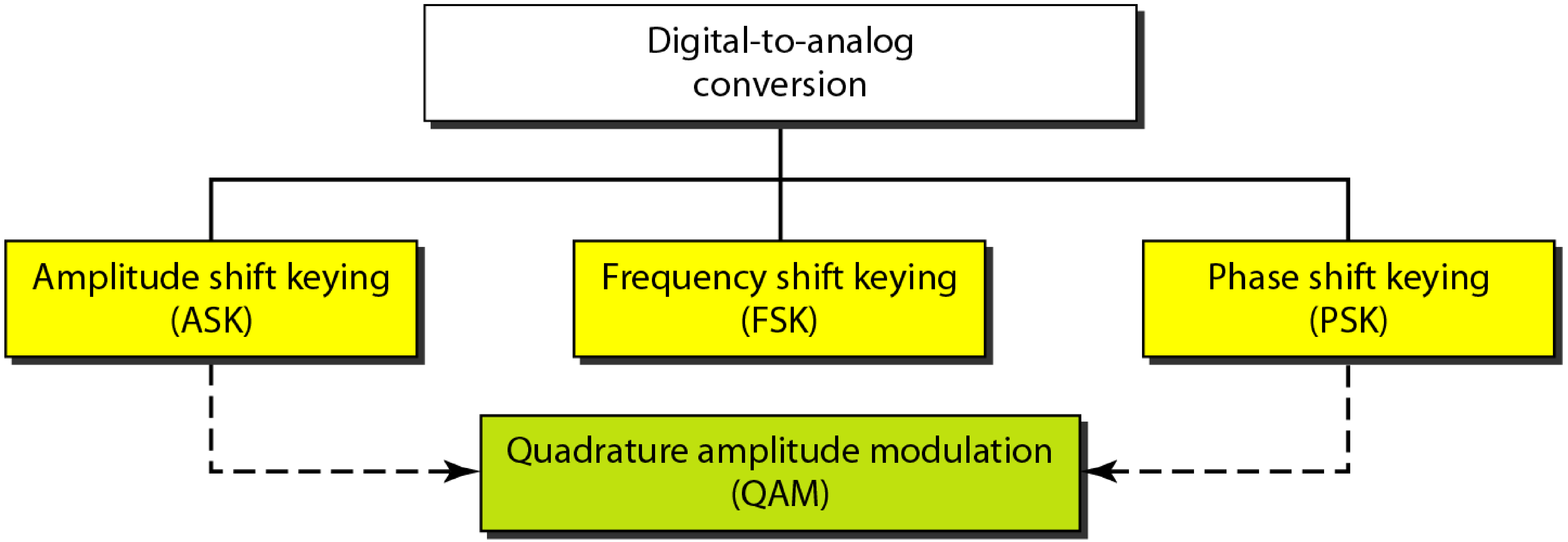
- Digital-to-analog conversion is the process of changing one of the characteristics of an analog signal based on the information in digital data.

- The figure shown below shows the relationship between the digital information, the digital-to-analog modulating process, and the resultant analog signal.



Digital-to-analog conversion

INTRODUCTION



Types of digital to analog conversion

ASPECTS OF CONVERSION

Before we discuss specific methods of digital-to-analog modulation, two basic issues must be reviewed: bit and baud rates and the carrier signal.

Example:

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

Solution

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

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

TYPES OF TRANSMISSION OF DIGITAL SIGNALS:

- Baseband Transmission
- Passband Transmission

Types of Reception of Passband Transmission:

- Coherent (Synchronous) detection
- Noncoherent (Envelope) detection

Requirements of Passband Transmission:

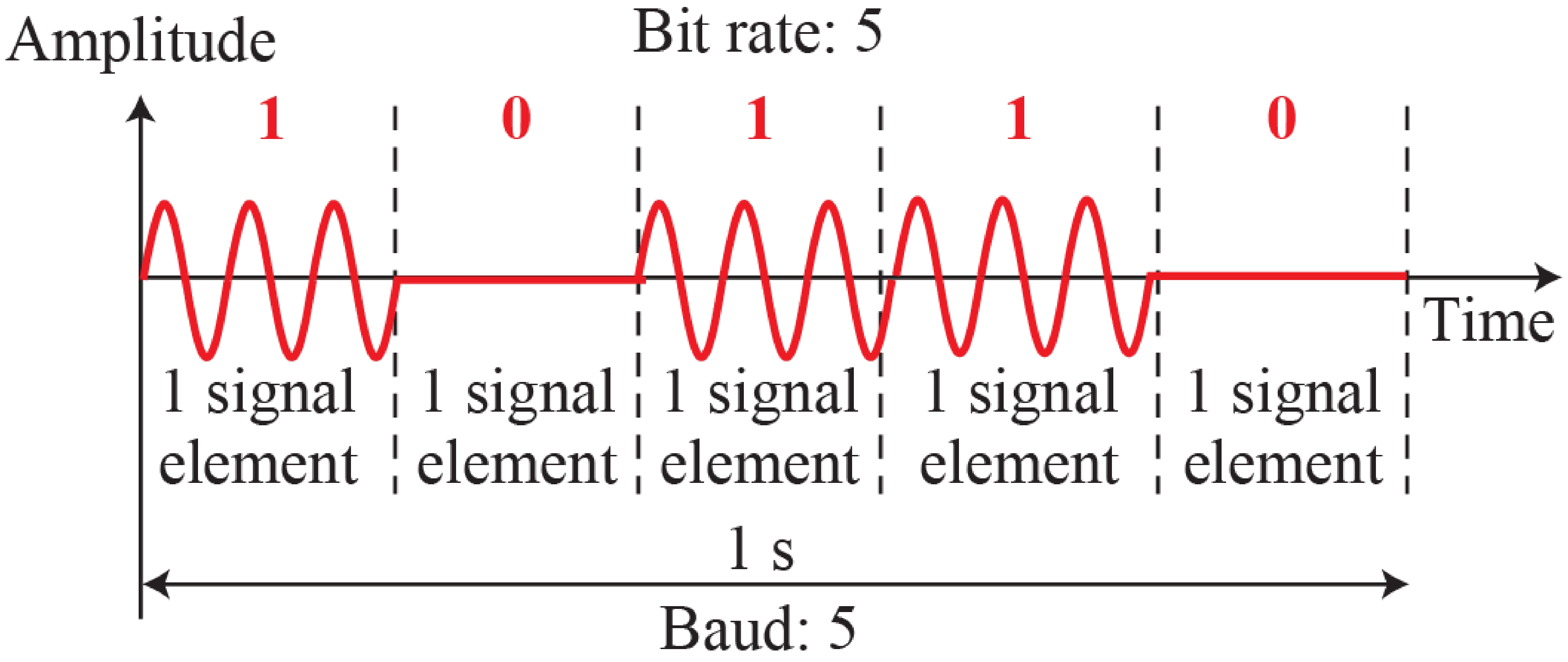
- Maximum data transmission rate.
- Minimum probability of symbol (bit) error.
- Minimum transmitted power.
- Minimum channel bandwidth.

Advantages of Passband over Baseband transmission:

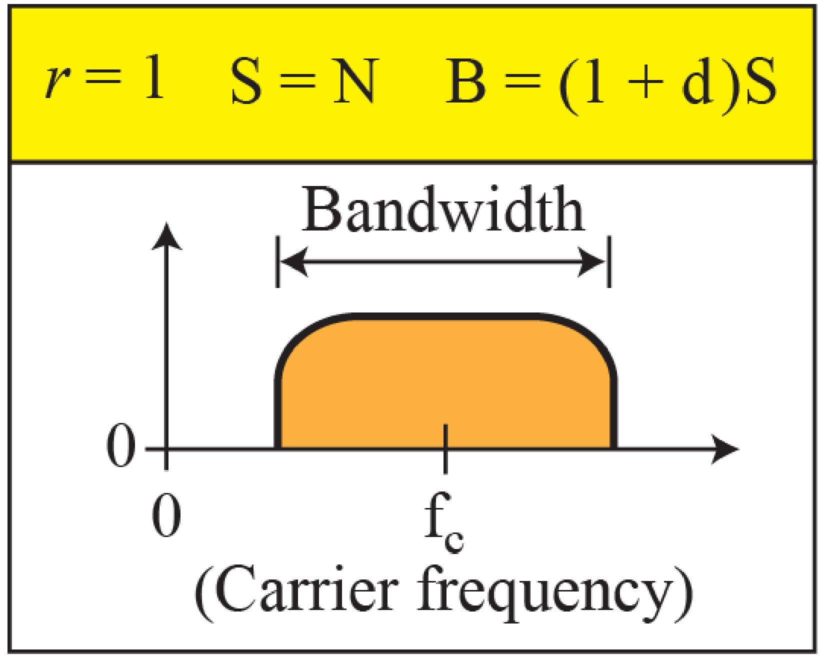
1) Long distance transmission.
2) Large number of modulation techniques are available.

AMPLITUDE SHIFT KEYING

In amplitude shift keying, the amplitude of the carrier signal is varied to create signal elements. Both frequency and phase remain constant while the amplitude changes. It is digital to analog conversion technique.

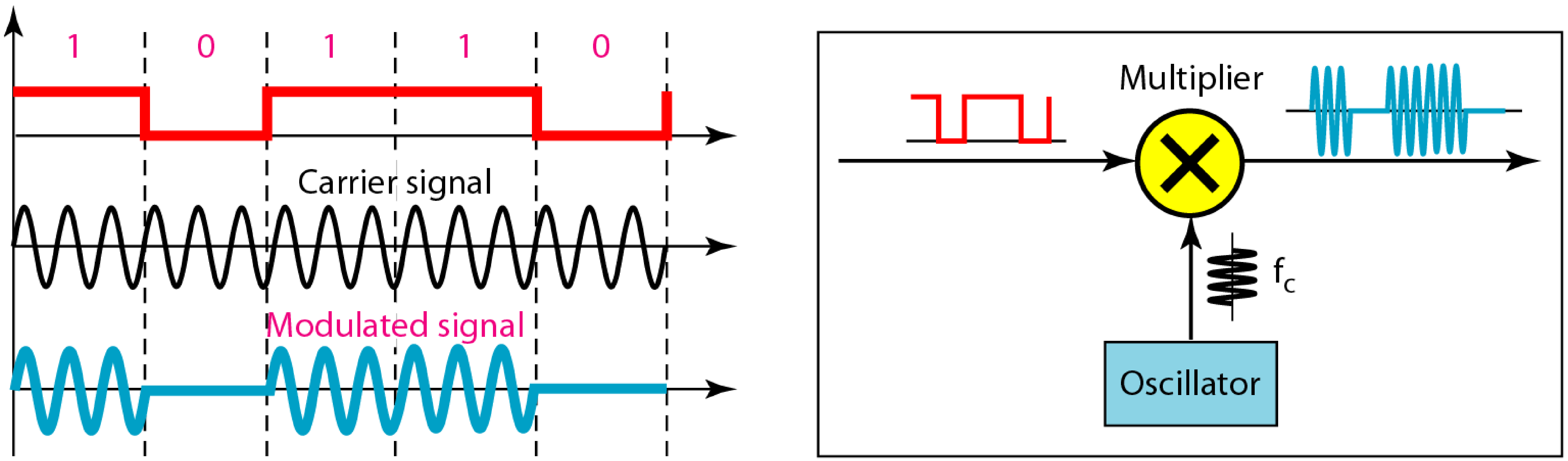


Binary amplitude shift keying



d is a factor of modulation and filtering process.

AMPLITUDE SHIFT KEYING



Implementation of binary ASK

AMPLITUDE SHIFT KEYING

Example: We have an available bandwidth of 100 kHz which spans from 200 to 300 kHz. What are the carrier frequency and the bit rate if we modulated our data by using ASK with d = 1? Note that half-duplex is used.

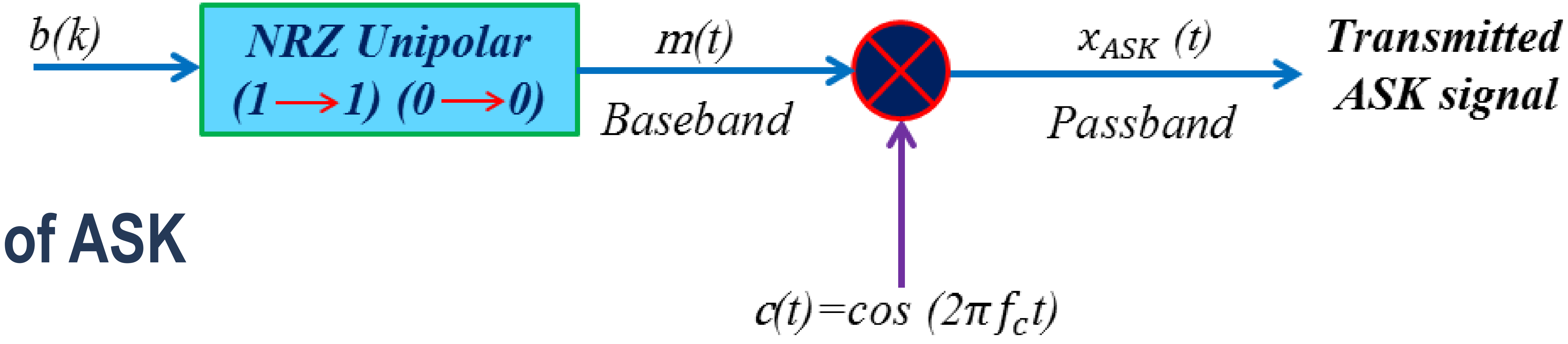
Solution

The middle of the bandwidth is located at 250 kHz. This means that our carrier frequency can be at fc = 250 kHz. We can use the formula for bandwidth to find the bit rate (with d = 1 and r = 1).

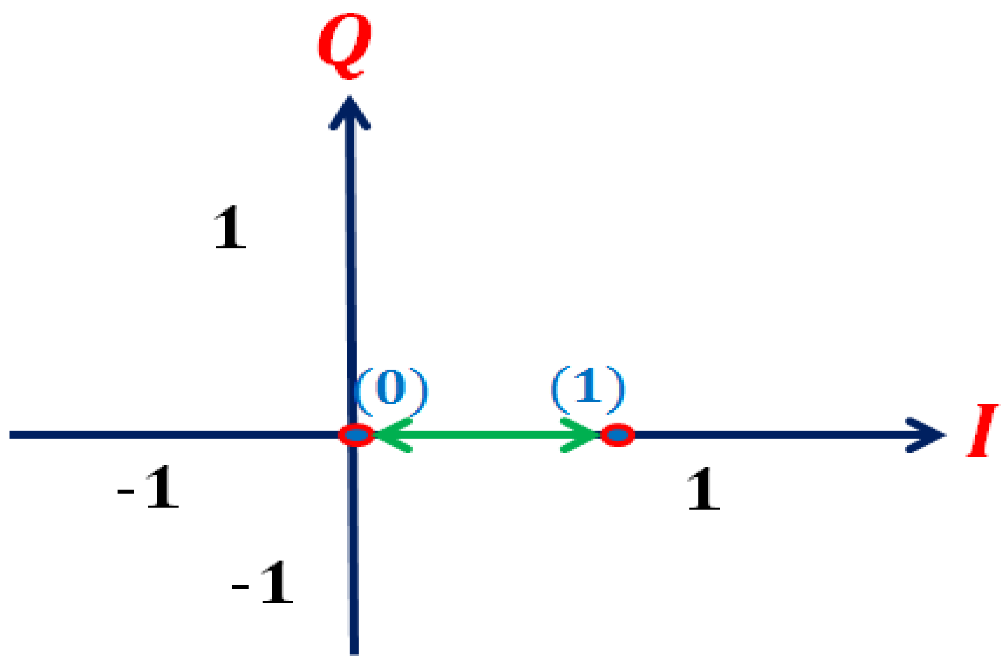
$$B = (1 + d) \times S = 2 \times N \times (1/r) = 2 \times N = 100 \text{ kHz} \longrightarrow N = 50 \text{ kbps}$$

AMPLITUDE SHIFT KEYING

ASK Transmitter



Constellation diagram of ASK signal



- I: Inphase channel
- Q: Quadrature channel
- Bit combinations are assigned in red.

