



# Signal Transmission Fundamentals

**Advanced Computing and Networking  
Introduction**

**Joint master program Skoltech and CMC MSU**

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# Signals, Data, Transmission



- **Data** - a description of facts, phenomena
- **Signals** - presentation of data during transmission
- **Transmission** is the process of interaction between a transmitter and a receiver in order to transmit a signal.

## Data

- **Origin data** can take many forms
  - analog vs digital
- **Analog data:** voice video
- **Discrete data (digital)**
  - text: letter, symbol
  - picture: pixel

## Signals

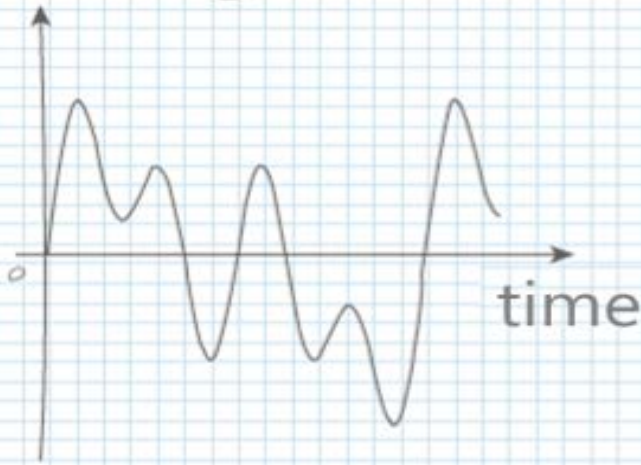
- **Signals** - analog vs digital



# The mathematical representation of the signal



voltage



Harmonic = simple oscillation

$$S(t) = \sin(\omega_1 t) + \sin(\omega_2 t) + \sin(\omega_3 t), \text{ где } \omega = 2\pi f$$

The mathematical signal representation

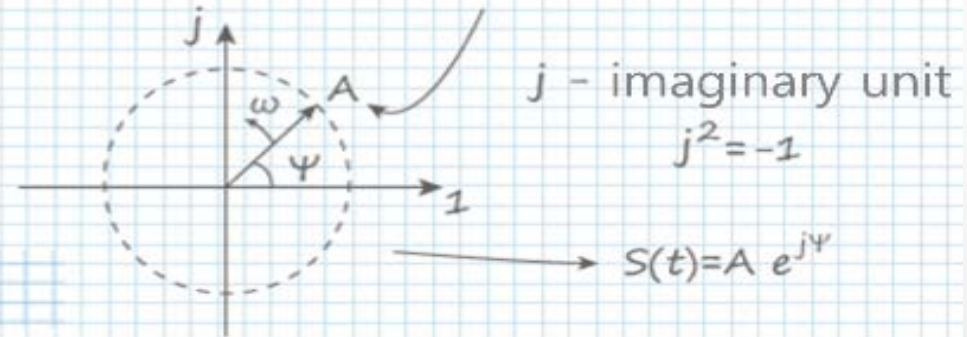
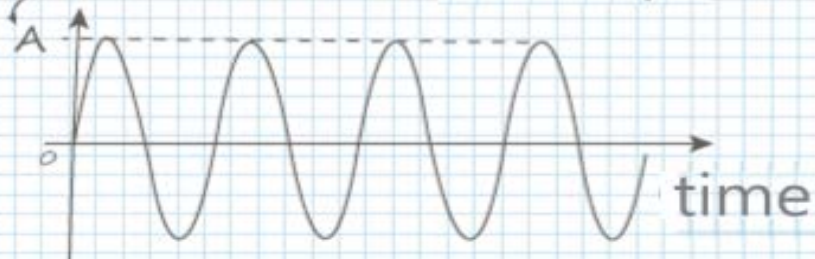
harmonic

can be represented as a rotating vector

$$S(t) = A \cdot \sin(\omega t + \psi)$$

amplitude

Initial phase



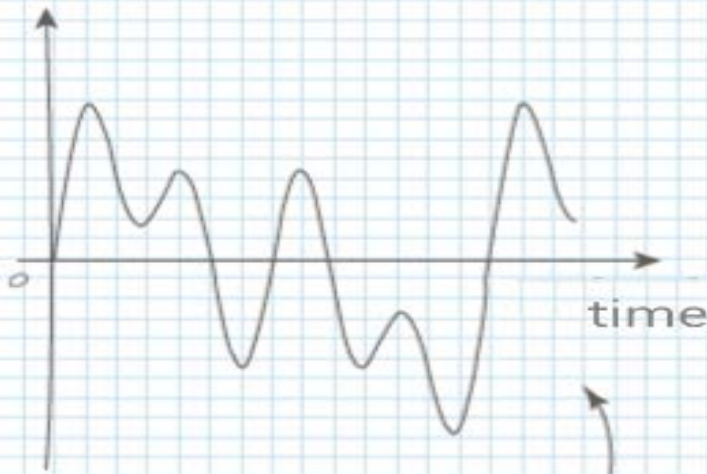




# Signal spectrum



voltage



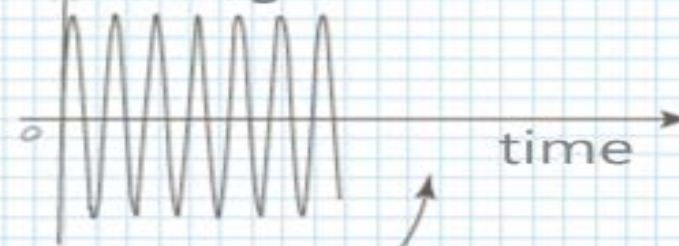
voltage



voltage

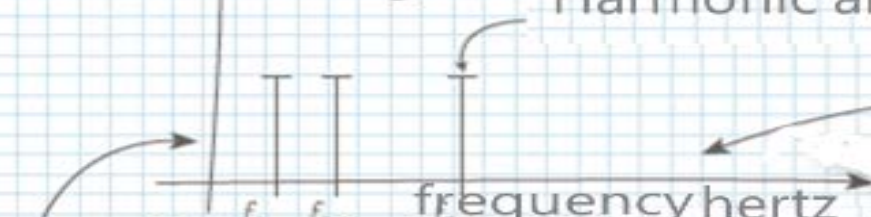


voltage



a complex signal can be decomposed into simple oscillations

voltage



Harmonic amplitude

The distribution diagram of simple oscillations in the signal = spectrum

each simple oscillation has its own frequency hertz = one oscillation





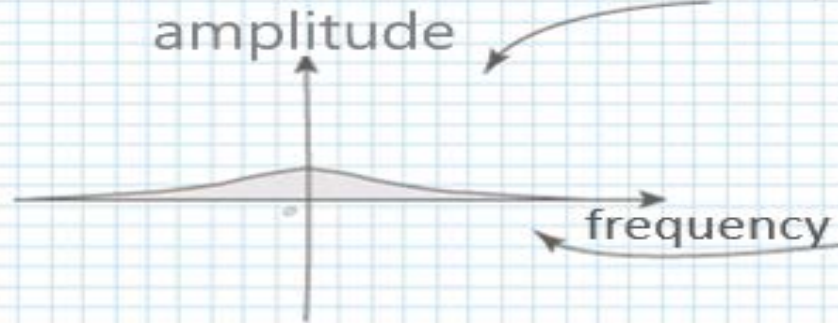
# Fourier transformation



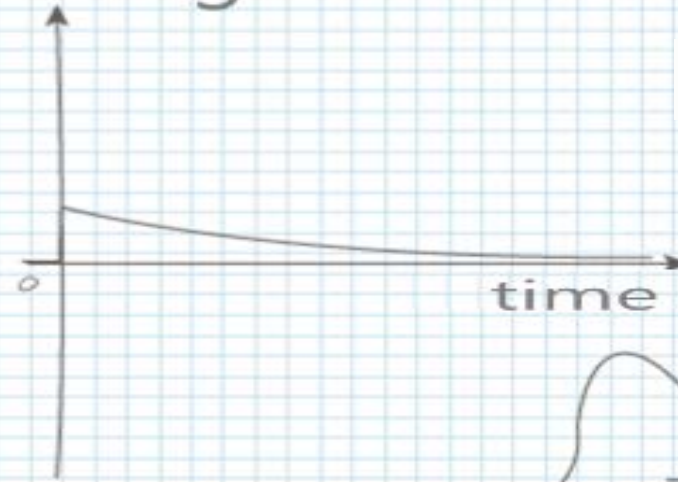
connects a signal and its spectrum voltage