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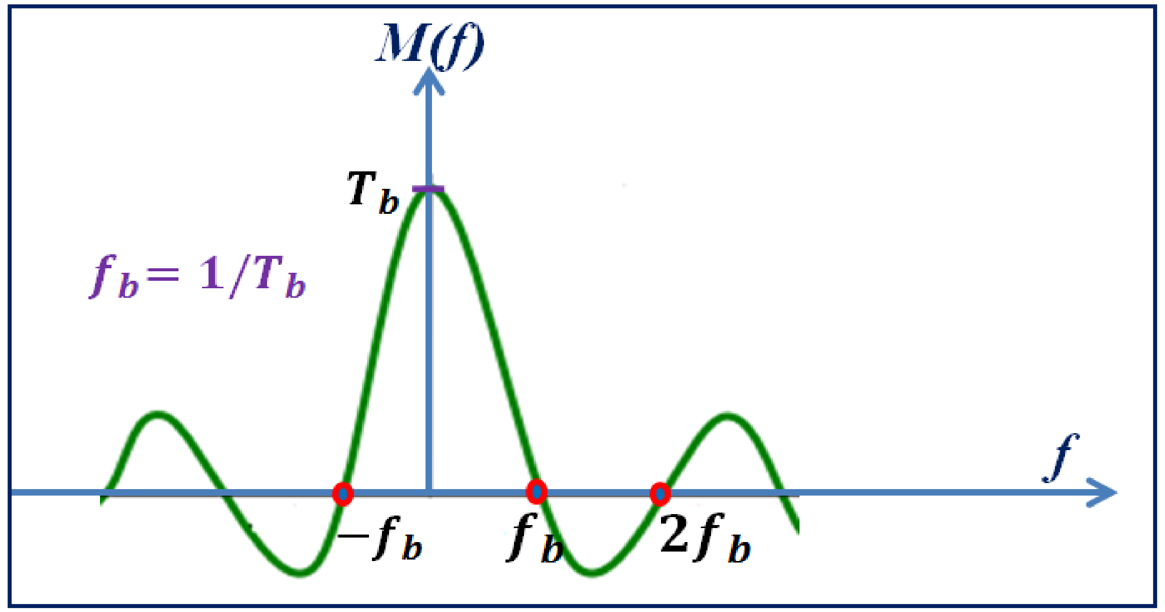
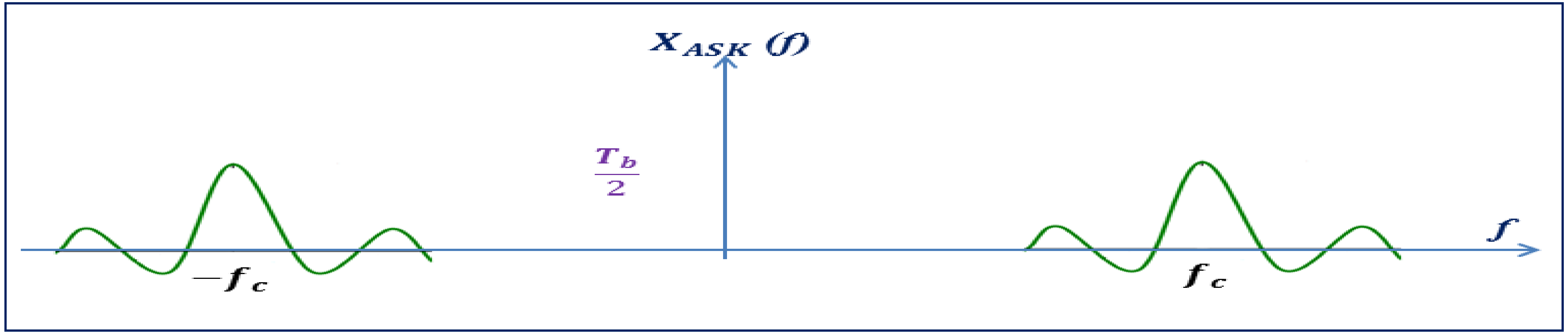
Spectrum of ASK signal

Baseband	Passband
$m(t) = rect\left(\frac{t}{T_b}\right)$	$x_{ASK}(t) = m(t)\cos(2\pi f_c t)$
$M(f) = T_b\, sinc(\pi f T_b)$	$X_{ASK}(f) = \frac{1}{2}M(f-f_c) + \frac{1}{2}M(f+f_c)$





$$R_s = \frac{1}{T_s}$$

R_s: Symbol rate (Baud rate)

T_s: Symbol duration

$$T_s = T_b$$

$$R_b = \frac{1}{T_b}$$

R_b: Bit rate

T_b: Bit duration

$$R_s = R_b$$

One bit per symbol is transmitted

$$B_{ASK} = 2f_b$$

$$f_b = \frac{1}{T_b}$$

B_{ASK}: Bandwidth

$$\eta_{ASK} = \frac{R_b}{B_{ASK}} = \frac{1/T_b}{2f_b} = \frac{f_b}{2f_b} = \frac{1}{2} \ bit/s/Hz$$

η_{ASK}: Spectral efficiency

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AMPLITUDE SHIFT KEYING

Bandwidth is proportional to baud rate.

B= (1+d) * baud rate, baud rate= bit rate/number of bits per sample.

d is a factor for modulation and filtering process. It varies between 0 and 1.

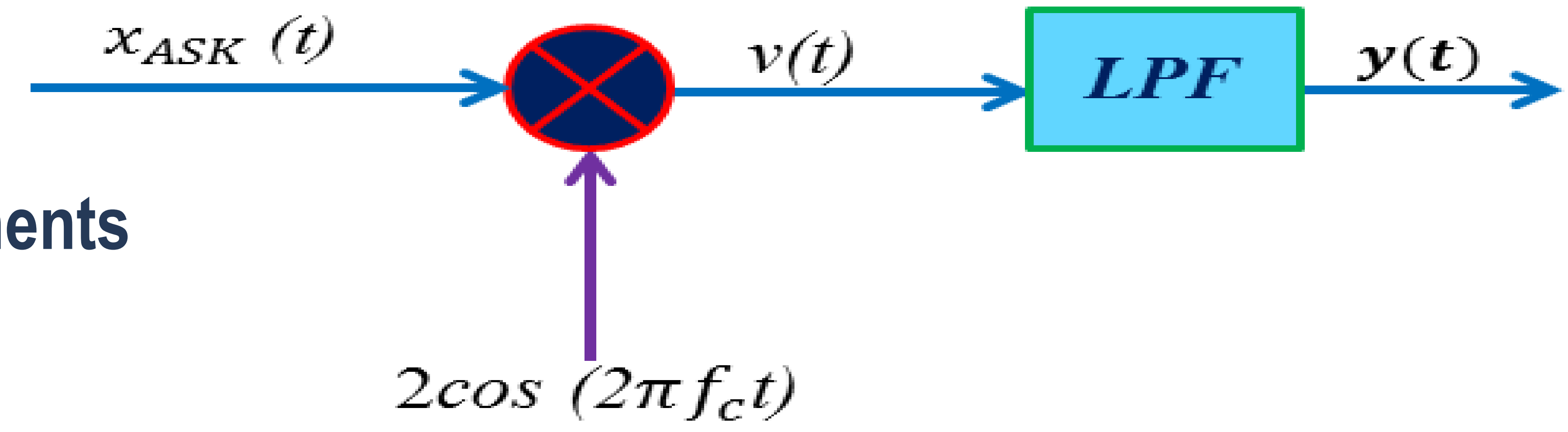
If d= 0, it is considered an ideal case. If d= 1, it is considered the worst case.

ASK Applications:

1- Signals broadcasting

2- Optical fiber communication for laser intensity modulation

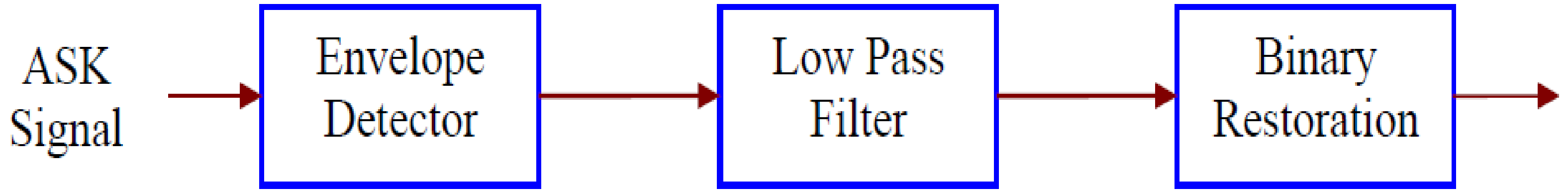
Coherent (Synchronous) ASK Receiver



- LPF to remove the high frequency components
- Efficient
- costly

Noncoherent ASK Receiver

- Square-law detection is used
- LPF to remove the high frequency components
- Low cost
- Poor performance when the received signal has low SNR.



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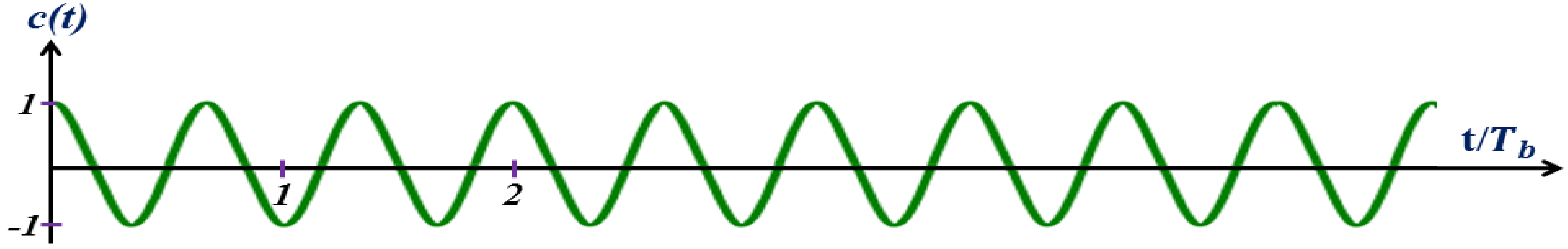
Example: The sequence b(k) = {1, 1, 0, 1, 0, 1} is transmitted at bit rate of R_b =200 b/s using an ASK digital system. The carrier signal c(t) is given as shown below.



a- Calculate the carrier frequency f_c.

b- Calculate the bit duration T_b and the required bandwidth B_{ASK}.

c- Sketch the baseband signal m(t).

d- Sketch the transmitted passband signal x_{ASK} (t).

Solution:-

a-

$$f_c = \frac{t}{T_b} \longrightarrow f_c = 1.5\frac{cycle}{T_b}$$

$$T_b = \frac{1}{R_b} = \frac{1}{200} = 5*10^{-3} = 5 \text{ msec}$$

$$f_c = 1.5\frac{cycle}{5\,msec} = 0.3\ KHz = 300 \text{ Hz}$$

$$\boldsymbol{R_b = \frac{1}{T_b}}$$

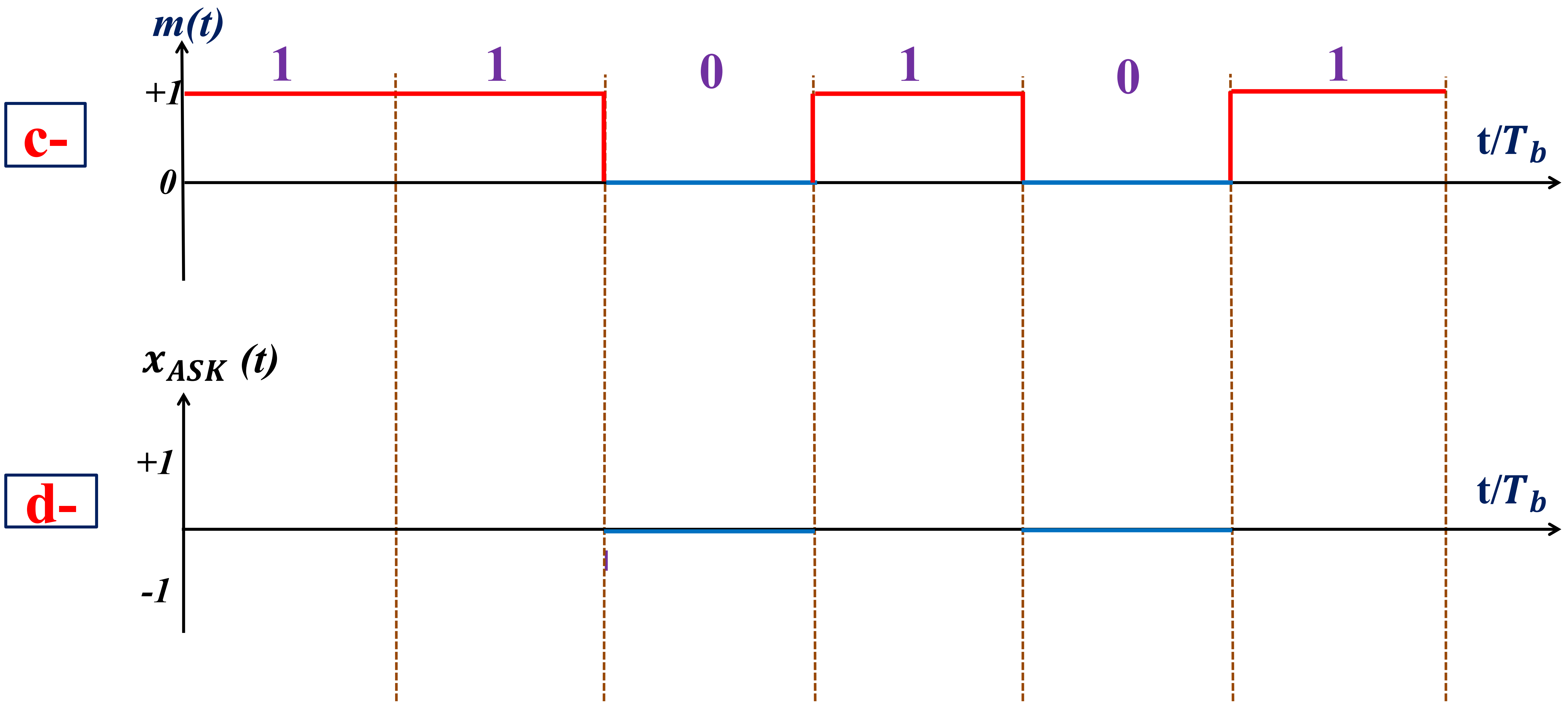
b-

$$T_b = \frac{1}{R_b} = 5 \text{ msec}$$

$$B_{ASK} = 2f_b$$

$$f_b = \frac{1}{T_b} = \frac{1}{5 \text{ msec}} = 200Hz$$

$$B_{ASK} = 2 * 200 = 400\ Hz$$



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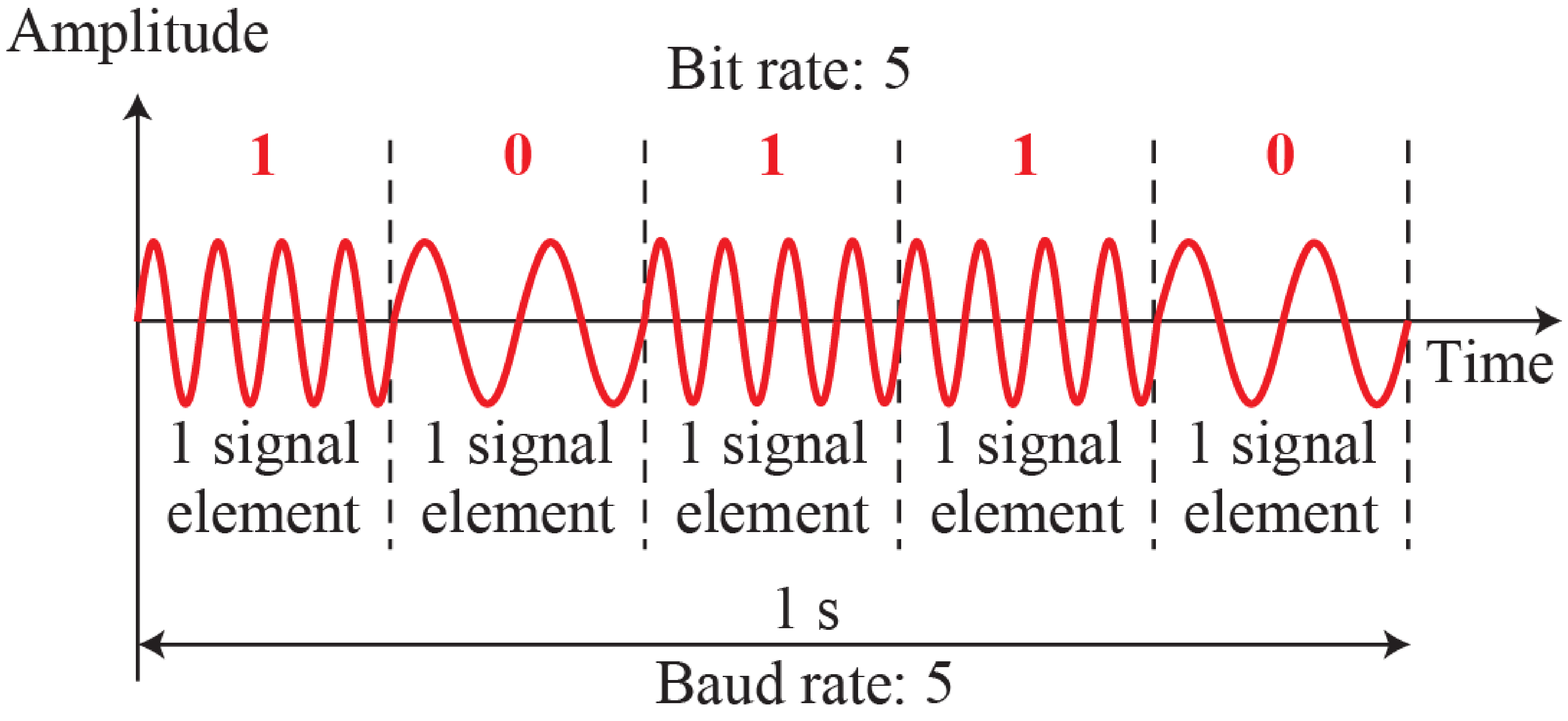
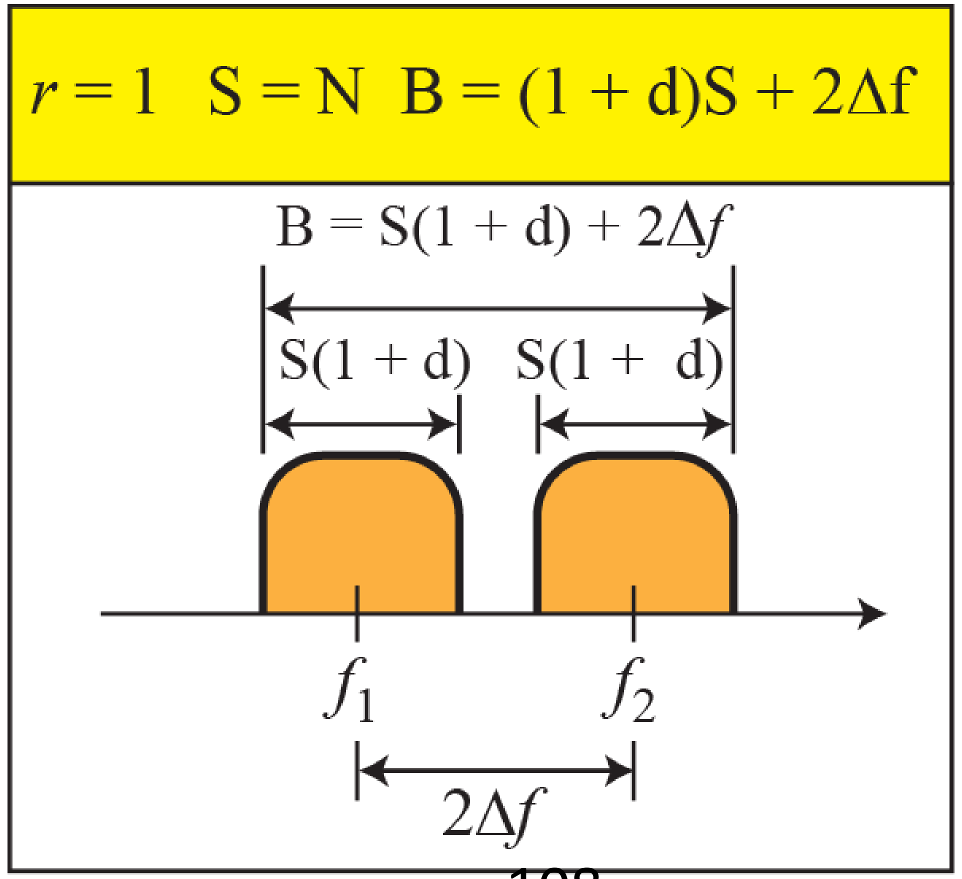
FREQUENCY SHIFT KEYING

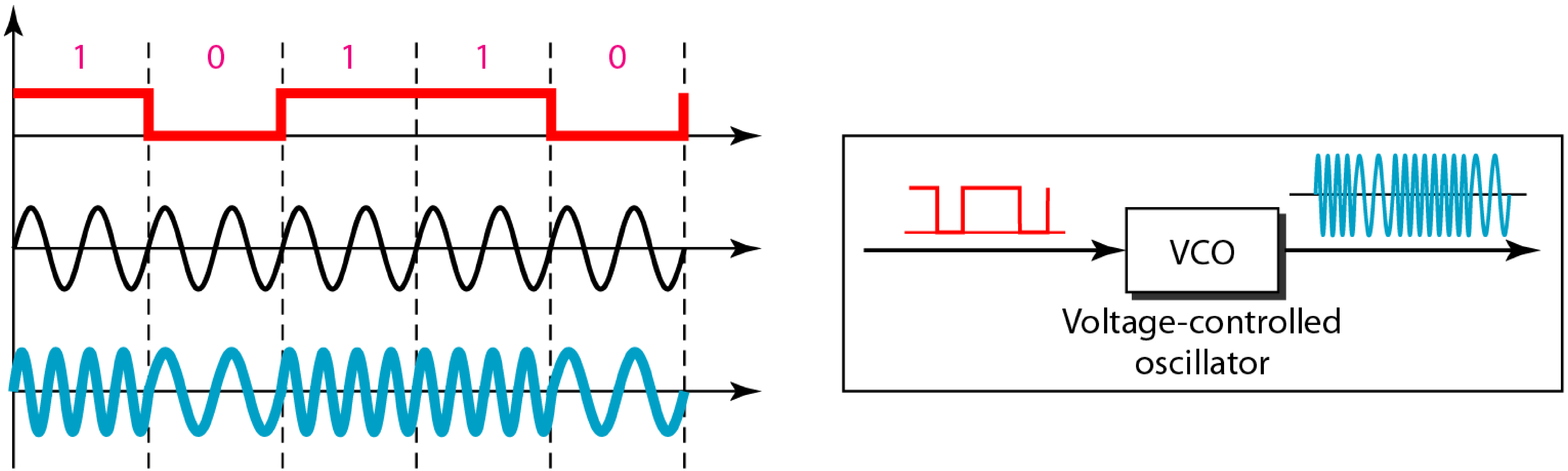
In frequency shift keying, the frequency of the carrier signal is varied to represent data. The frequency of the modulated signal is constant for the duration of one signal element, but changes for the next signal element if the data element changes. Both peak amplitude and phase remain constant for all signal elements. It is used to convert digital data to analog data.



Binary frequency shift keying



FREQUENCY SHIFT KEYING



Implementation of binary FSK

FREQUENCY SHIFT KEYING

Example: We have an available bandwidth of 100 kHz which spans from 200 to 300 kHz. What should be the carrier frequency and the bit rate if we modulated our data by using FSK with d = 1?

Solution

This problem is similar to the previous example, but we are modulating by using FSK. The midpoint of the band is at 250 kHz. We choose $2\Delta f$ to be 50 kHz; this means:

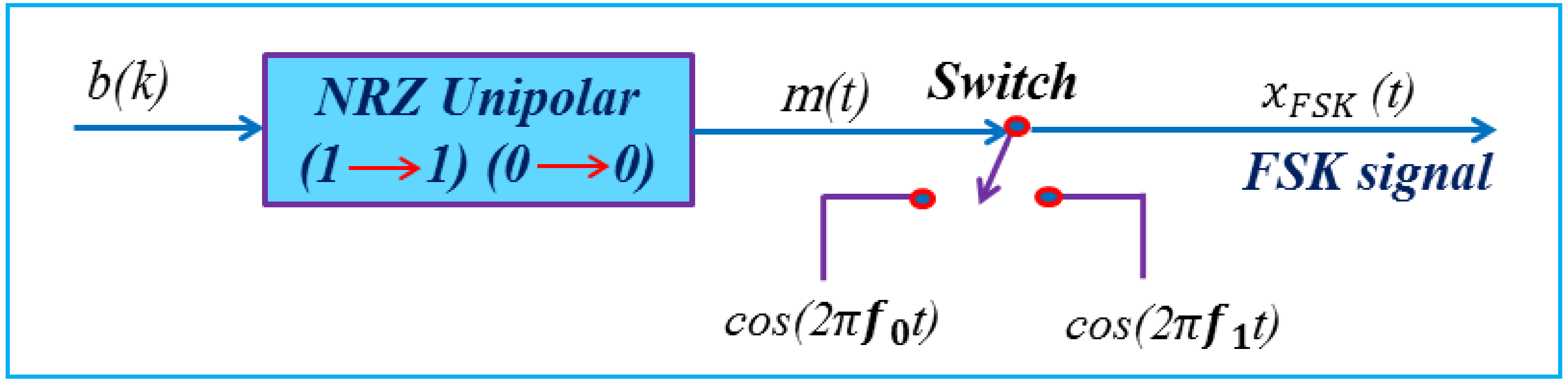
$$B = (1 + d) \times S + 2\Delta_f = 100 \longrightarrow 2S = 50 \text{ kHz} \longrightarrow S = 25 \text{ kbaud} \longrightarrow N = 25 \text{ kbps}$$

FREQUENCY SHIFT KEYING

FSK Transmitter

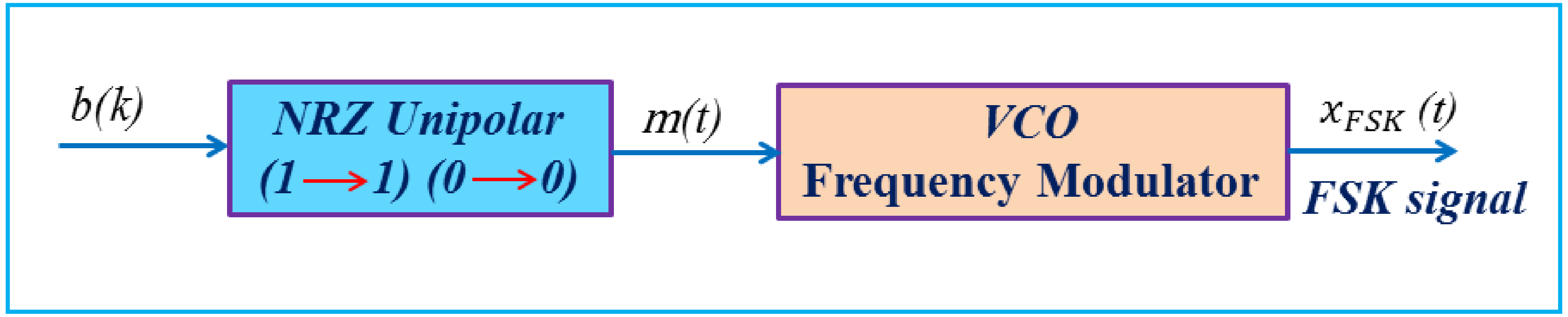
Discontinuous-Phase (DPFSK)

- m(t) controls a switch that selects the frequency f_0 or f_1.



Continuous-Phase (CPFSK)

- Voltage-Controlled Oscillator (VCO) as frequency modulator.
  - Avoid discontinuities in x_{FSK} (t).



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1

- **Frequency (f_0, f_1 and f_c):**

$$\boxed{f_1 = f_c + \Delta f}$$
$$\boxed{f_0 = f_c - \Delta f}$$
$$\boxed{f_c = \frac{f_1 + f_0}{2}}$$

f_0: Low Frequency. (Hz)

f_1: High Frequency. (Hz)

f_c: Carrier Frequency. (Hz)

2

- **Frequency Deviation (Δf) : (assuming $f_1 > f_0$)**

$$\boxed{\Delta f = \frac{f_1 - f_0}{2}}$$

Δf: Frequency Deviation. (Hz)

3

- **Modulation index (h) :**

$$\boxed{h = 2\Delta f * T_b}$$

h: Modulation index.

T_b: Bit duration. $(sec.)$

f_b: Bit frequency. (Hz)

R_b: Bit rate. (b/s)

$$\boxed{R_b = \frac{1}{T_b}} \qquad \boxed{f_b = \frac{1}{T_b}}$$

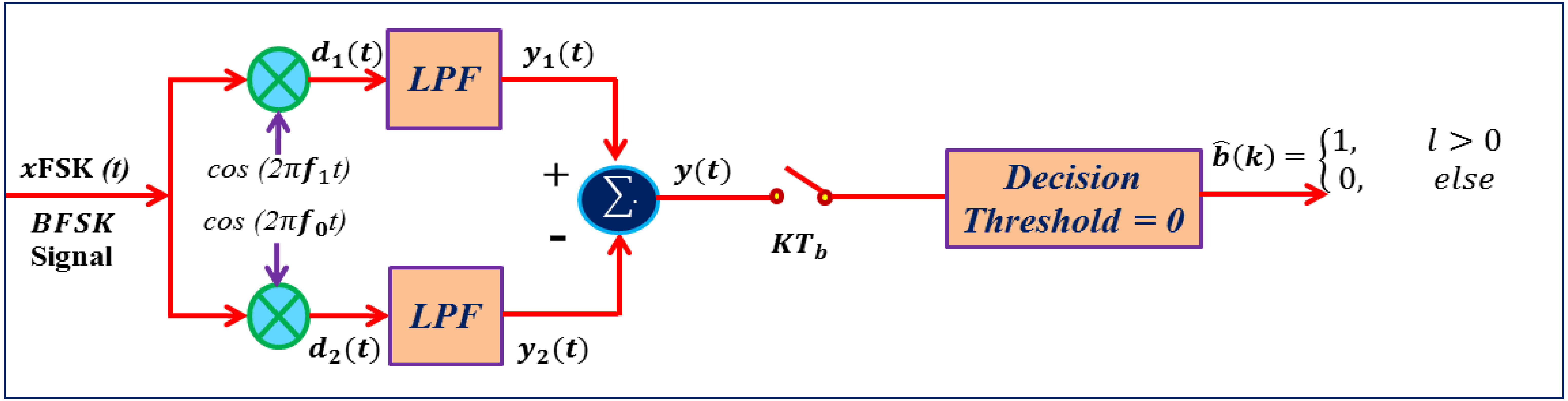
4

- **Bandwidth (B_{FSK}) :**

$$\boxed{B_{FSK} = 2\Delta f + 2f_b}$$

B_{FSK}: Bandwidth

Coherent (Synchronous) BFSK Receiver

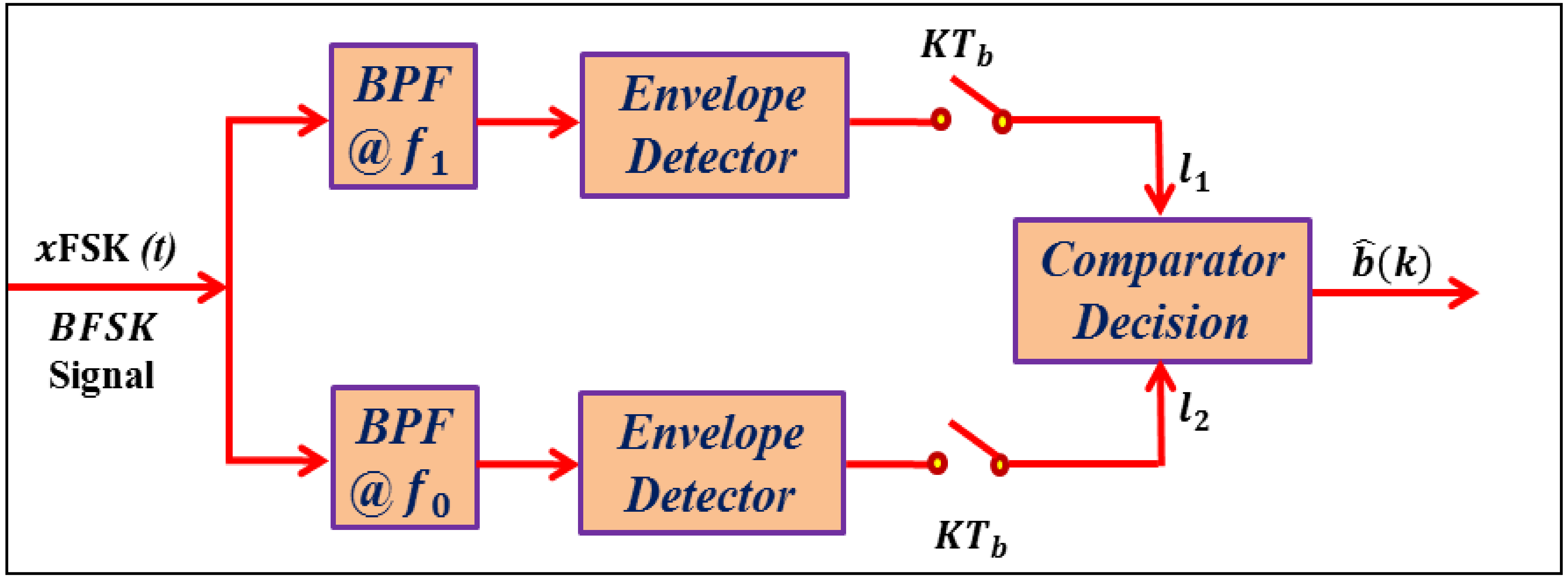


Noncoherent BFSK Receiver

Band-Pass Filter (BPF):- is a device that passes frequencies within a certain range and rejects (attenuates) frequencies outside that range.

Low-Pass Filter (LPF):- is a filter that passes signals with a frequency lower than a selected cutoff frequency and attenuates signals with frequencies higher than the cutoff frequency.

Coherent receiver is costly as compared to noncoherent.



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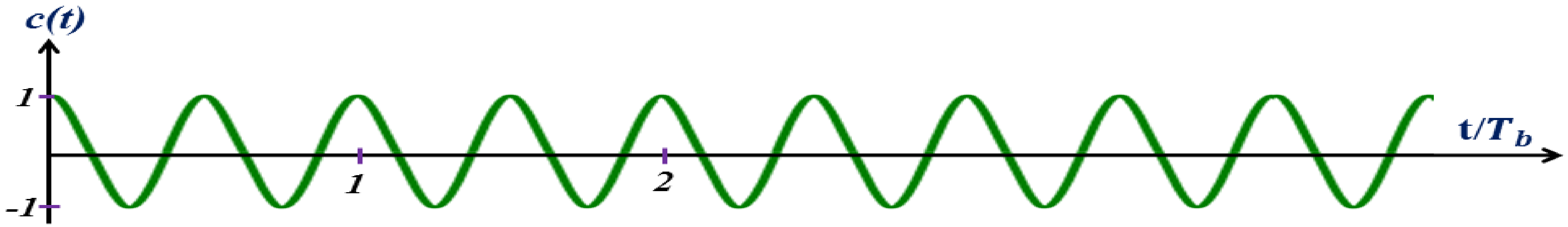
Example: The sequence b(k) = {1, 0, 0, 1, 1, 0} is transmitted at bit rate of R_b = 1 Kb/s using an FSK digital system, the frequency f_1 = 3 KHz and f_0 = 1 KHz. The carrier signal c(t) is given as shown below.



a- Calculate the carrier frequency f_c.

b- Calculate the frequency deviation Δf.

c- Calculate the modulation index h.

d- Sketch the baseband signal m(t).

e- Sketch the transmitted passband signal x_{FSK} (t).