$$S(\omega) = \int_{-\infty}^{+\infty} s(t) e^{-j\omega t} dt$$

Direct transformation



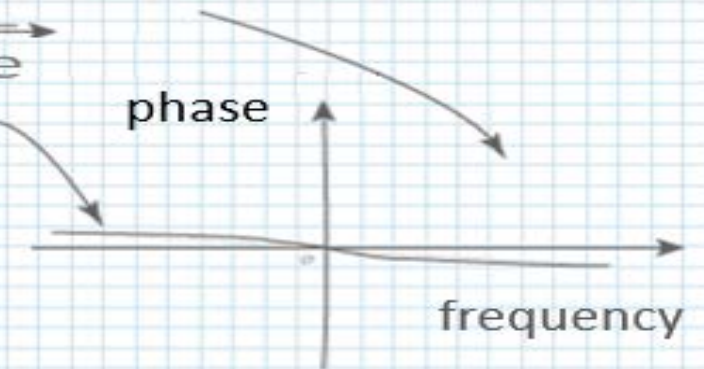
amplitude spectrum



envelope

**Euler's formula**

$$e^{iy} = \cos y + i \sin y.$$



phase spectrum

$$s(t) = \frac{1}{2\pi} \int_{-\infty}^{+\infty} S(\omega) e^{j\omega t} d\omega$$

Invers transformation



# Discrete Fourier Transformation

Direct Fourier Transformation:

$$X_k = \sum_{n=0}^{N-1} x_n e^{-\frac{2\pi i}{N} kn} \quad k = 0, \dots, N-1$$

Invers Fourier Transformation :

$$x_n = \frac{1}{N} \sum_{k=0}^{N-1} X_k e^{\frac{2\pi i}{N} kn} \quad n = 0, \dots, N-1.$$

$N$  is the number of signal values measured over a period, as well as the number of decomposition components;

$x_n$ ,  $n = 0, \dots, N - 1$ , are the measured signal values (in discrete time  $N - 1$ ), which are input data for direct transformation and output for the inverse one;

$X_k$ , ( $k = 0, \dots, N - 1$  are the indexes of the amplitudes in a complex form of the sinusoidal signals composing the original one) are output for direct conversion and input for inverse; since the amplitudes are complex, then they can be used to calculate both the amplitude and phase;

$k$  is the frequency index.

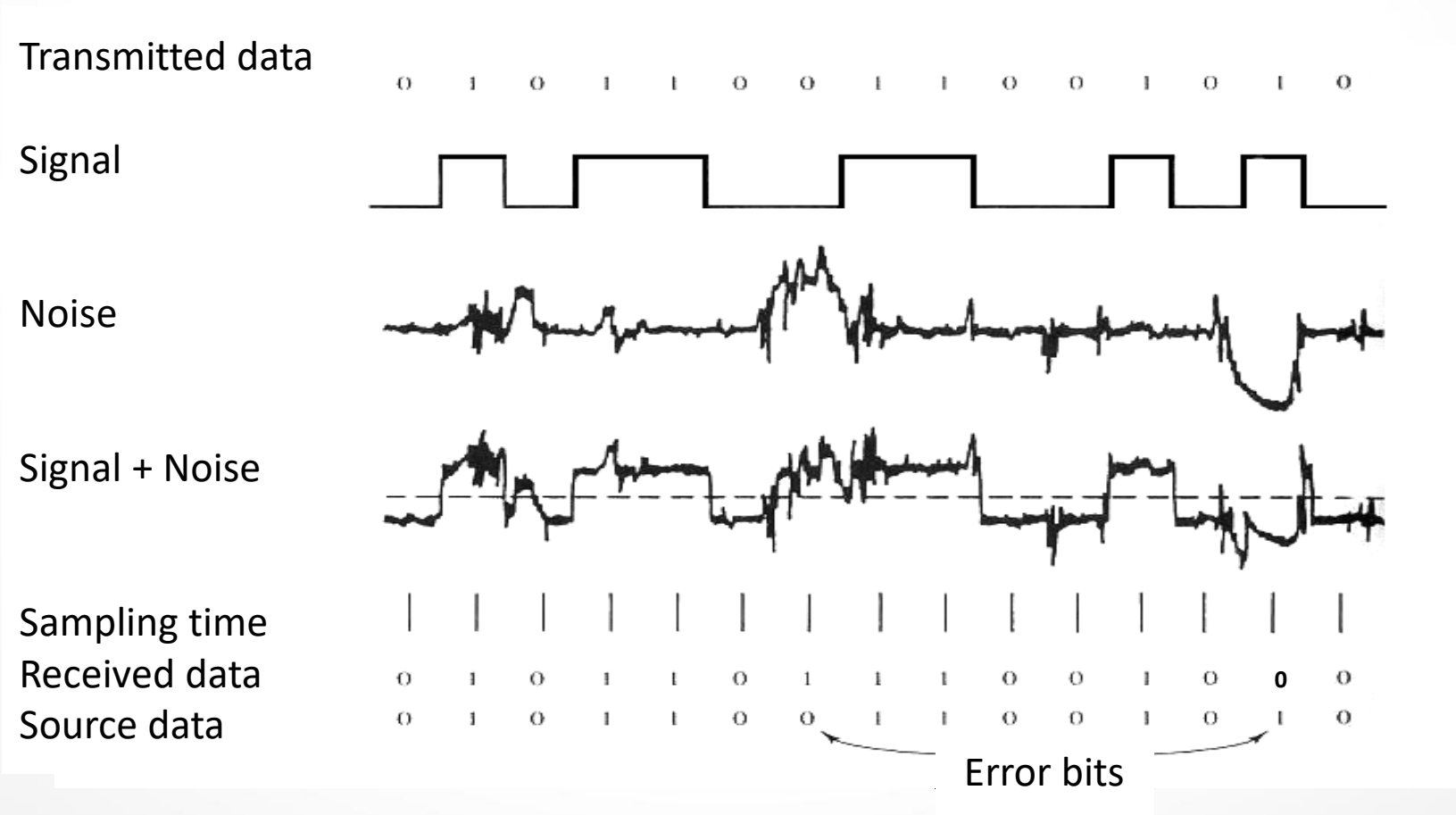
The frequency of the  $k$ -th signal is  $k / T$ , where  $T$  is the period of time during which the input data was taken.



# Effect of noise on analogue vs on digital signals



Digital signal  
The form is known in advance





# Analog vs Digital Transmission

- attenuation and distortion in the digital case is not as strong as in the analog case;
- with digital signal, it is easier to restore its original shape, which is known exactly, unlike an analog one;
- the error accumulates in analog signal;
- digital transmission is more reliable: the wave form is known.
- over a digital network, you can transfer as data as voice as music and at a higher speed compare to analog one.
- digital transmission is cheaper because it is not necessary to restore the wave form.
- a digital network is easier to manage.





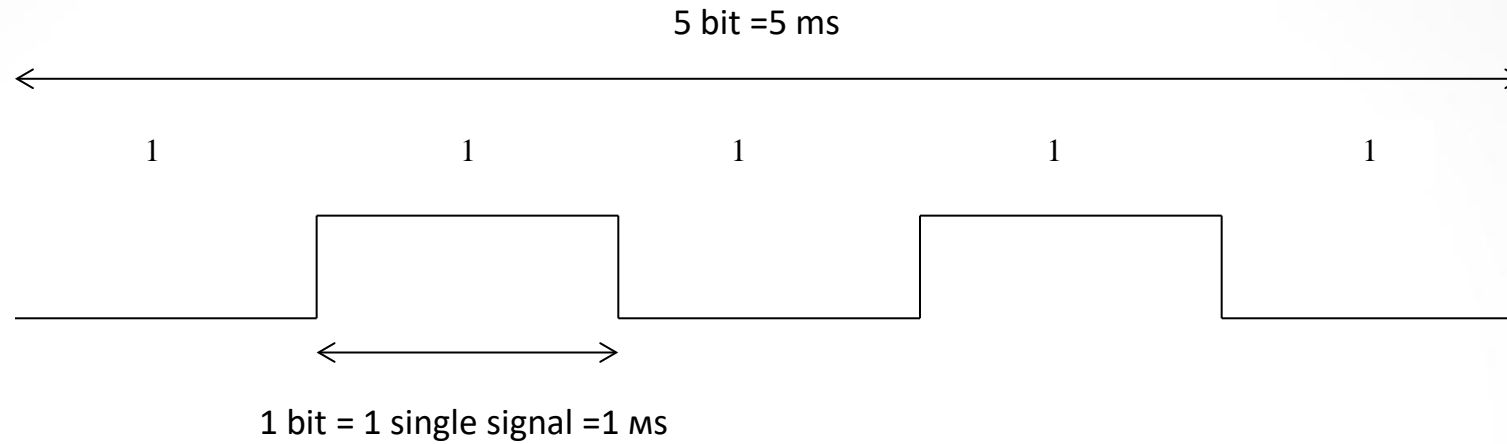
# Channel media, bandwidth and capacity

- Different media distort the shape and quench its energy depending on the frequency of the signal in different ways.
- Different media distort the waveform of the signal and damp its energy in different ways depending on the signal frequency.
- Channel bandwidth - the spectrum of frequencies that a channel passes without a significant reduction in signal power.
- The channel capacity (transmission speed)\_ depends on the method of encoding data at the physical level and the signal speed - the rate of change of the signal value. This rate of change of signal per second is measured in baud. (J.M.Baudot)

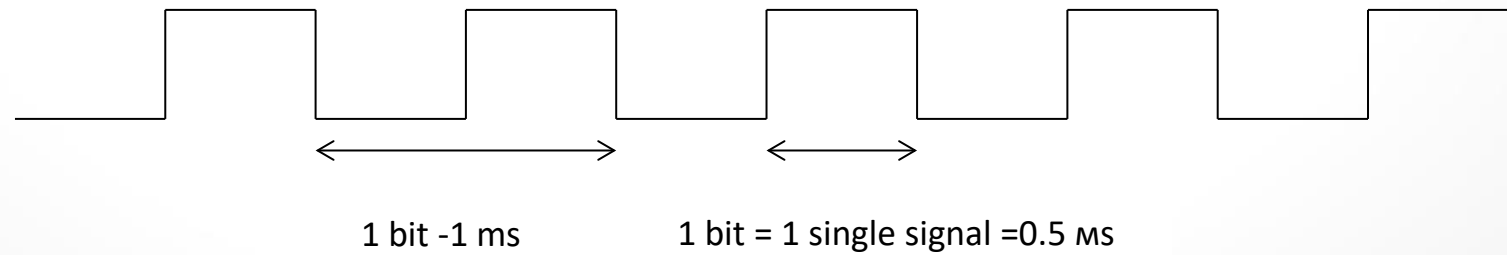


# Signal speed

NRZ I



Manchester





# The relationship of channel capacity and its bandwidth



**Channel capacity** - the maximum speed at which the channel is capable of transmitting data.

Interconnection between channel capacity and its bandwidth determines **Nyquist-Kotelnikov Theorem (1924)**

$$R_{\text{max data rate}} = 2D \log_2 L, \text{ where}$$

$D$  - channel bandwidth (the maximum frequency in the signal spectrum),

$L$  - number of signal levels.

## **Kotelnikov Theorem - Signal Recovery**

The analog signal  $u(t)$ , which does not contain frequencies higher than  $F_{\text{max}}$  (Hz), is completely determined by the sequence of its values at times spaced  $1 / (2F_{\text{max}})$  apart from each other.





# Attenuation

- The attenuation coefficient of the medium characterizes the attenuation (in power) over 1km long segment of the medium.
- Measured in decibels (dB). The decibel [dB] is determined for the power ratio :

$$10 * \log_{10} Z, \text{ where } Z \text{ is the power ratio between signal and noise.}$$

- The attenuation in the line is equal to the product of the attenuation coefficient by the line

$$\text{length: } z(L) = \alpha L$$

- To measure power, the decibel unit (dBm) is used. It is equal to the ratio of the signal power of 1 mW to the noise power on the line, expressed in dB

$$p[\text{dB}] = 10 \log_{10} Z[\text{mW}]$$

- Difference between input and output signal powers (in dB) there is attenuation in an element of the medium. For example in fiber:

$$\Delta p_{LIN} = p_{IN} - p_{OUT} = \alpha L$$



# The relationship of channel capacity and its bandwidth in case of noise

- **Shannon's limit** for a channel with noise is channel noise is measured as the ratio of useful signal power to noise power:

$$S/N \text{ (measured in 1dB} = 10 \log_{10} (S / N)).$$

- The Shannon limit is treated as the maximum transmission rate for which it is possible (to choose a signal-code design) to correct errors in a channel with a given signal to noise ratio:

$$R_{max} = D \log_2 (1 + S / N) \text{ bps,}$$

where  $S / N$  is the signal-to-noise ratio in the channel; here the number of levels in the signal doesn't matter. (This is a theoretical limit that is rarely achieved in practice.)

- **Direct Theorem:** If the message rate is less than the communication channel capacity ( $R < C$ ), then there are codes  $\{x_1, x_2, \dots, x_M\}$  and decoding methods such that the average and maximum decoding error probabilities tend to zero when the block length tends to infinity.
- **Inverse Theorem** If the transmission rate is greater than the channel capacity, that is,  $R > C$ , then there are no such transmission methods in which the probability of error tends to zero (Per  $\rightarrow 0$ ) with increasing length of the transmitted block, ( $L \rightarrow \infty$ ).



# Presentation of data on the physical level



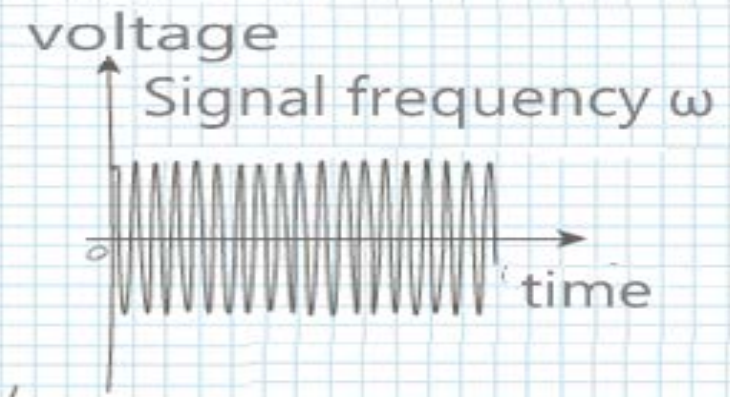
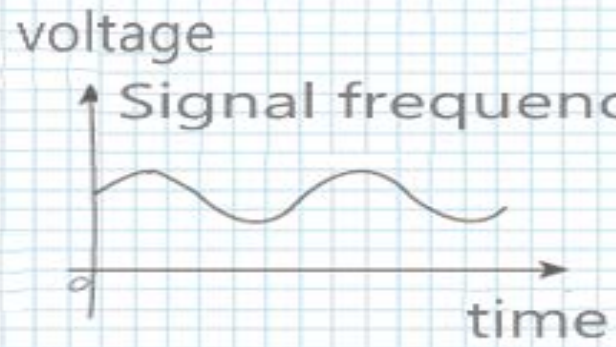