Solution:-

a-

$$f_c = \frac{f_1 + f_0}{2}$$

$$f_c = \frac{3\,K + 1\,K}{2} = 2\ \text{KHz}$$

b-

$$\Delta f = \frac{f_1 - f_0}{2}$$

$$\Delta f = \frac{3\,K - 1\,K}{2} = 1\,KHz$$

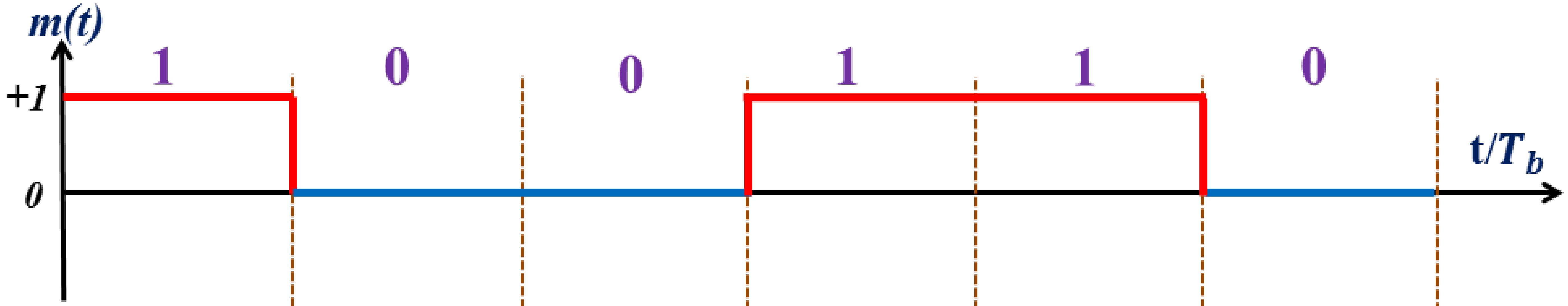
c-

$$h = 2\Delta f \, * T_b$$

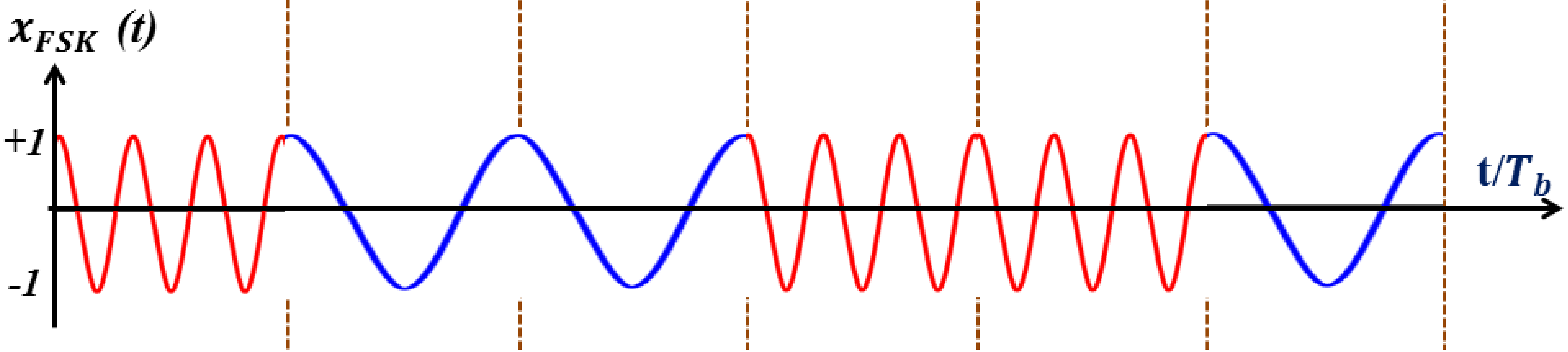
$$T_b = \frac{1}{R_b} = \frac{1}{1\,K} = 1\ \text{msec}$$

$$h = 2 * 1\,K * 1\,m = 2$$

d-



e-



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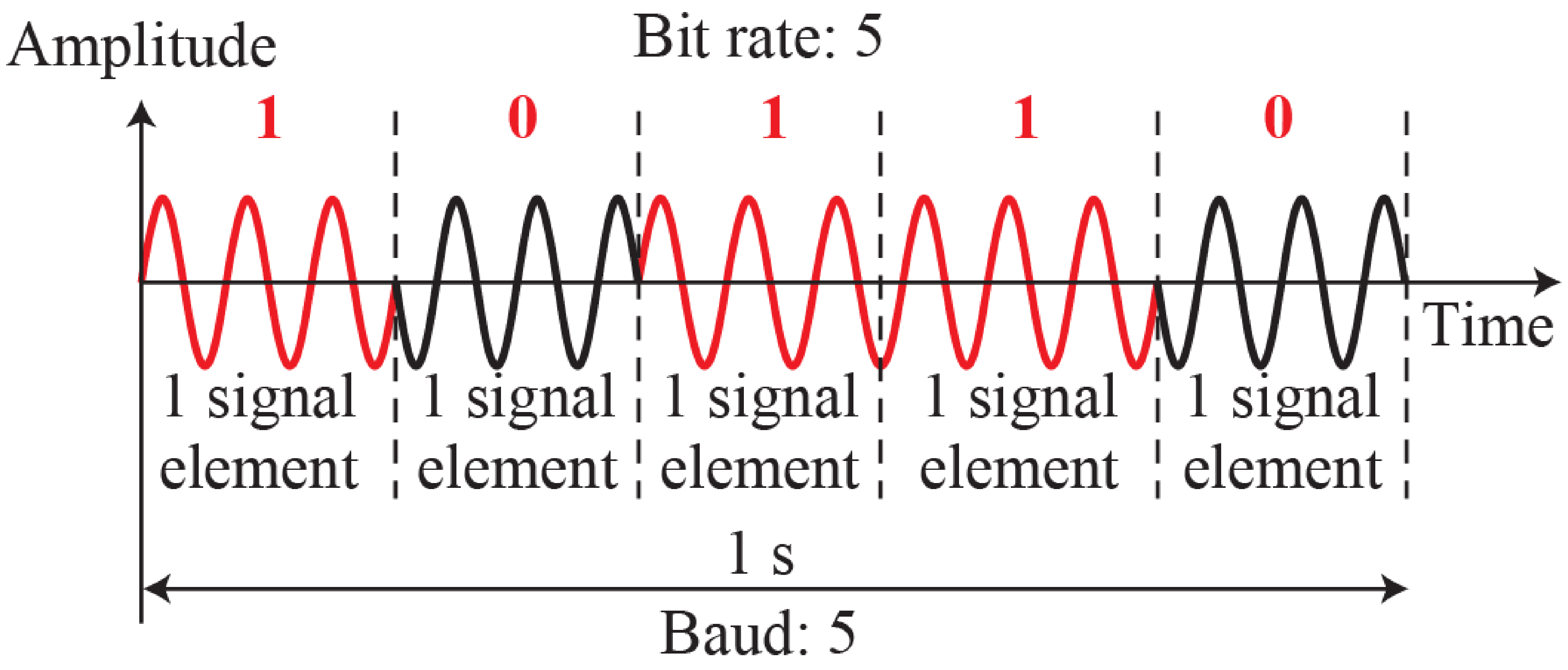
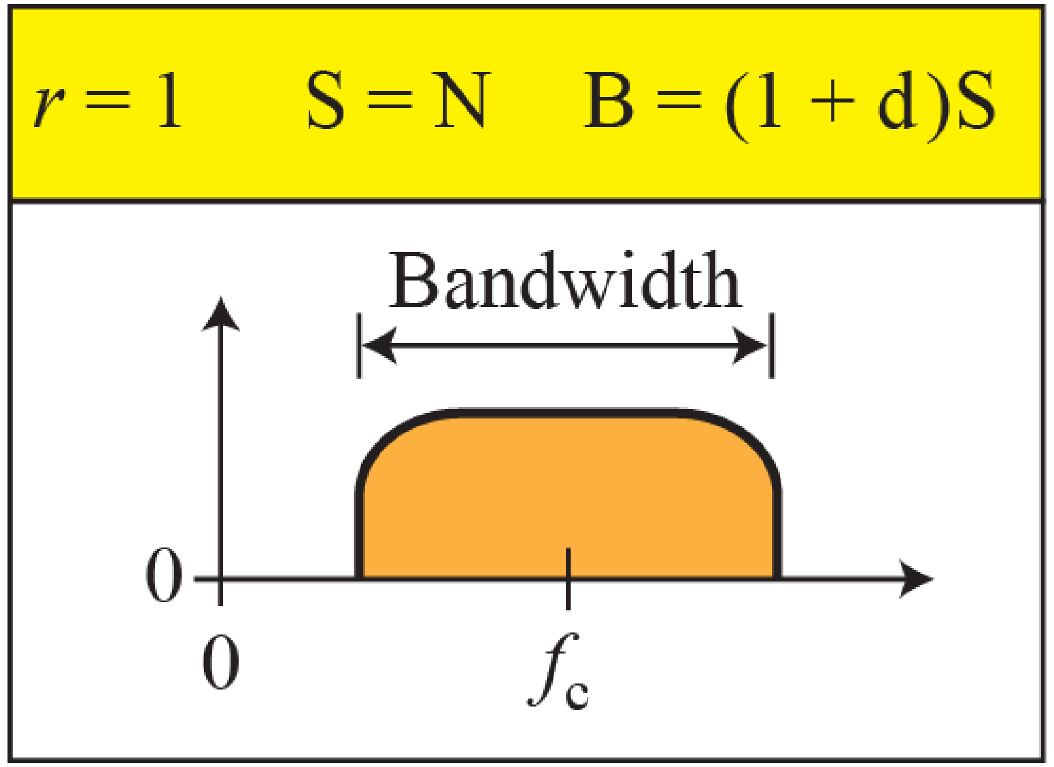
PHASE SHIFT KEYING

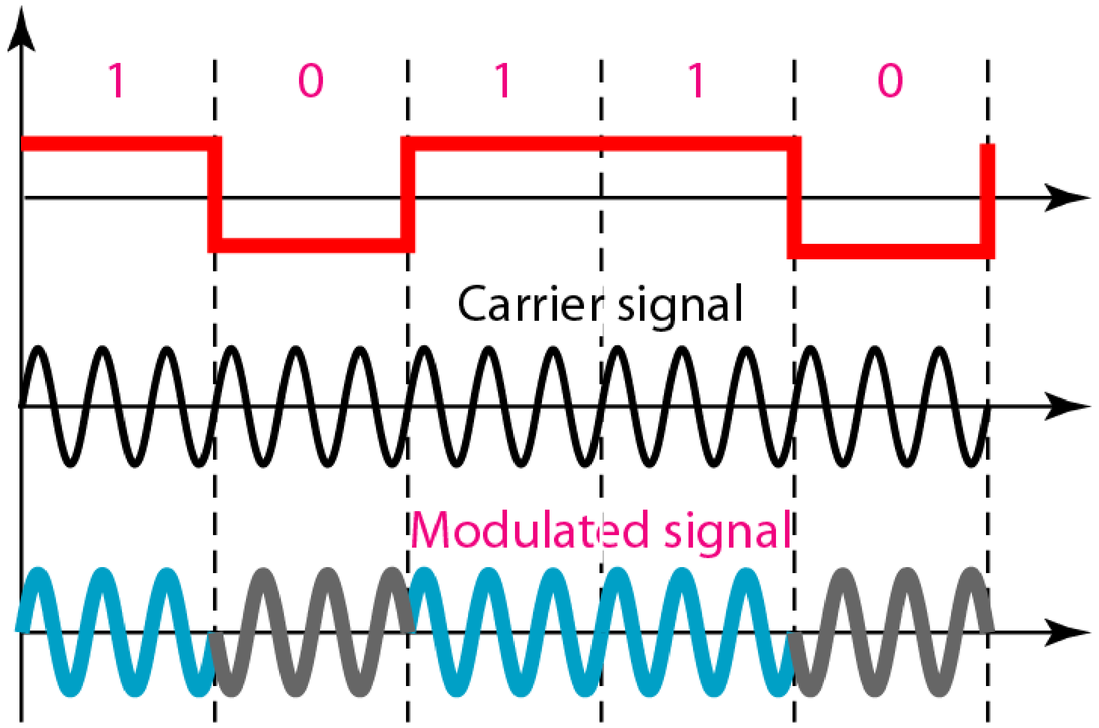
In phase shift keying, the phase of the carrier is varied to represent two or more different signal elements. Both peak amplitude and frequency remain constant as the phase changes. PSK is more common than ASK or FSK.

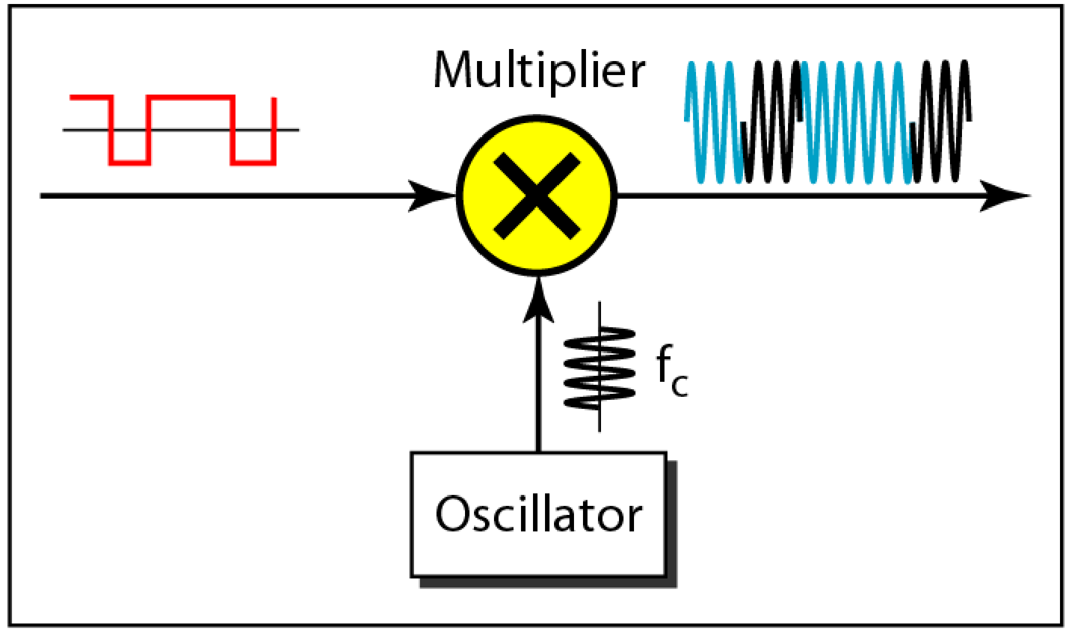




Binary Phase shift keying

PHASE SHIFT KEYING





Implementation of binary PSK

PHASE SHIFT KEYING

Example: Find the bandwidth for a signal transmitting at 12 Mbps for QPSK. The value of d = 0.

Solution

For QPSK, 2 bits are carried by one signal element. This means that $r = 2$. So the signal rate (baud rate) is $S = N \times (1/r) = 6$ Mbaud. With a value of $d = 0$, we have $B = S = 6$ MHz.

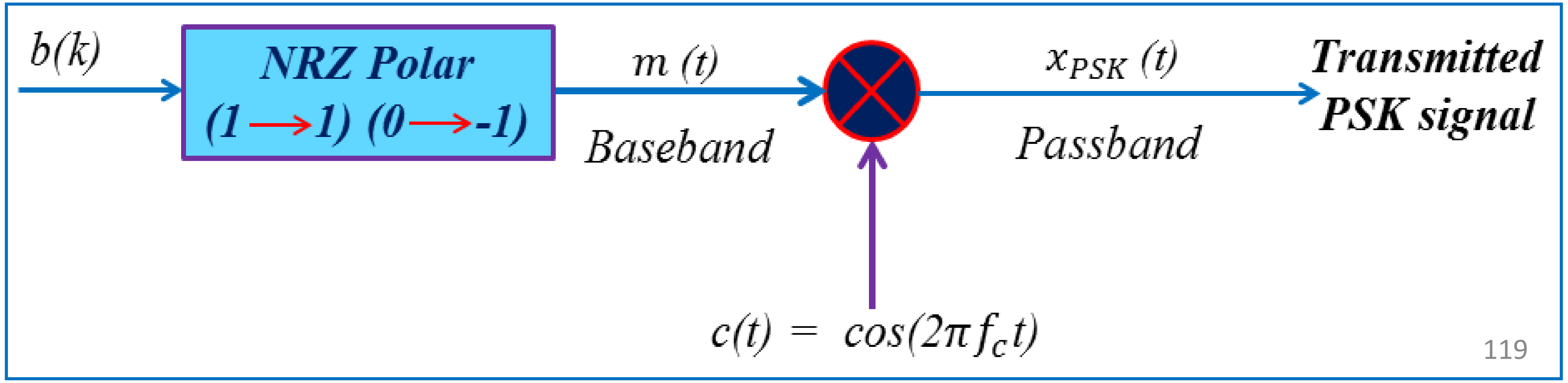
PHASE SHIFT KEYING :

- Binary Phase Shift Keying (BPSK)
- Differential Phase Shift Keying (DPSK)

Binary Phase Shift Keying (BPSK)

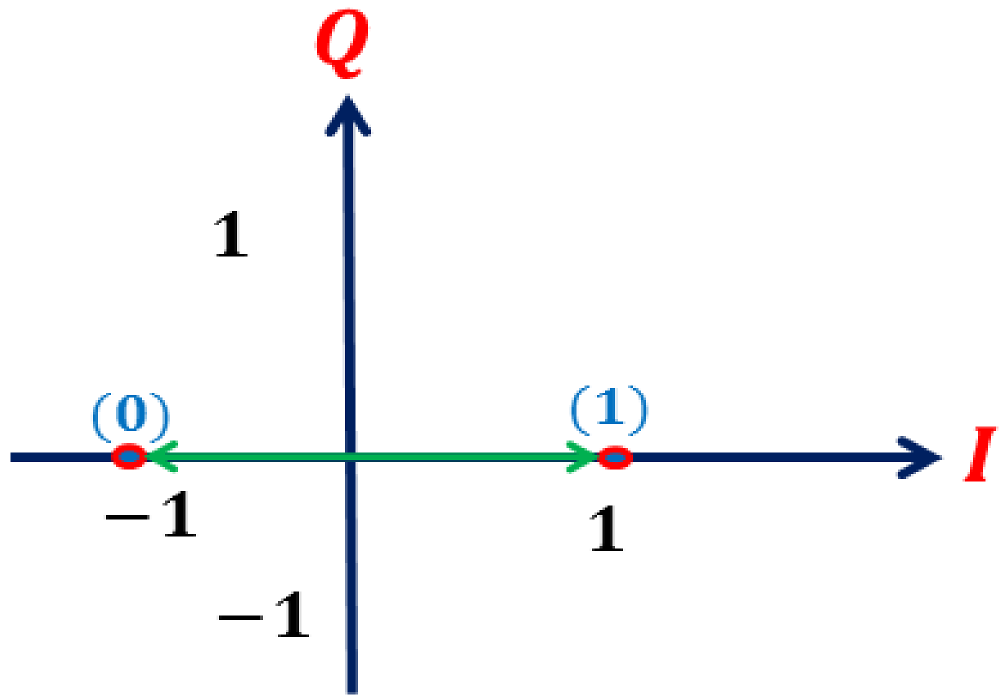
BPSK Transmitter

- In binary-PSK, the phase of the carrier is switched between two values: (0 for binary 1) and (π for binary 0).