# Analog representation (Modulation)

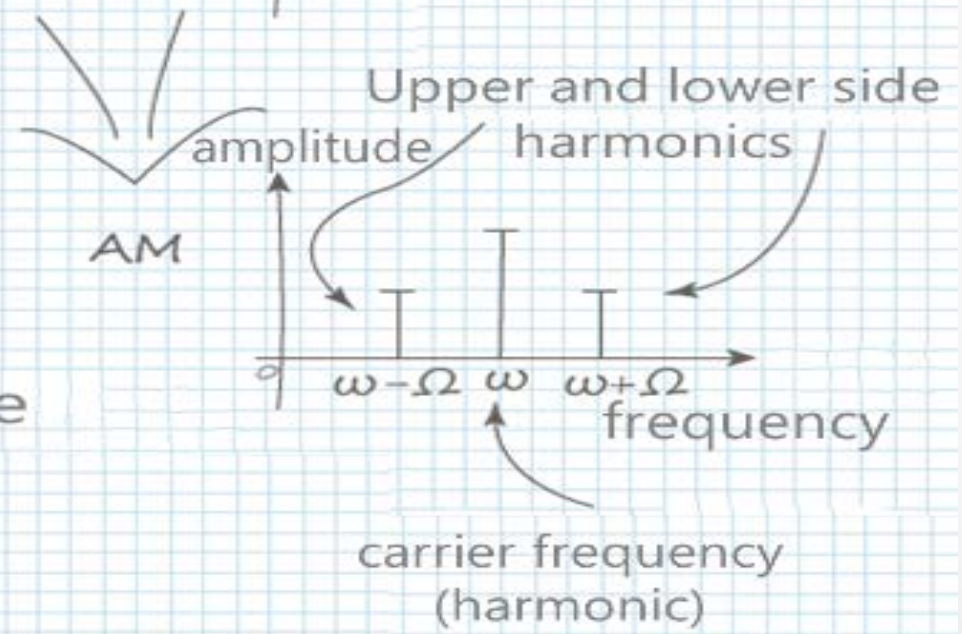
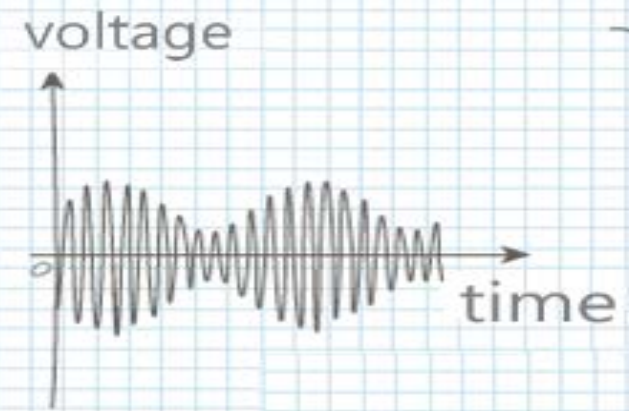


$$S(t) = A \cdot \sin(\omega t + \Psi)$$

amplitude      frequency      phase  
modulations



+



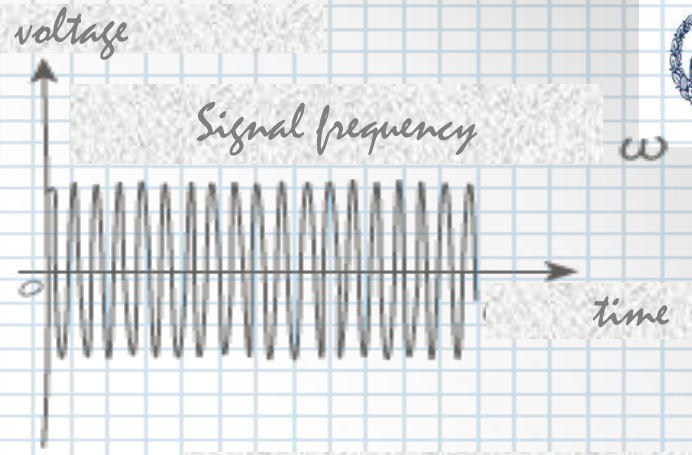
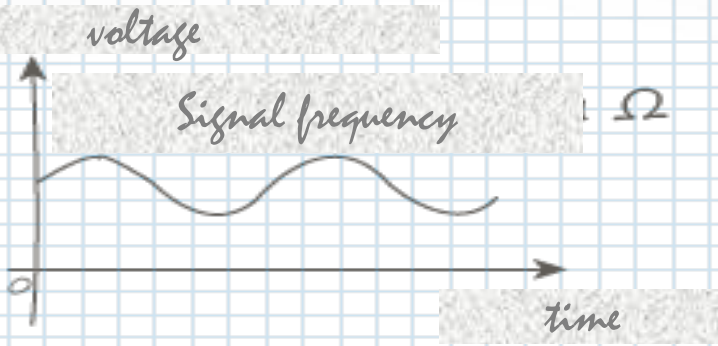


# Analog representation

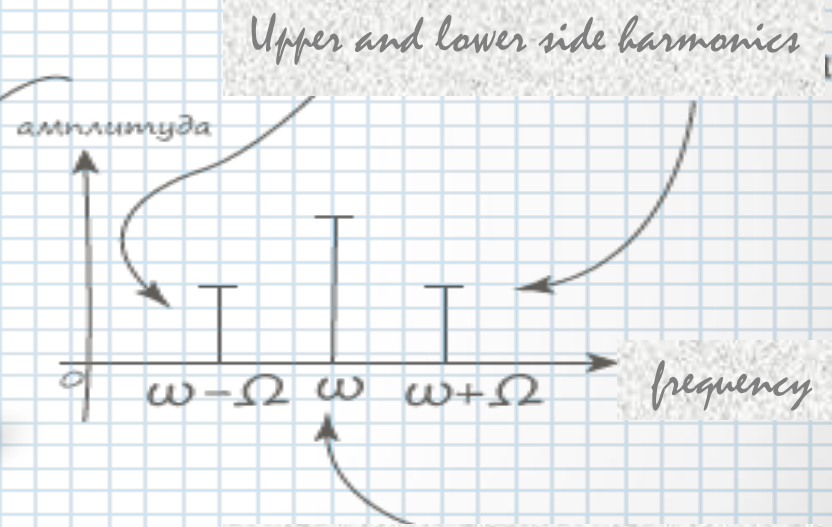
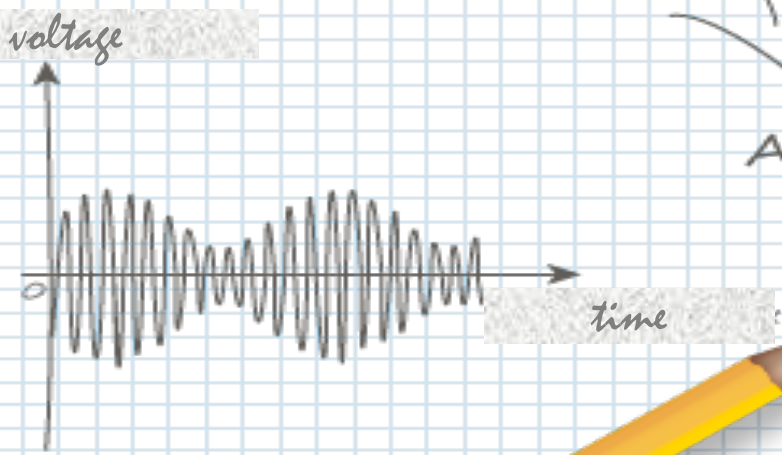


$$S(t) = A \cdot \sin(\omega t + \varphi)$$

amplitude frequency phase modulations



+



$$s(t) = s(1 + M \cos(\omega t))$$

carrier frequency (harmonic)



# Digital representation (manipulation)

Potential Code NRZ (Non Return to Zero)

0 - high potential

1 - low potential

Bipolar Code NRZI

0 - there is no difference in signal level at the beginning of the bit interval

1 - signal level difference at the beginning of the interval

AMI Bipolar Code (Alternate Mark Inversion)

0 - no signal

1 - positive or negative potential, inverse to the potential in the previous period

Manchester code

0 - transition from high to low potential in the middle of the interval

1 - transition from low to high potential in the middle of the interval

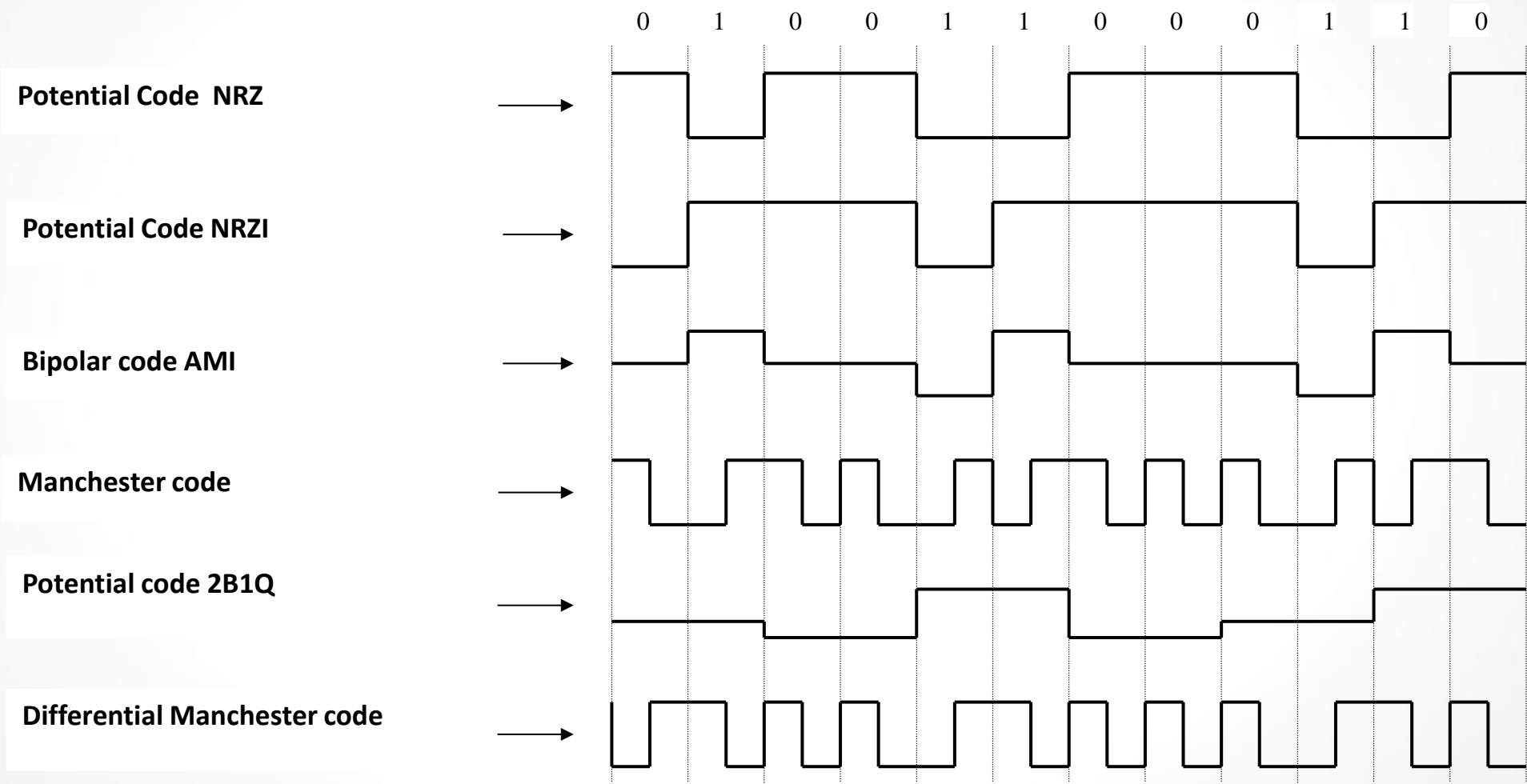
Potential Code 2B1Q

Uses 4 signal levels, the level value determines the value of a pair of data bits





# Code examples

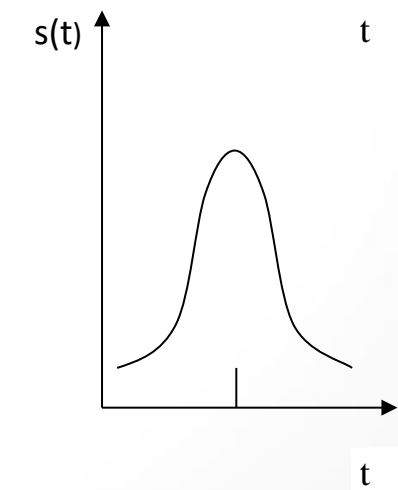
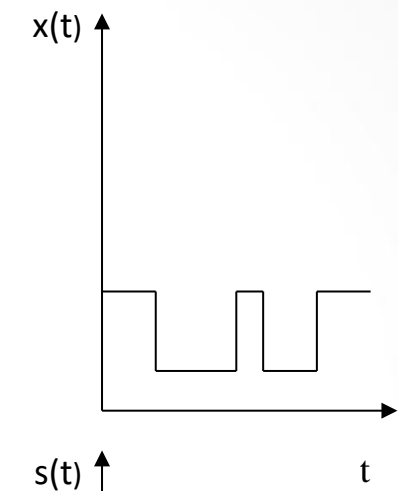
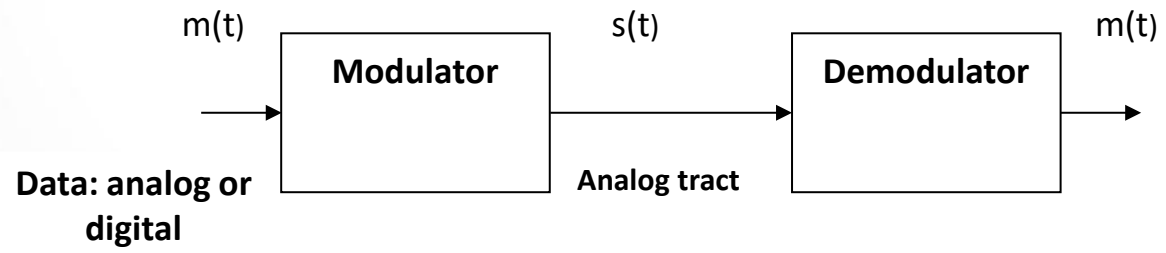
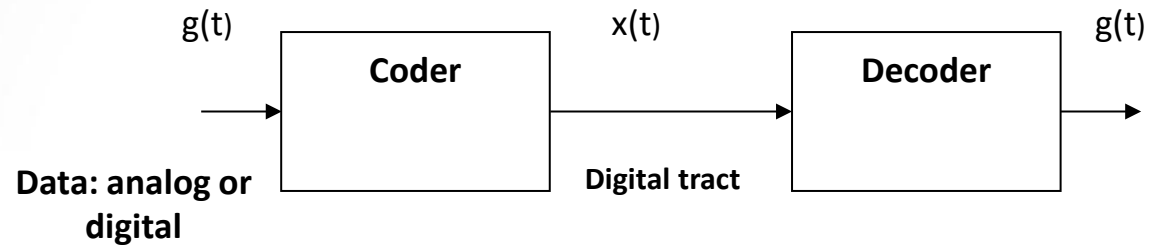




# Types of data & signal combinations



# The scheme of analog and digital transmission





# Signals & Data combinations



- digital data - analogue signal (modem)
- analogue data - digital signal (digitization)
- digital data - digital signal (number of signal levels)
- analog data – analog signal (matching the frequency spectra)