Constellation diagram of BPSK signal

- Bit combinations are assigned in red.
- One bit per symbol is transmitted



$$\boxed{T_s = T_b}$$

$$\boxed{R_s = R_b}$$

R_s: Symbol rate (Baud rate) (b/s).
T_s: Symbol duration (sec.).
R_b: Bit rate (b/s).
T_b: Bit duration (sec.).

$$\boxed{R_s = \frac{1}{T_s}}$$

$$\boxed{R_b = \frac{1}{T_b}}$$

$$\boxed{B_{PSK} = 2f_b}$$

B_{PSK}: Bandwidth of PSK

$$\boxed{\eta_{PSK} = \frac{R_b}{B_{PSK}} = \frac{1/T_b}{2f_b} = \frac{f_b}{2f_b} = \frac{1}{2}\ bit/s/Hz}$$

η_{ASK}: Spectral efficiency

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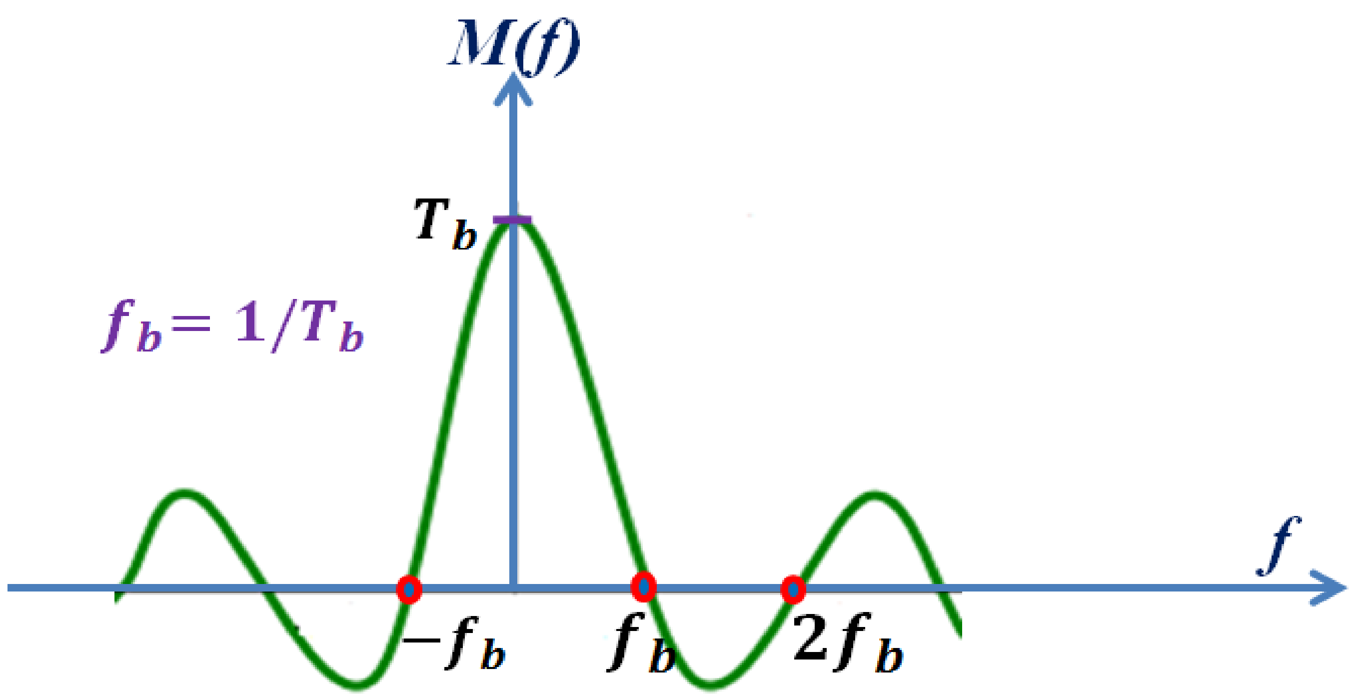
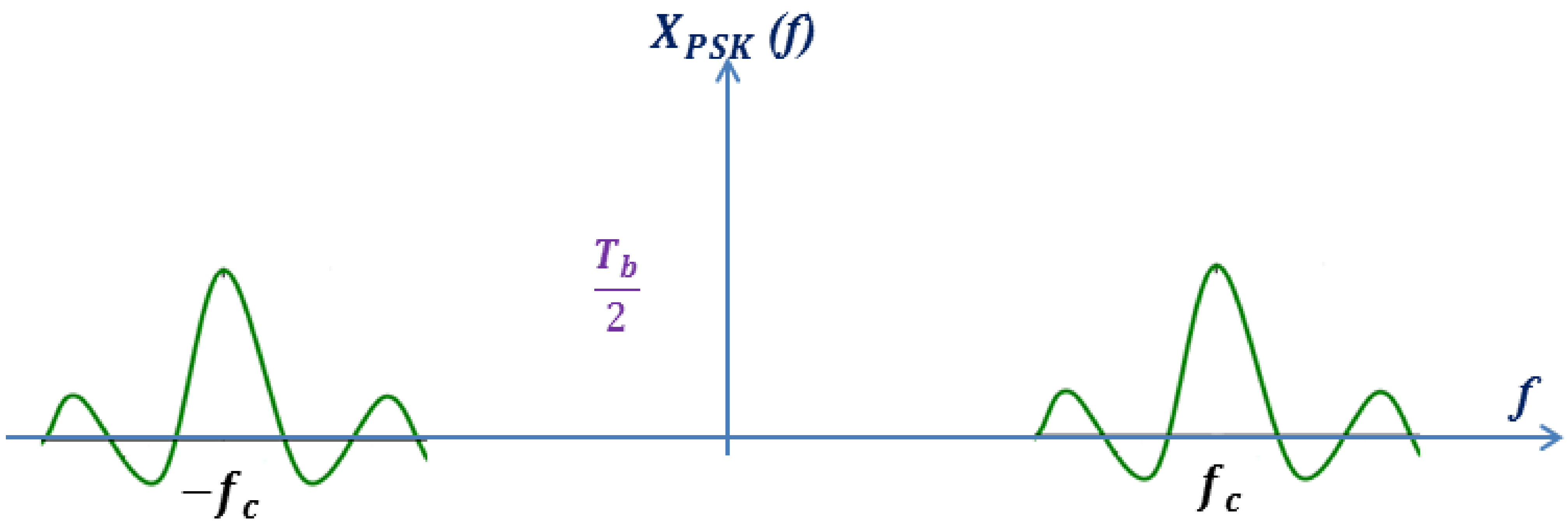
Spectrum of Binary PSK signal

Baseband	Passband
$m(t) = rect\left(\frac{t}{T_b}\right)$ $M(f) = T_b\, sinc(\pi f T_b)$	$x_{PSK}(t) = m(t)\cos(2\pi f_c t)$ $X_{PSK}(f) = \frac{1}{2}M(f - f_c) + \frac{1}{2}M(f + f_c)$





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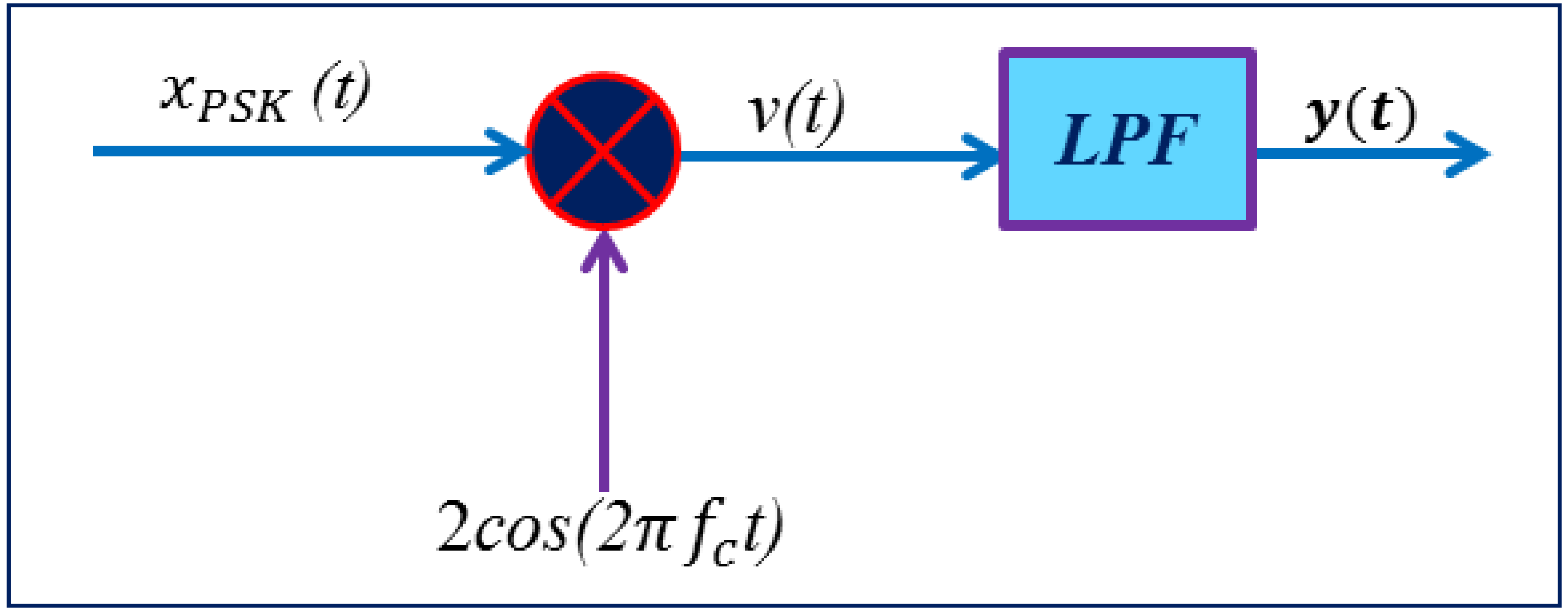
Coherent Binary PSK Receiver

- Noncoherent (envelope) detection cannot be used to detect BPSK signal: because the message information resides in the phase (constant envelope and frequency).
- For example, by squaring the received signal, it is difficult to determine whether the received bit is (+1) or (-1).
- Coherent (synchronous) detection must be used



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Differential Phase Shift Keying (DPSK)

PSK signals may be demodulated noncoherently by means of an ingenious method known as differential PSK

Advantages and disadvantages of DPSK

Advantages of DPSK:

1- DPSK does not need coherent receiver.
2- DPSK bandwidth is smaller than BPSK bandwidth.
3- It reduces cost.

Disadvantages of DPSK:

1- Bit error rate (BER) of DPSK is higher than that of BPSK.
2- Error propagation in DPSK is more than that in BPSK.

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Example: The bit sequence b(k) = {1, 0, 1, 1, 0, 0, 1, 0} has been transmitted using a PSK digital system with bit rate of R_b = 1 Kb/s. The carrier frequency is f_c = 1 KHz.

a- Determine the bit duration T_b and the symbol duration T_s.

b- Sketch the baseband signal m(t).

c- Sketch the passband transmitted PSK-signal x_{PSK} (t).

Solution:-

a-

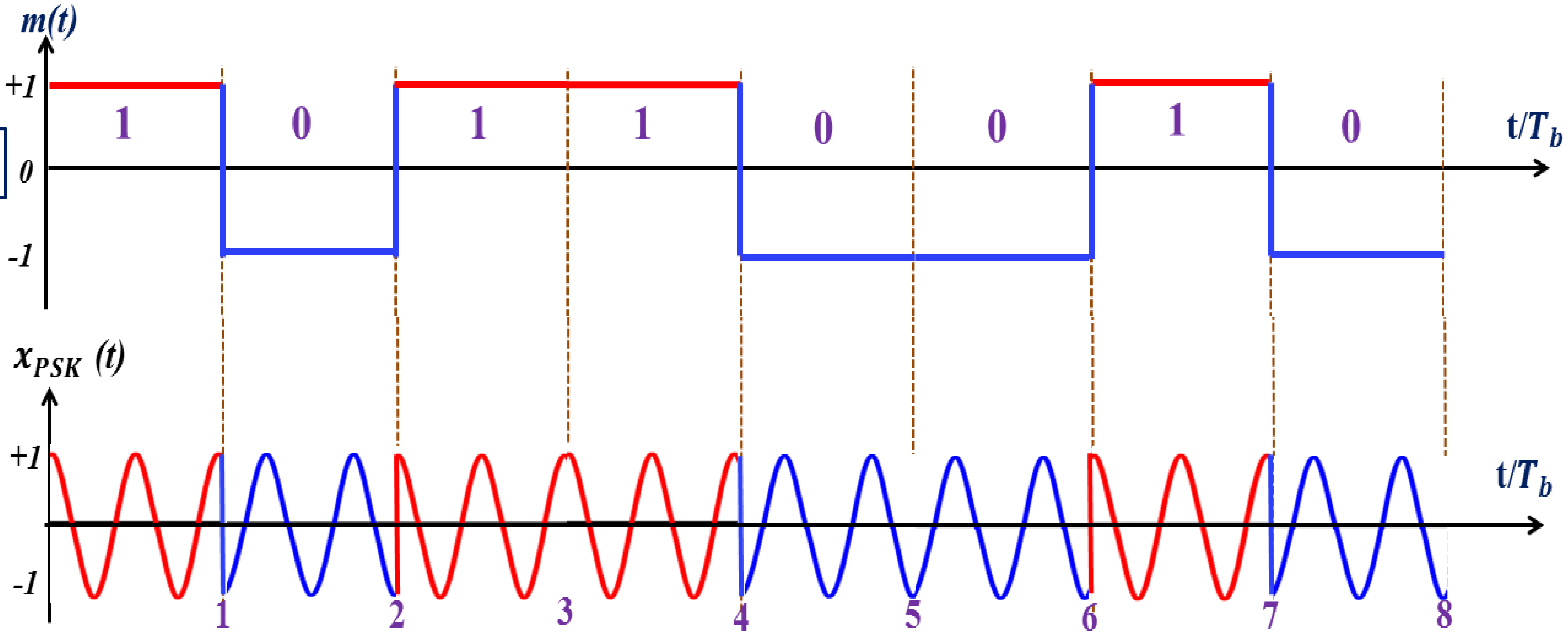
$$\boxed{R_b = \frac{1}{T_b}}$$

$$T_b = \frac{1}{R_b} = \frac{1}{1\ K} = 1 \text{ msec}$$

$$\boxed{T_b = T_s = 1 \text{ msec}}$$

b-

c-



ASK, FSK AND PSK COMPARISON

Parameters	ASK	FSK	PSK
Variable Characteristics	Amplitude	Frequency	Phase
Bandwidth	B=(1+d)S	B=(1+d)S+2 Δf	B=(1+d)S
Noise Immunity	As noise is very sensitive to amplitude, here noise immunity is low	Here noise immunity is high.	Here also noise immunity is high.

ASK, FSK AND PSK COMPARISON

Complexity	Simple	Moderately complex	Very complex
Error probability	High	low	Low
Performance in presence of noise	Poor	Better than ask	Better than fsk
Bit rate	Suitable up to 100 bits/sec	Suitable up to 1200 bits/sec	Suitable for higher bit rate

College of Electronics Engineering

Systems & Control Engineering Department

Digital Communications (SCE3316)

Lecture 7
(Probability of Error for ASK, FSK, PSK and DPSK)

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INTRODUCTION

- In digital transmission, the number of bit errors is the number of received bits of a data stream over a communication channel that have been altered due to noise, interference, distortion, or bit synchronization errors.
- The bit error rate (BER) is the number of bit errors per unit time. The bit error ratio (also BER) is the number of bit errors divided by the total number of transferred bits during a studied time interval. Bit error ratio is a unitless performance measure, often expressed as a percentage.
- The bit error probability (pe) is the expected value of the bit error ratio. The bit error ratio can be considered as an approximate estimate of the bit error probability. This estimate is accurate for a long-time interval and a high number of bit errors.