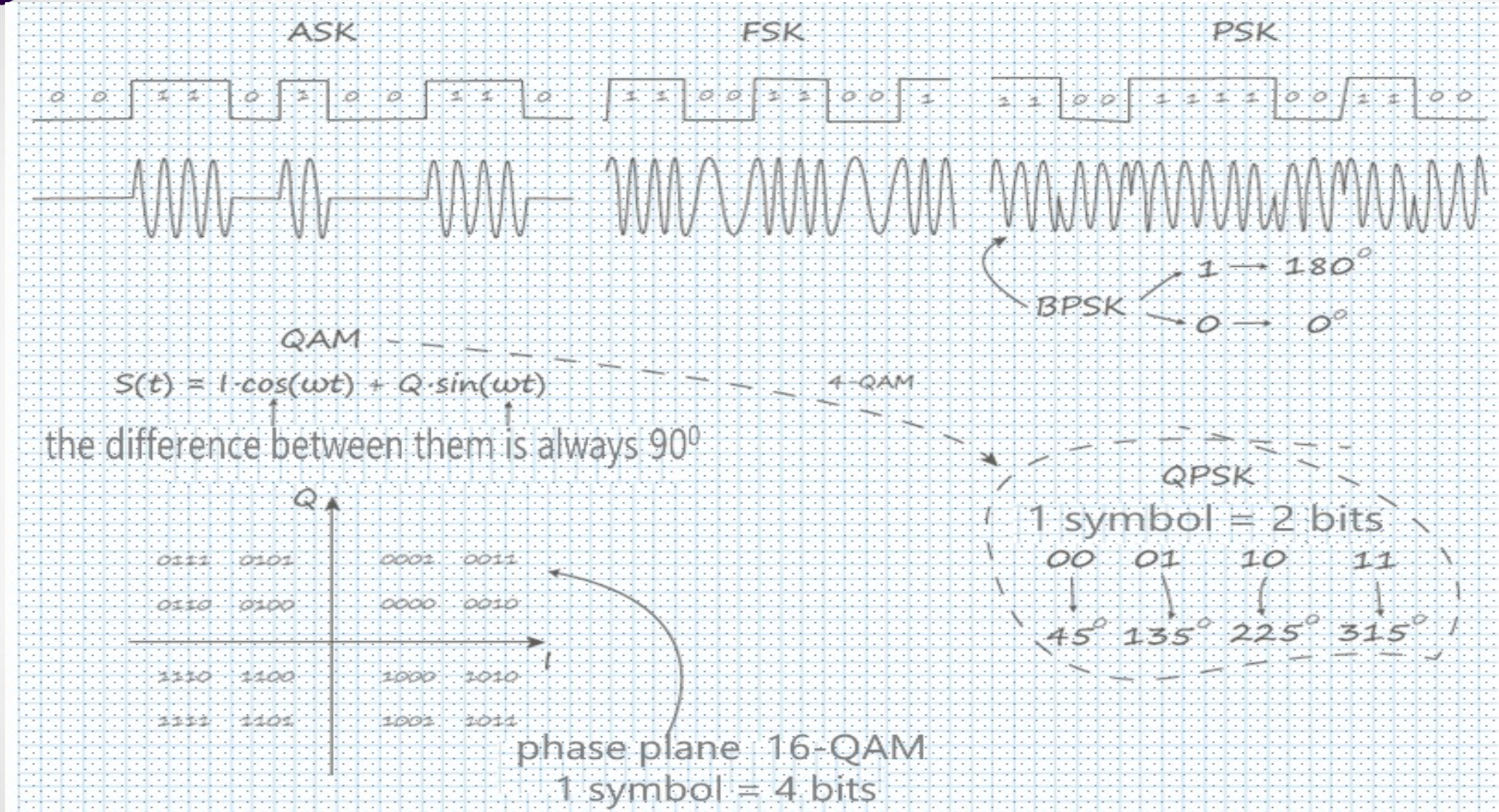


# Digital data – Analog signal

- Telephone networks were created for the transmission and switching of analog signals in the voice frequency range from 300 to 3400 Hz.
- A modem (Modulator – DEM modulator) converts a digital signal into an analog signal in the appropriate frequency range and vice versa.
- There are three main modulation methods for converting digital data into analog form:
  - amplitude modulation
  - frequency modulation
  - phase modulation.



# Digital Data Modulation scheme

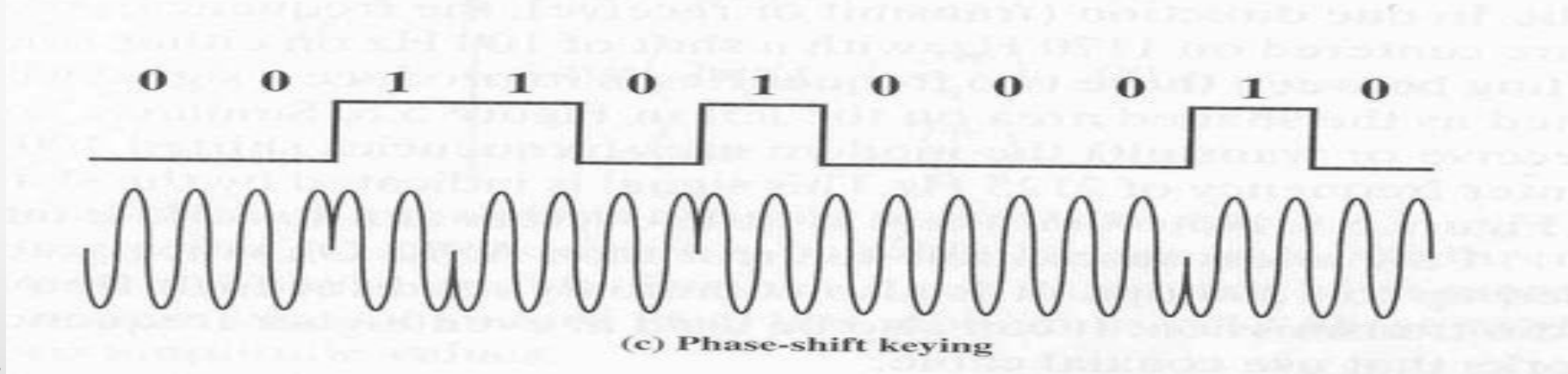
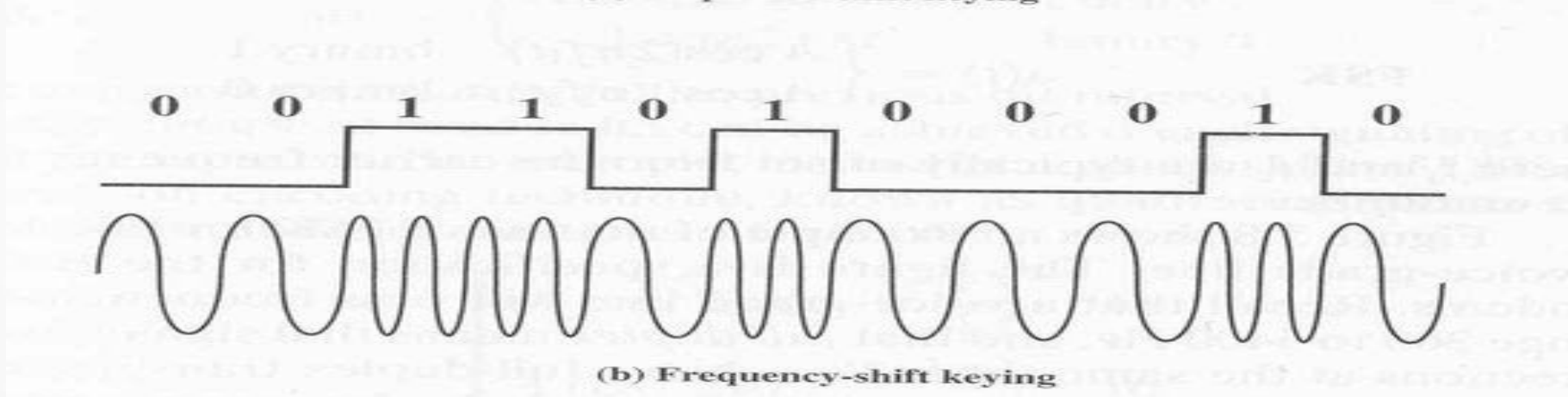
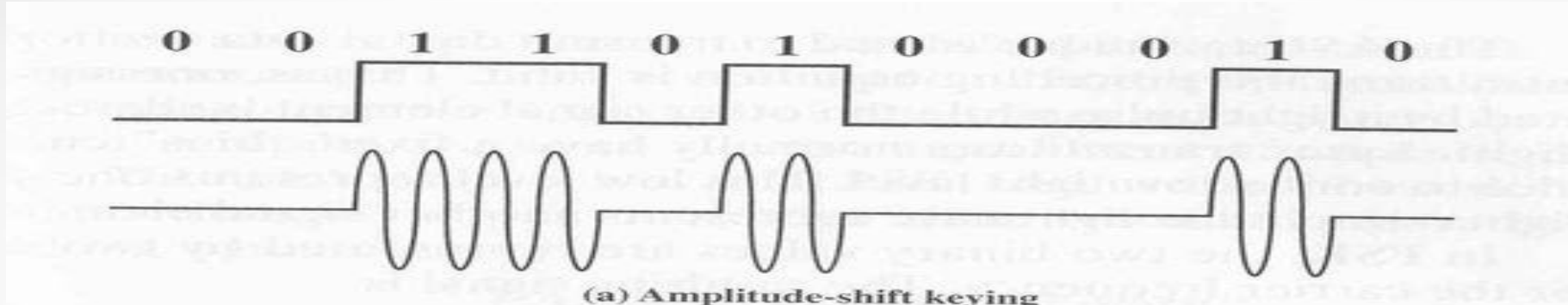


Media dependency





# Digital data – Analog signal

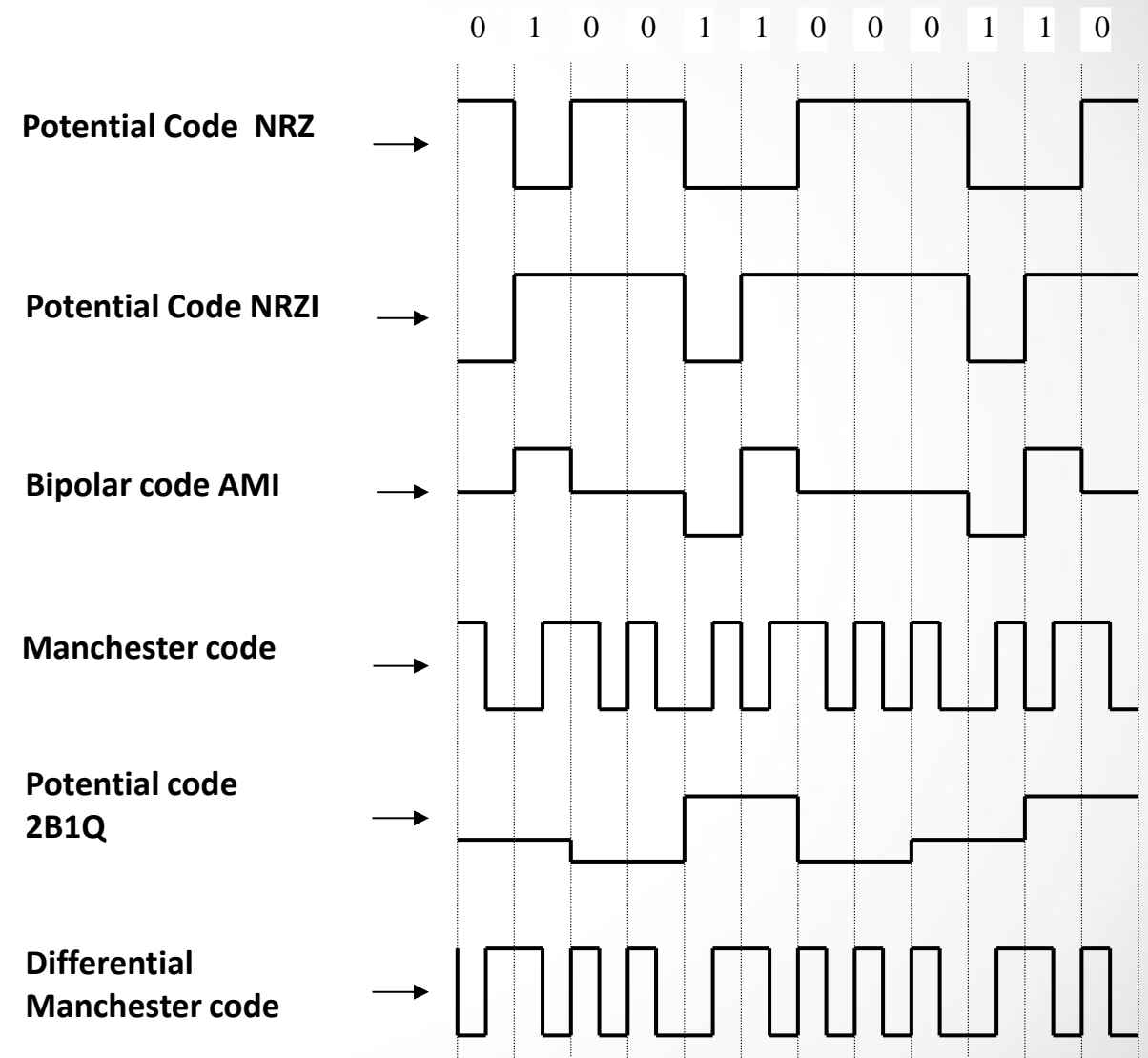




# Digital data - Digital signal



<p>Potential Code NRZ (Non Return to Zero)</p> <p>0 - high potential</p> <p>1 - low potential</p>
<p>Bipolar Code NRZI</p> <p>0 - there is no difference in signal level at the beginning of the bit interval</p> <p>1 - signal level difference at the beginning of the interval</p>
<p>AMI Bipolar Code (Alternate Mark Inversion)</p> <p>0 - no signal</p> <p>1 - positive or negative potential, inverse to the potential in the previous period</p>
<p>Manchester code</p> <p>0 - transition from high to low potential in the middle of the interval</p> <p>1 - transition from low to high potential in the middle of the interval</p>
<p>Potential Code 2B1Q</p> <p>Uses 4 signal levels, the level value determines the value of a pair of data bits</p>