PROBABILITY OF ERROR OF ASK

Probability of error $P_e = \frac{1}{2} erfc\left[\frac{1}{8} r_{max}{}^2\right]^{\frac{1}{2}}$

$$r_{max}{}^2 = \frac{2}{\eta} \int_0^T [S_1(t) - S_2(t)]^2 dt$$

In ASK system,

$$S_1(t) = Acos\omega_0 t$$

$$\& \; S_2(t) = 0$$

$$\therefore \; P(t) = S_1(t) - S_2(t) = Acos\omega_0 t$$

$$r_{max}{}^2 = \frac{2}{\eta} \int_0^T A^2 \, cos^2\omega_0 t \; dt$$

$$= \frac{2}{\eta} \int_0^T \frac{A^2}{2} [1 + cos2\omega_0 t] dt$$

$$= \frac{2}{\eta} \int_0^T \frac{A^2}{2} dt + \int_0^T \frac{A^2}{2} cos2\omega_0 t dt$$

$$= \frac{2}{\eta} \cdot \frac{A^2}{2} [t]_0^T + \frac{A^2}{\eta} \frac{sin\omega_0 t}{2\omega_0}\Big]_0^T$$

$$r_{max}{}^2 = \frac{A^2 T}{\eta}$$

η = N : NOISE POWER SPECTRAL DENSITY

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PROBABILITY OF ERROR OF ASK

Probability of error $P_e = \frac{1}{2} erfc\left[\frac{1}{8} r_{max}{}^{2}\right]^{\frac{1}{2}}$

$$= \frac{1}{2} erfc\left[\frac{1}{8} . \frac{A^2 T}{\eta}\right]^{\frac{1}{2}}$$

$$= \frac{1}{2} erfc\left[\frac{E_s}{4\eta}\right]^{\frac{1}{2}} \quad [\because E_s = \frac{A^2 T}{2}]$$

$$\therefore Probability\ of\ error\ for\ ASK = \frac{1}{2} erfc\left[\frac{E_s}{4\eta}\right]^{\frac{1}{2}}$$

$$1/2\ erfc\ [\ y/\sqrt{2}\ ] = Q(y)$$

$$\therefore Probability\ of\ error\ for\ ASK = Q\left[\frac{E_s}{2\eta}\right]^{\frac{1}{2}}$$

PROBABILITY OF ERROR OF FSK

As the Probability of error $P_e = \frac{1}{2} erfc\left[\frac{1}{8} r_{max}{}^2\right]^{\frac{1}{2}}$

$$r_{max}{}^2 = \frac{2}{\eta}\int_0^T [S_1(t) - S_2(t)]^2 dt$$

In FSK system,

$$S_1(t) = Acos(\omega_0+\Omega)t$$

$$\& \; S_2(t) = -Acos(\omega_0-\Omega)t$$

$$r_{max}{}^2 = \frac{2}{\eta}\int_0^T A^2[cos(\omega_0+\Omega)t \; - cos(\omega_0-\Omega)t]^2 dt$$

$$= \frac{2A^2}{\eta}\int_0^T [cos^2(\omega_0+\Omega)t + cos^2(\omega_0-\Omega)t - 2cos(\omega_0+\Omega)tcos(\omega_0-\Omega)t]dt$$

$$= \frac{2A^2}{\eta}\int_0^T \frac{1+cos2(\omega_0+\Omega)t}{2}dt + \int_0^T \frac{1+cos2(\omega_0-\Omega)t}{2}dt - \int_0^T cos2(\omega_0)tdt - \int_0^T cos2(\Omega)tdt$$

$$= \frac{2A^2}{\eta}\left\{T + \frac{1}{2}\frac{\sin 2(\omega_0+\Omega)}{2(\omega_0+\Omega)}\mathrm{t} + \frac{1}{2}\frac{\sin 2(\omega_0-\Omega)}{2(\omega_0-\Omega)}\mathrm{t} - \frac{sin\omega_0 t}{2\omega_0} - \frac{sin\Omega t}{2\,\Omega}\right\}$$

$$= \frac{2A^2}{\eta}\left\{T - \frac{\sin 2\Omega T}{2\Omega} - \frac{\sin 2\omega_0\mathrm{T}}{2\omega_0} + \frac{1}{2}\frac{\sin 2(\omega_0+\Omega)T}{2(\omega_0+\Omega)} + \frac{1}{2}\frac{\sin 2(\omega_0-\Omega)T}{2(\omega_0-\Omega)}\right\}$$

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PROBABILITY OF ERROR OF FSK

Let $\omega_0 T \gg 1$ & $\omega_0 \gg \Omega$
Which are usually encountered in practical systems, the last '3' terms can be neglected.

Hence $$r_{max}{}^2 = \frac{2A^2T}{\eta}\left[1 - \frac{\sin 2\Omega T}{2\Omega}\right]$$

The quantity $r_{max}{}^2$ attains its maximum value when 'Ω' is so selected that

$$2\Omega T = \frac{3\pi}{2}$$

For this value of Ω , we have $r_{max}{}^2 = 2.42\frac{A^2T}{\eta}$

$\therefore$ The probability of error $P_e = \frac{1}{2}\text{erfc}\left[\frac{1}{8}r_{max}^2\right]^{1/2}$

$$= \frac{1}{2}\text{erfc}\left[\frac{1}{8}2.42\frac{A^2T}{\eta}\right]^{1/2}$$

$$= \frac{1}{2}\text{erfc}\left[0.3\frac{A^2T}{\eta}\right]^{1/2}$$

$\therefore$ The probability of error $P_e = \frac{1}{2}\text{erfc}\left[0.6\frac{E_S}{\eta}\right]^{1/2}$

When one of two *orthogonal* frequencies are transmitted,

$2\Omega T = m\pi$ (m an integer) and

probability of error for FSK $P_e = \frac{1}{2}\text{erfc}\left(\frac{E_s}{2\eta}\right)^{1/2}$

PROBABILITY OF ERROR OF PSK

As the Probability of error $P_e = \frac{1}{2} erfc \left[\frac{1}{8} {r_{max}}^2\right]^{\frac{1}{2}}$

$${r_{max}}^2 = \frac{2}{\eta} \int_0^T [S_1(t) - S_2(t)]^2 dt$$

In PSK system,

$$S_1(t) = A cos\omega_0 t$$

$$\& \ S_2(t) = -A cos\omega_0 t$$

$$\therefore \ P(t) = S_1(t) - S_2(t) = 2A cos\omega_0 t$$

$${r_{max}}^2 = \frac{2}{\eta} \int_0^T (2A \ cos\omega_0 t)^2 \ dt$$

$$= \frac{2}{\eta} . 4A^2 \int_0^T [cos^2 \omega_0 t] dt$$

$$= \frac{8A^2}{\eta} . \int_0^T \frac{1 + cos2\omega_0 t}{2} dt$$

$$= \frac{4A^2}{\eta} . \int_0^T dt + \int_0^T cos2\omega_0 t dt$$

$$= \frac{4A^2}{\eta} [t]_0^T + \left[\frac{sin\omega_0 t}{2\omega_0}\right]_0^T$$

$${r_{max}}^2 = \frac{4A^2 T}{\eta}$$

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PROBABILITY OF ERROR OF PSK

Probability of error $P_e = \frac{1}{2} erfc\left[\frac{1}{8} {r_{max}}^2\right]^{\frac{1}{2}}$

$$= \frac{1}{2} erfc\left[\frac{1}{8} . \frac{4A^2T}{\eta}\right]^{\frac{1}{2}}$$

$$= \frac{1}{2} erfc\left[\frac{A^2T}{2\eta}\right]^{\frac{1}{2}}$$

$$= \frac{1}{2} erfc\left[\frac{E_s}{\eta}\right]^{\frac{1}{2}} \qquad [\because E_s = \frac{A^2T}{2}]$$

$$\therefore probability\ error\ for\ PSK\ = \frac{1}{2} erfc\left[\frac{E_s}{\eta}\right]^{\frac{1}{2}}$$

$1/2\, erfc\,[\, y/\sqrt{2}\,] = Q(y)$

$$\therefore probability\ error\ for\ PSK\ = \quad Q\left[\frac{2E_s}{\eta}\right]^{\frac{1}{2}}$$

Probability of Error of DPSK

The Pe in the case of DPSK is

$$Pe = \frac{1}{2} e^{-Es/\eta}$$

Example 1:

A binary band pass system transmits binary data at the rate of $2.5 * 10^6$ $bits/second$. During the course of transmission , zero mean AWGN of 2 sided PSD Signal to 10^{-14} w/Hz is added to the signal. In the absence of noise, the amplitude of the received sinusoidal wave for digit '1' or '0' is 1mV. Find the average probability of symbol error of BPSK.

$$P_e = \frac{1}{2} erfc \sqrt{\frac{E_b}{\eta}}$$

$$E_b = \frac{A^2 T}{2}$$

$$T = \frac{1}{R_b} = \frac{1}{2.5 * 10^6} = 0.4 * 10^{-6} \; sec$$

A= 1mV

$$\therefore E_b = \frac{A^2 T}{2} = \frac{1 * 10^{-6} * 0.4 * 10^{-6}}{2} = 0.2 * 10^{-12}$$

$$P_e = \frac{1}{2} erfc \sqrt{\frac{E_b}{\eta}}$$

$$= \frac{1}{2} erfc \sqrt{\frac{0.2 * 10^{-12}}{2 * 10^{-14}}} = \frac{1}{2} erfc \sqrt{10}$$

Pe= ??

Example 2:

probability of error of ASK, PSK, FSK and DPSK

- ASK $\rightarrow P_e = \frac{1}{2} \cdot Q\left(\sqrt{\frac{E_s}{4\eta}}\right)$

- FSK $\rightarrow P_e = \frac{1}{2} Q\left(\sqrt{\frac{0.6 E_s}{\eta}}\right)$

- PSK $\rightarrow P_e = \frac{1}{2} Q\left(\sqrt{\frac{E_s}{\eta}}\right)$

- DPSK $\rightarrow P_e = \frac{1}{2} e^{-E/\eta}$

A binary receiver system receives a bit rate of 1 Mbps. The waveform amplitude is 5mV and the noise power spectral density is $0.5 * 10^{-11}$ W/Hz. Calculate the average bit error probability for ASK, BFSK, BPSK and DPSK

Example 2:

Sol: $R = 1\,Mbps \Rightarrow T_b = \frac{1}{R} = \frac{1}{10M} = \frac{1}{10^6} = 10^{-6}\ sec.$

$\frac{\eta}{2} = 0.5*10^{-11} \Rightarrow \eta = 1*10^{-11}\ w/Hz$, $A = 5mV$

$\Rightarrow E_s = \frac{A^2 T}{2} = \frac{(5*10^{-3})^2 * 10^{-6}}{2}$

$= 12.5*10^{-12}\ J$

$\rightarrow$ ASK $\Rightarrow P_e = \frac{1}{2} Q\left(\sqrt{\frac{E_s}{4\eta}}\right)$

$= \frac{1}{2} Q\left(\sqrt{\frac{12.5*10^{-12}}{4*1*10^{-11}}}\right) = \frac{1}{2} Q\left(\sqrt{0.3125}\right) = \frac{1}{2} Q(x)$

$= \frac{1}{2} * \frac{e^{-0.3125}}{\sqrt{\pi} \cdot \sqrt{0.3125}} = 0.369$

The next is to find the probability of FSK, PSK and DPSK

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Digital Communications (SCE3316)

Lecture 8
PSK and QAM

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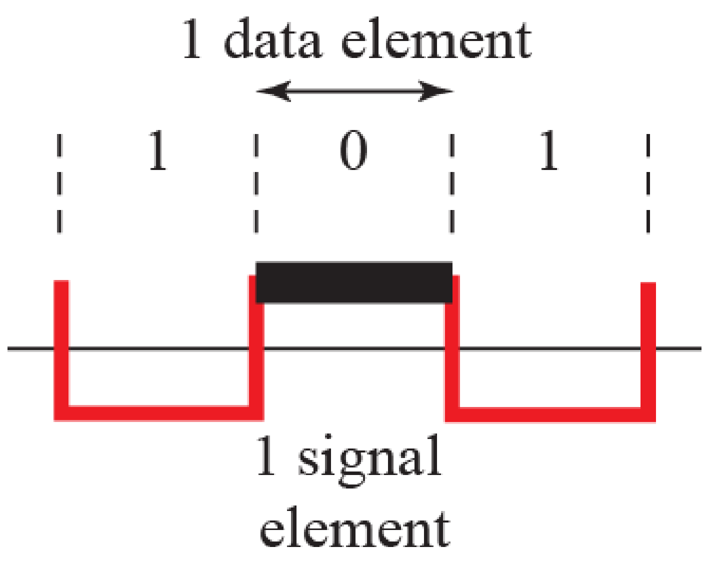
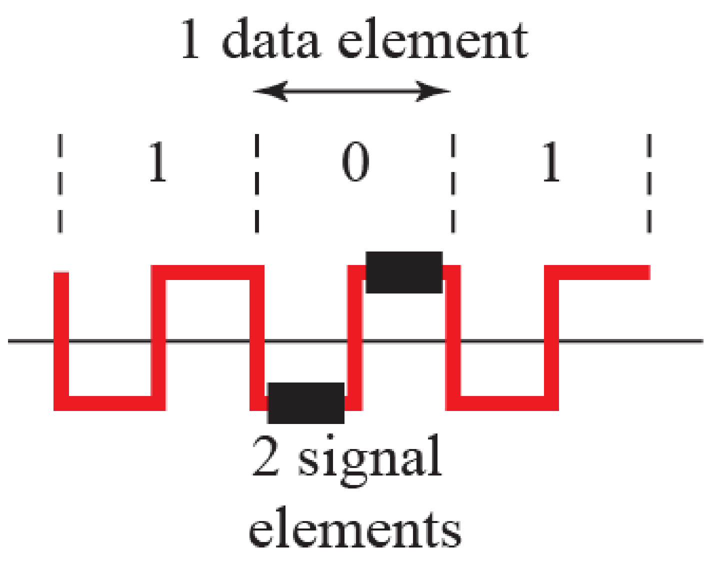
MODULATION FOR DATA COMMUNICATION: QPSK

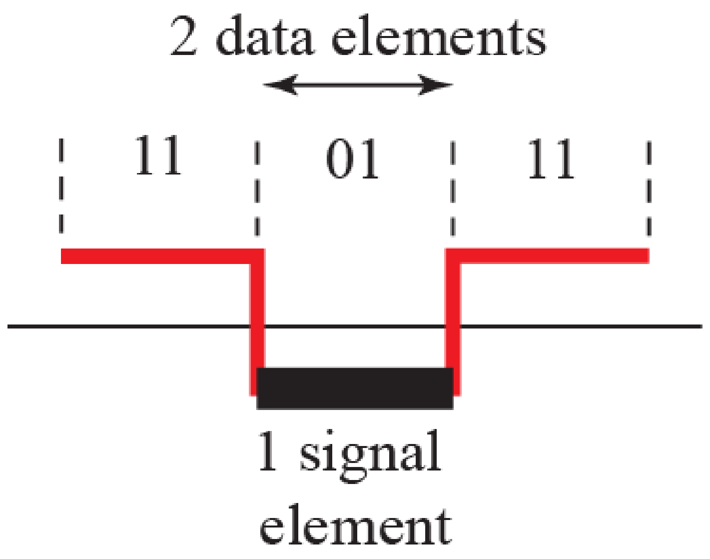
- One way to increase the binary data rate while not increasing the bandwidth required for the signal transmission is to encode more than 1 bit per phase change.
- In the system known as quadrature, quaternary, or quadra phase PSK (QPSK or 4-PSK), more bits per baud are encoded, the bit rate of data transfer can be higher than the baud rate, yet the signal will not take up additional bandwidth.
- In QPSK, each pair of successive digital bits in the transmitted word is assigned a particular phase.
- Each pair of serial bits, called a dibit, is represented by a specific phase.



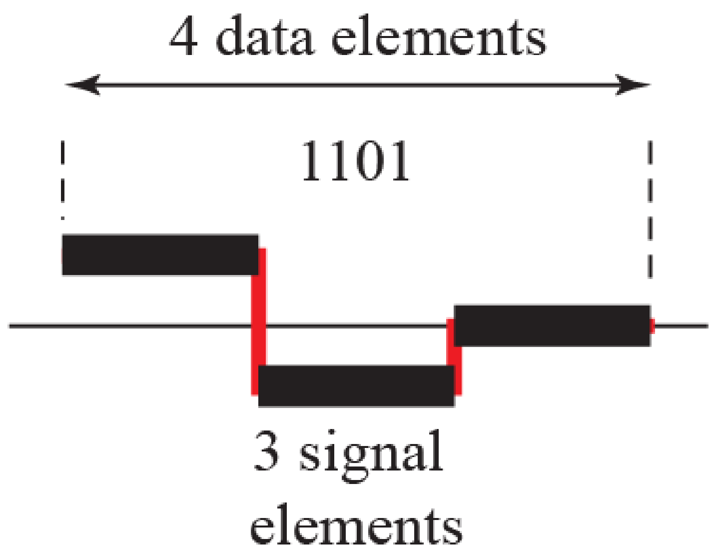
a. One data element per one signal element (r = 1)



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



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



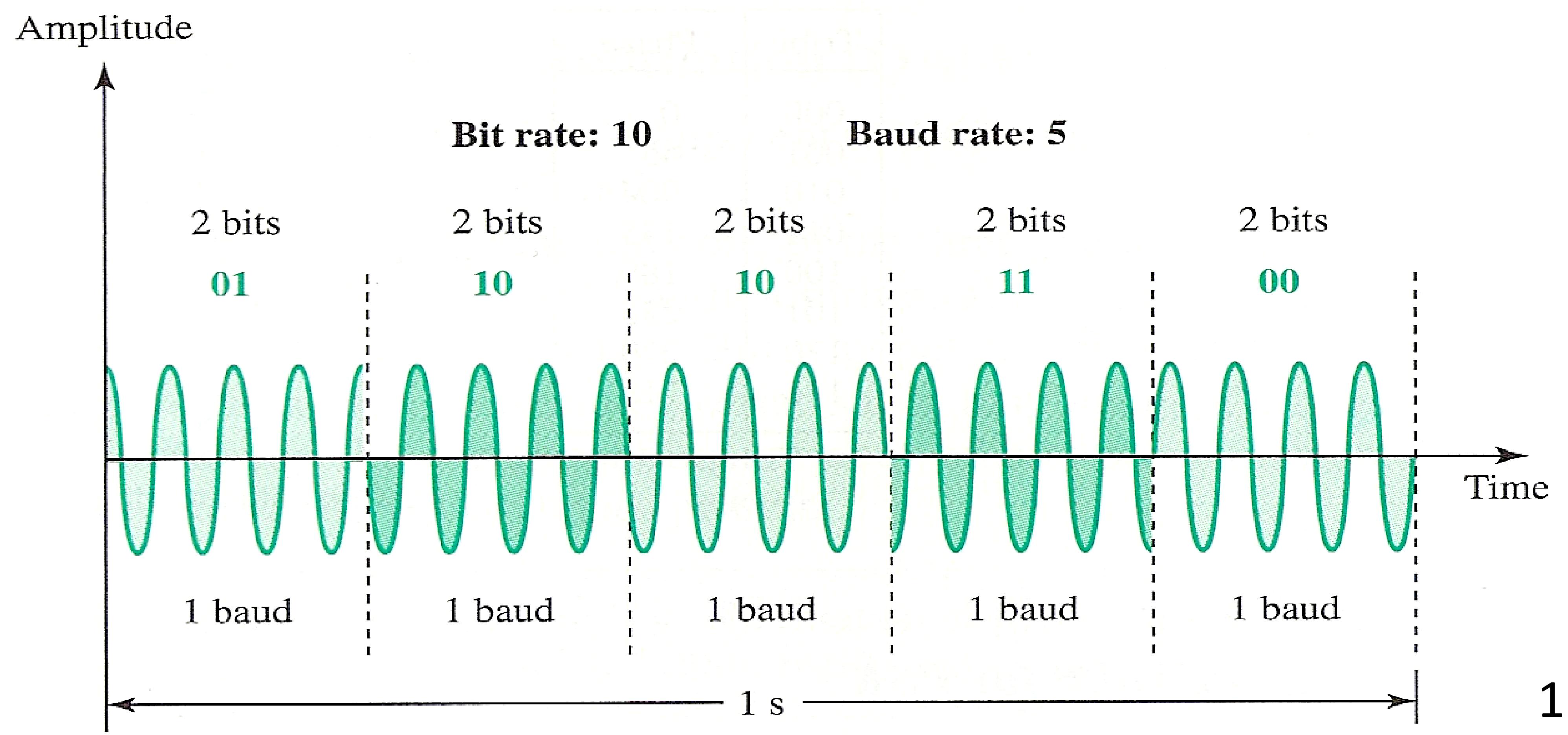
d. Four data elements per three signal elements $\left(r = \frac{4}{3}\right)$

Signal elements versus data elements

QUADRATURE PSK (QPSK)

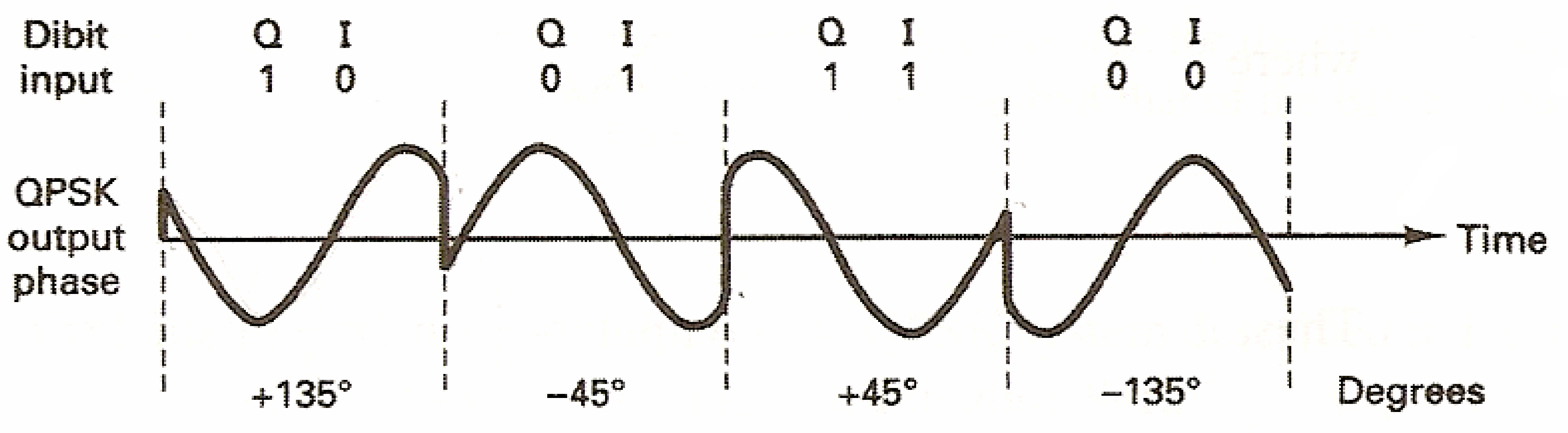
- The term “quadrature” implies that there are four possible phases (4-PSK) that the carrier can have at a given time.
- The pair of bits represented by each phase is called dibit.
- The rate of change (baud) in this signal determines the signal bandwidth.
- The throughput or bit rate for QPSK is twice the baud rate.

QPSK (Quadrature Phase-Shift Keying) is a kind of Phase Shift Keying.



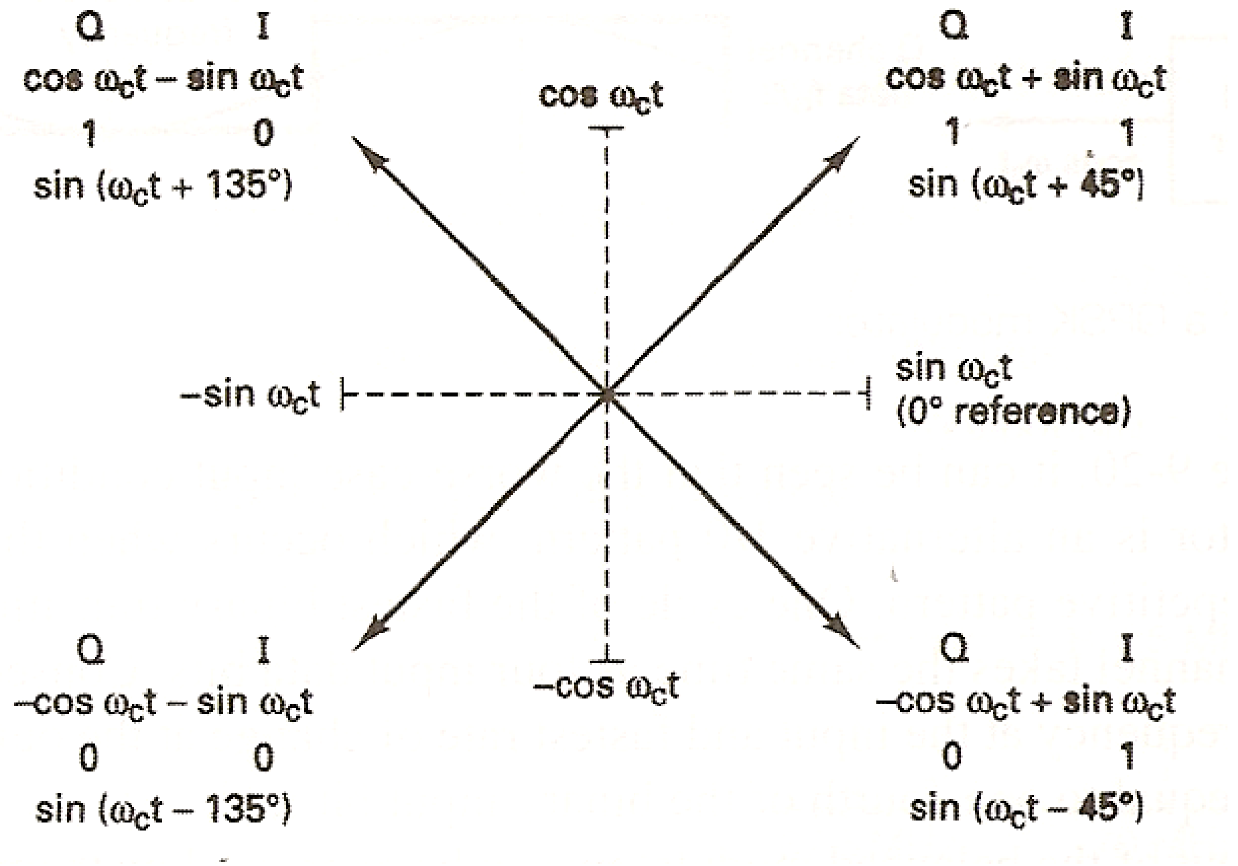
QPSK = 4-PSK

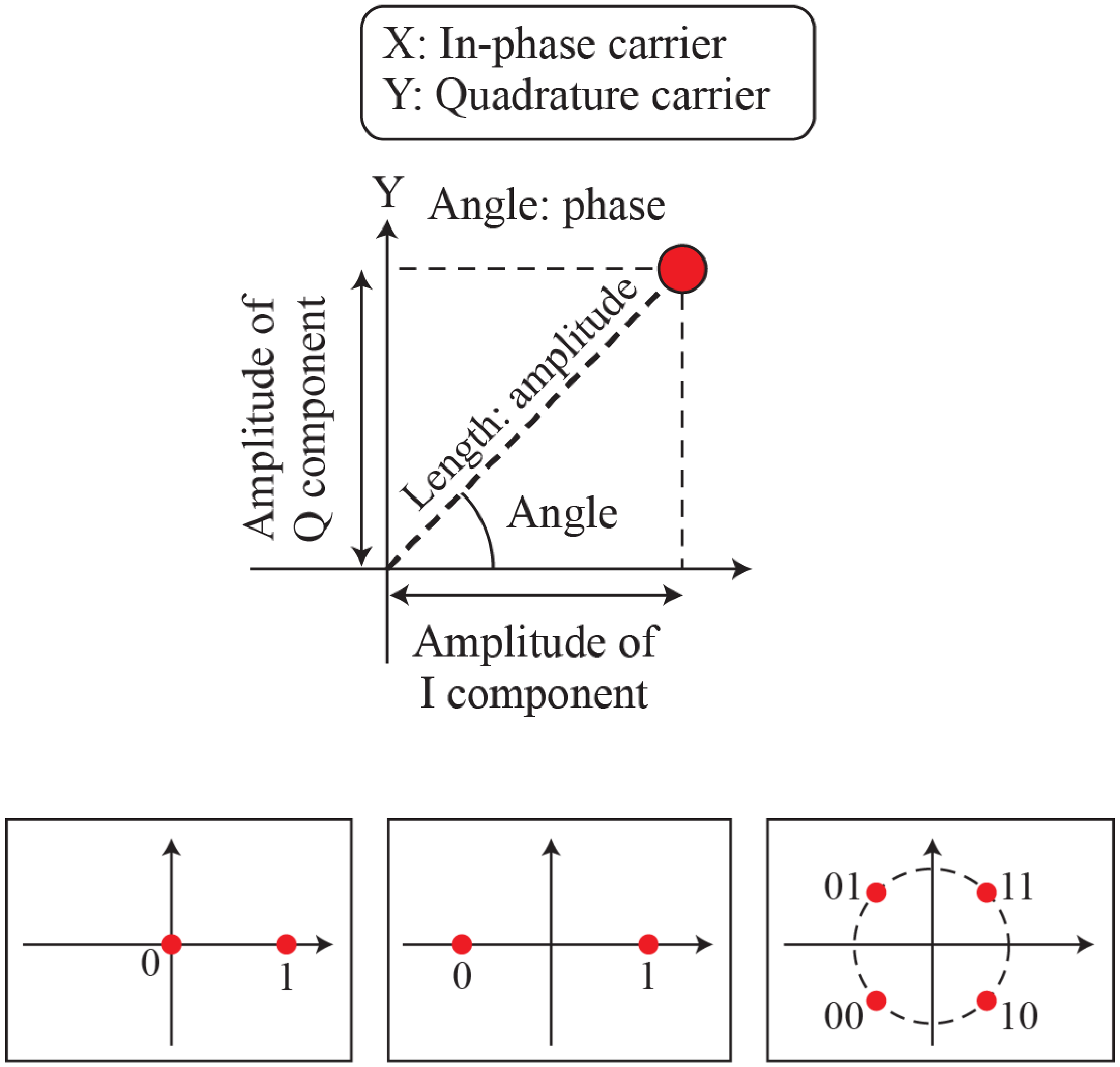
PSK that uses phase shifts of 90º=π/2 rad ⇒ 4 different signals are generated, each representing 2 bits



Assumption

Binary input Q	Binary input I	QPSK output phase
0	0	−135°
0	1	−45°
1	0	+135°
1	1	+45°



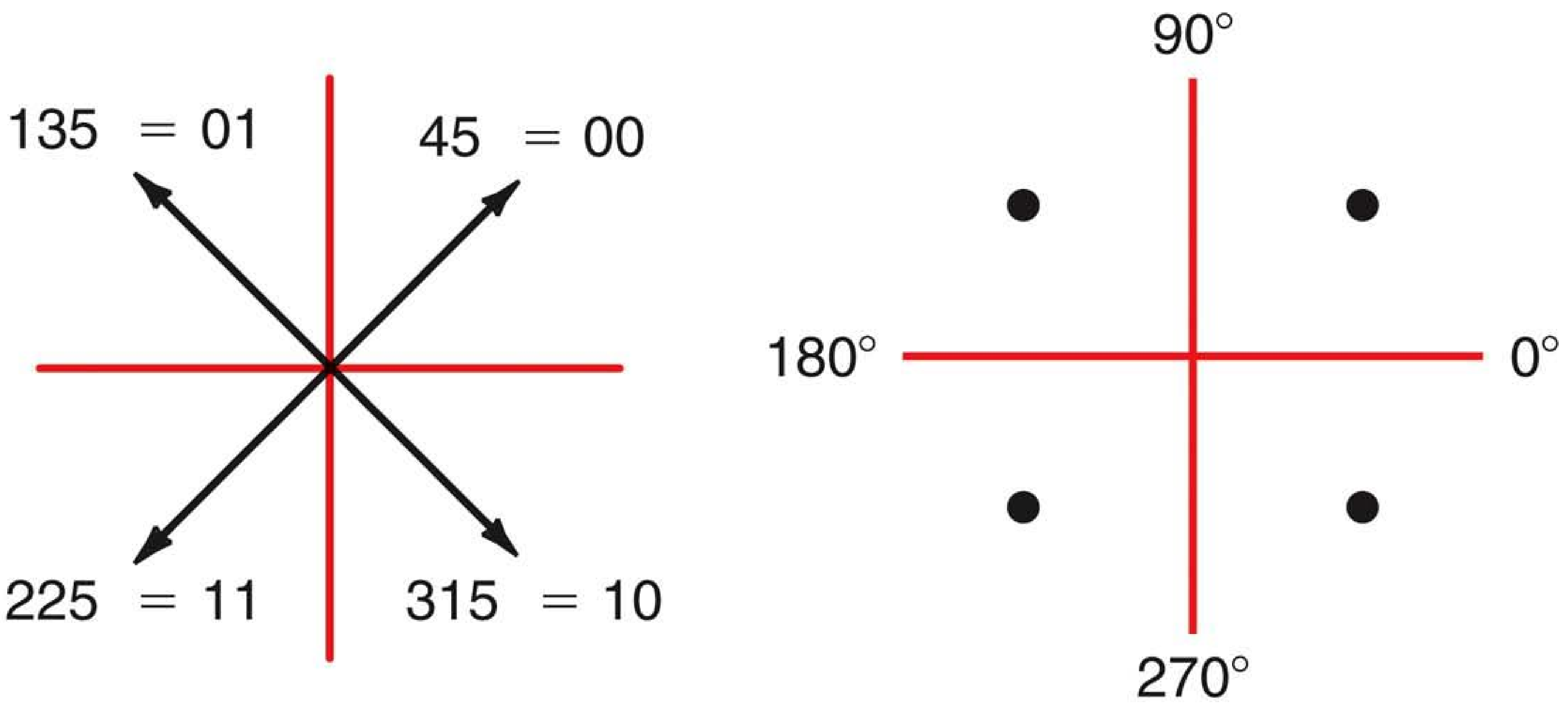


a. BASK (OOK) b. BPSK c. QPSK

Concept of a constellation diagram

Dibit		Phase shift
0	0	45
0	1	135
1	1	225
1	0	315

(*a*)



(*b*) (*c*)

Quadrature PSK modulation
(a) Phase angle of carrier for different pairs of bits. (b) Phasor representation of carrier sine wave. (c) Constellation diagram of QPSK.

QPSK BIT RATE, BAUD AND BANDWIDTH

Example – The CCITT V.22 (like Bell 212A) modem uses QPSK to send data at 1200 bits per second; What is the baud rate.

The CCITT V.22 (like Bell 212A) modem uses QPSK to send data at 1200 bits per second; however, the phases change only 600 times per second, conveying two bits per change - this is a 600-baud modem.

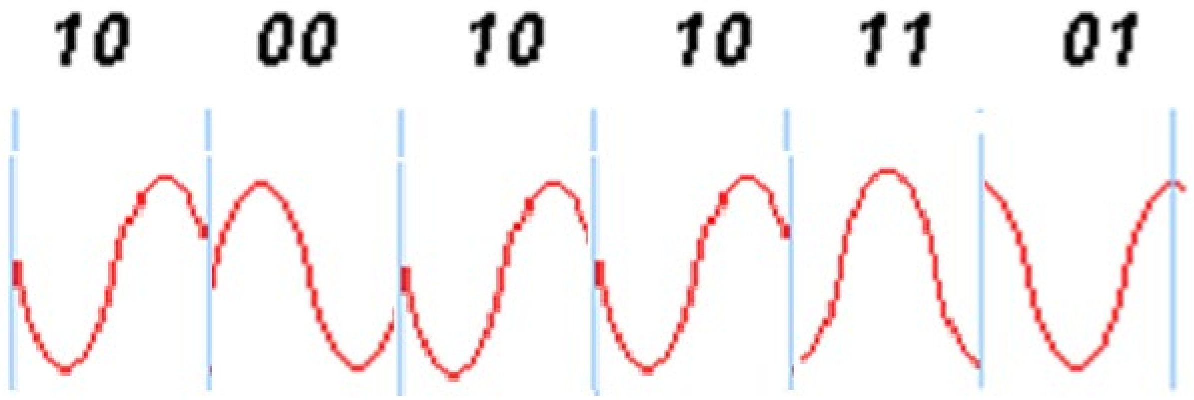
Example – For a QPSK system, with the following input bit sequence 100010101101, an input bit rate equal to 20Mbps,

a. Draw the QPSK modulated waveform. Start with output phase 0 for binary input 00.

b. Determine the minimum bandwidth required.

Solution – 100010101101

a.

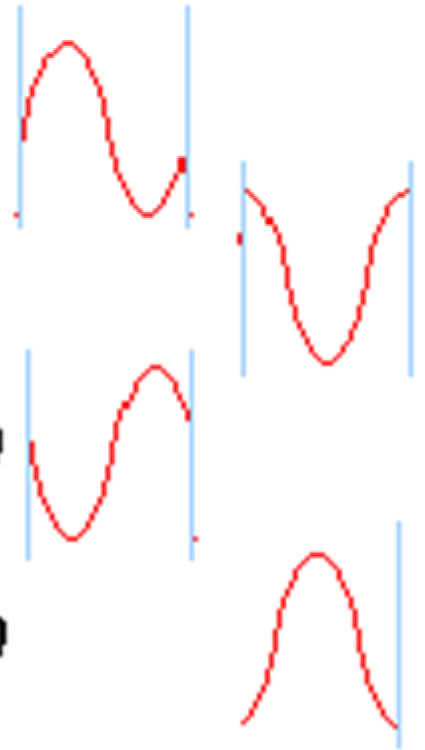


b. B = Rb/r

B = 20M/2 = 10MHz.

Assumption:

Binary Input Q	Binary Input I	QPSK output phase
0	0	0
0	1	90
1	0	180
1	1	270



QUADRATURE AMPLITUDE MODULATION (QAM)

PSK is limited by the ability of the equipment to distinguish small differences in phase. This factor limits its potential bit rate. So far, we have been altering only one of the three characteristics of a sine wave at a time, but what if we alter two? Why not combine ASK and PSK? The idea of using two carriers, one in-phase and the other quadrature, with different amplitude levels for each carrier is the concept behind quadrature amplitude modulation (QAM).

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QUADRATURE AMPLITUDE MODULATION (QAM)

The number of bits necessary to produce a given number of conditions is expressed mathematically as

$$N = \log_2 M$$

where N = number of bits necessary
M = number of conditions, levels

Mathematically, baud is the reciprocal of the time of one output signaling element, and a signaling element may represent several information bits. Baud is expressed as

$$\text{baud} = \frac{1}{t_s}$$

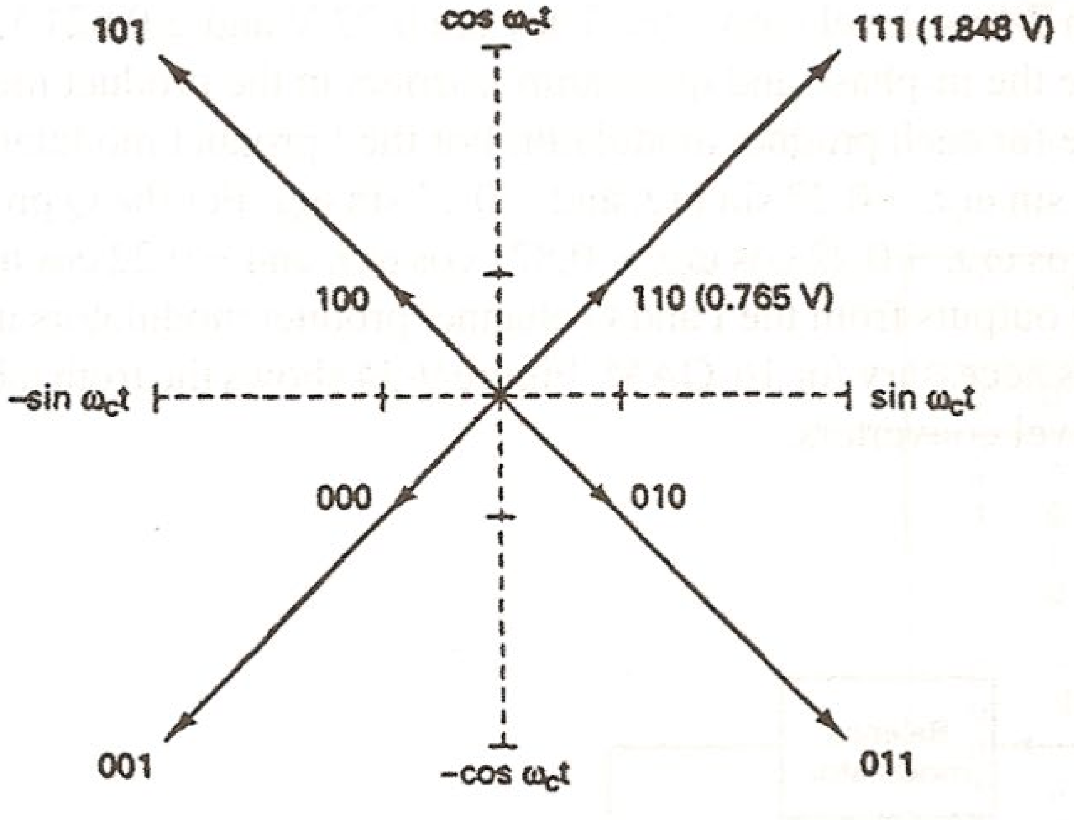
where baud = symbol rate (baud per second)
t_s = time of one signaling element (seconds)

QUADRATURE AMPLITUDE MODULATION (QAM)

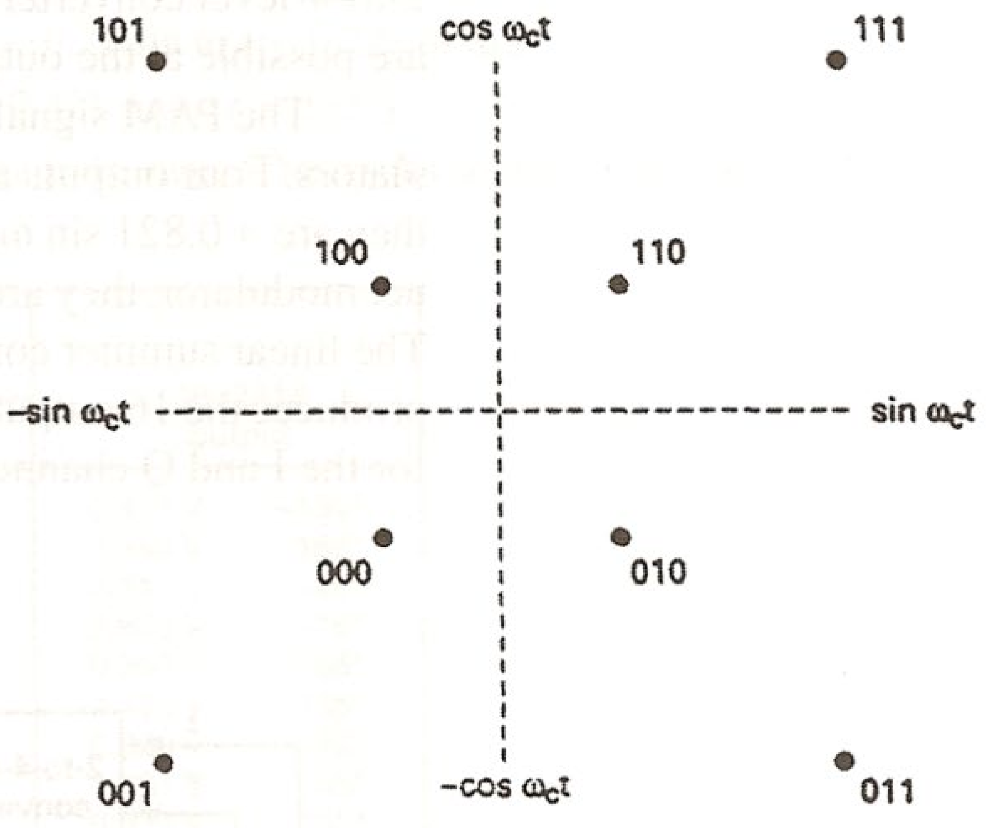
A combination of ASK and PSK: both phase and amplitude varies.

8-QAM

It is an M–ary encoding technique where M=8



(b)



(c)

Binary input			8-QAM output	
Q	I	C	Amplitude	Phase
0	0	0	0.765 V	–135°
0	0	1	1.848 V	–135°
0	1	0	0.765 V	–45°
0	1	1	1.848 V	–45°
1	0	0	0.765 V	+135°
1	0	1	1.848 V	+135°
1	1	0	0.765 V	+45°
1	1	1	1.848 V	+45°

QAM Modulator a) Truth Table b) Phasor diagram c) Constellation diagram

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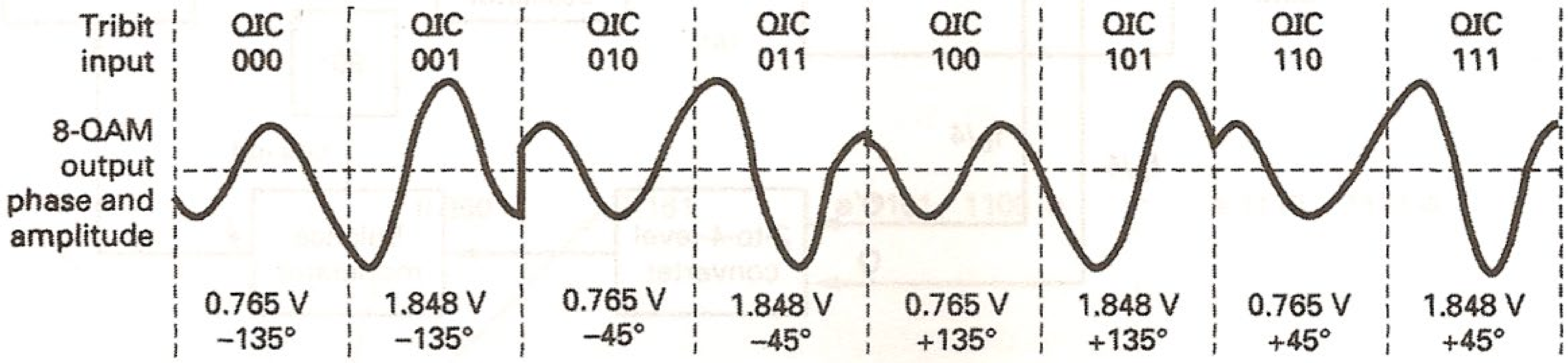
QUADRATURE AMPLITUDE MODULATION (QAM)

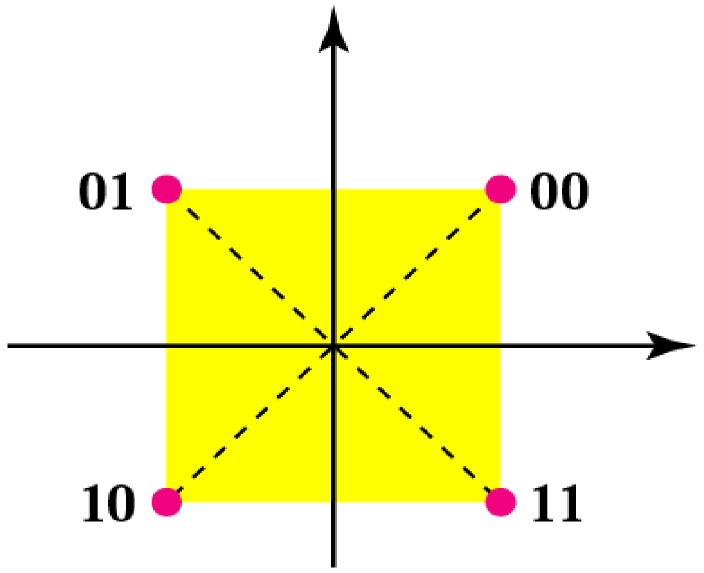
8-QAM



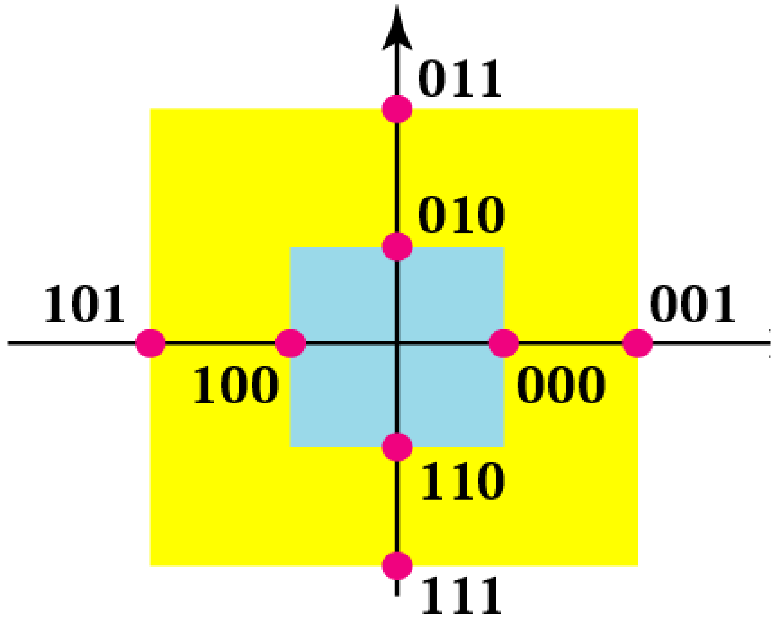
Output phase and amplitude versus-time relationship for 8-QAM

Min. bandwidth required = Rb/3

QUADRATURE AMPLITUDE MODULATION (QAM)

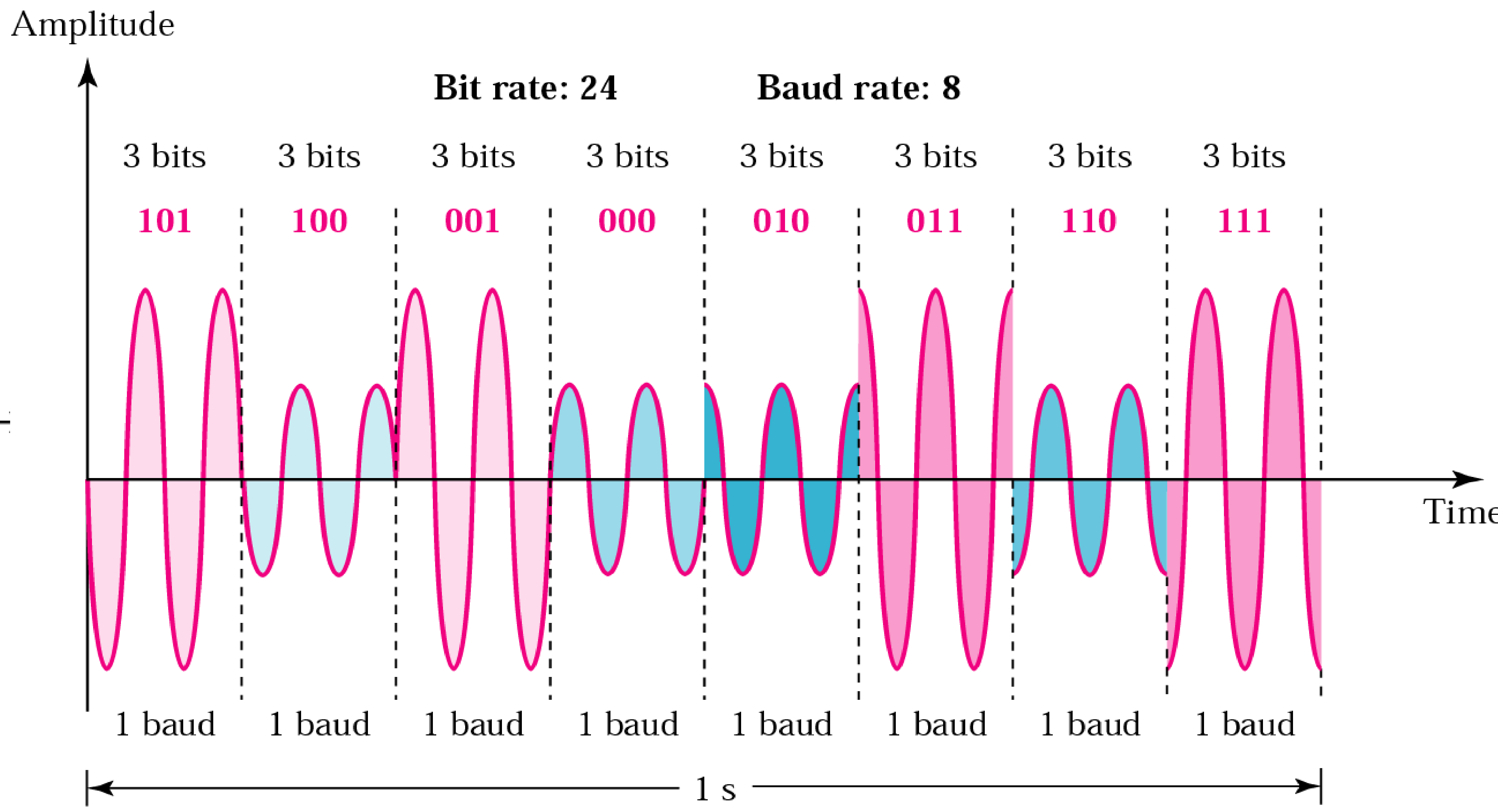


4-QAM
1 amplitude, 4 phases



8-QAM
2 amplitudes, 4 phases

8-QAM



Example: Calculate the bit rate and baud rate of BPSK and 8-PSK if the bit duration (Tb) and signaling time (Ts) are both 0.01msec.

BPSK

Bit rate = baud rate

= 1/0.01

= 100kbps

Or 100kbaud/sec

8-PSK

baud rate = 1/0.01

= 100kbaud/s

Then bit rate (Rb)

= 3x100k

=300kbps

BANDWIDTH EFFICIENCY

- It is used to compare the performance of one digital modulation technique to another
- Ratio of the transmission bit rate to the minimum bandwidth required for a particular modulation scheme.

$$B\eta = \frac{transmission\ bitrate(bps)}{\min.\ bandwidth(Hz)}$$

BANDWIDTH EFFICIENCY

Example: For an 8-PSK system operating with an information bit rate of 24kbps, determine:

a) baud rate
b) min. bandwidth
c) bandwidth efficiency

a. baud = 24 kbps / 3 = 8000

b. B = 24 kbps / 3 = 8000

c. $B\eta$ = 24, 000 / 8000
= 3 bits per second per cycle of bandwidth