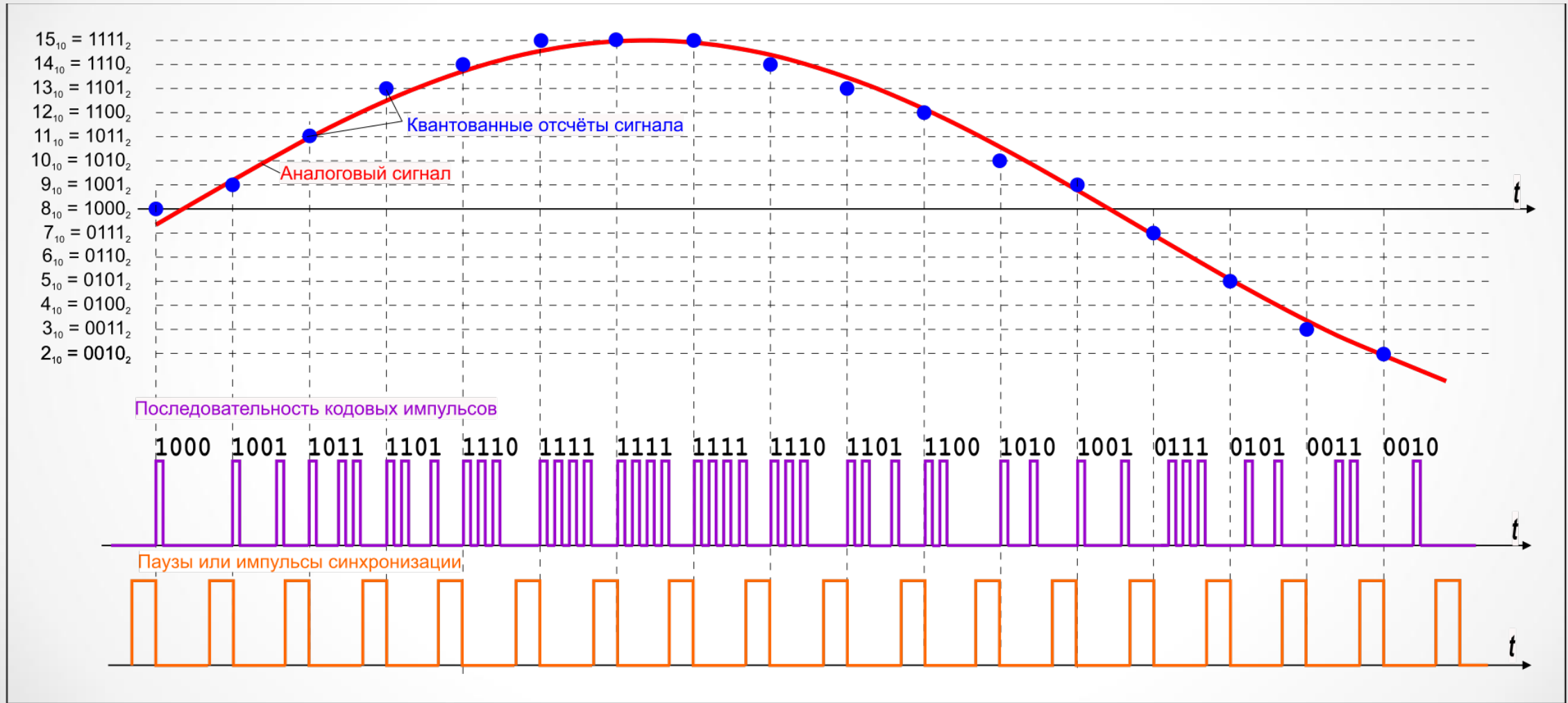


# Analog data – Digital signal

- ADC (Analog to Digital Converter) converts analog data into digital form
- DAC (Digital-to-Analog Converter) performs the reverse procedure
- A device that combines the functions of both an ADC and a DAC is called a codec (codec)
- The two main methods for converting an analog signal to digital are:
  - pulse code modulation and
  - delta modulation

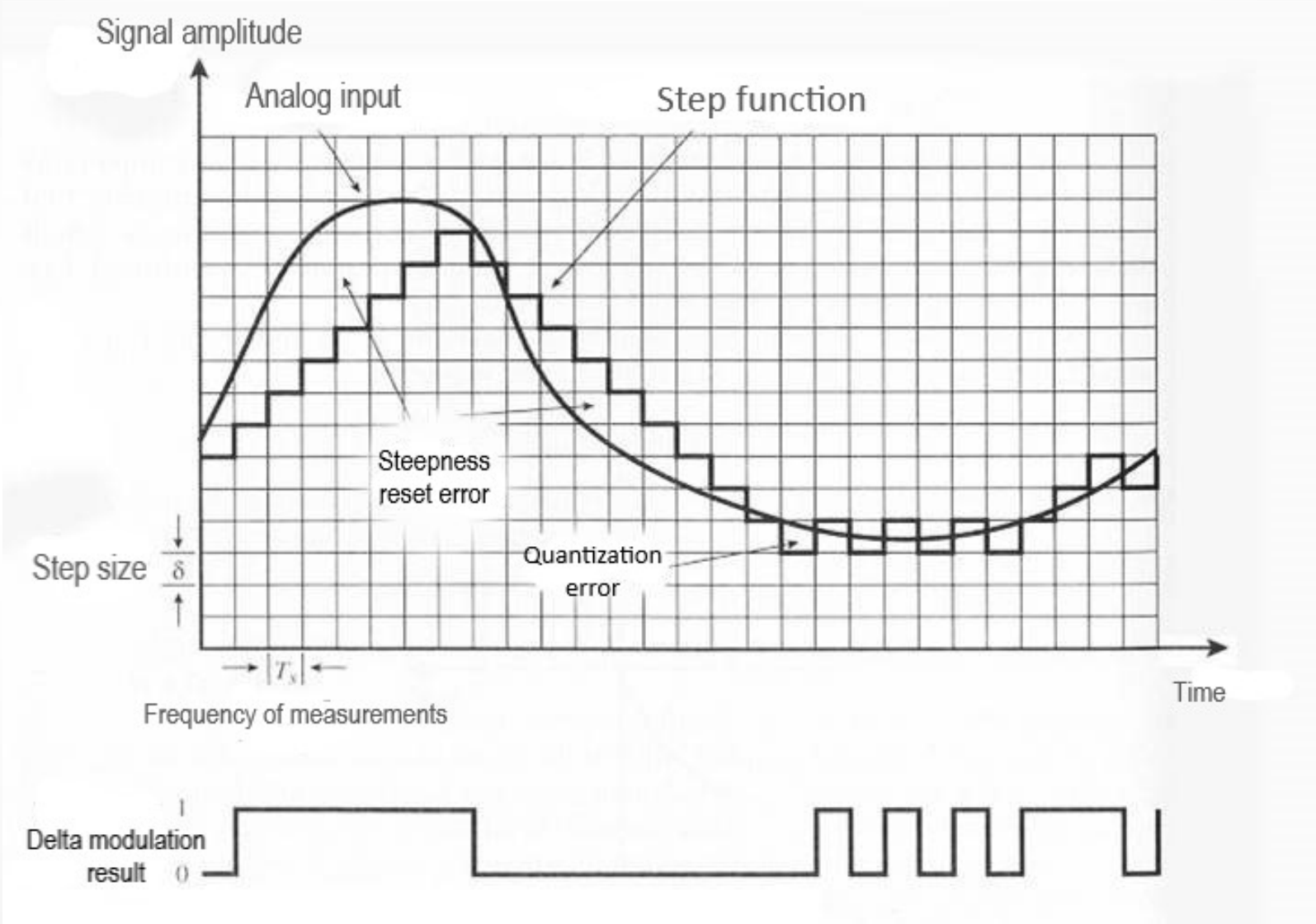


# Puls Code Modulation (PCM)



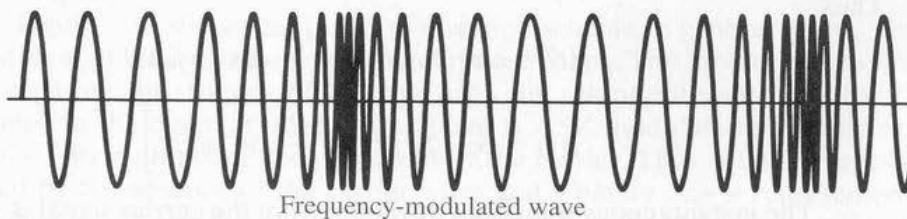
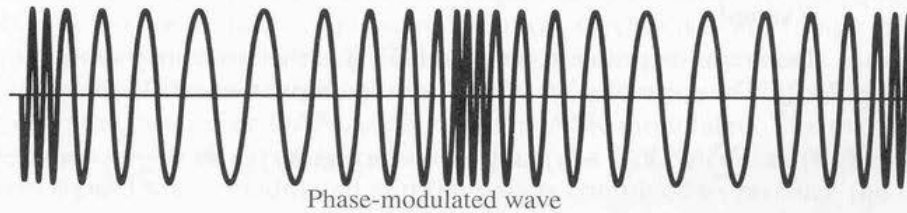
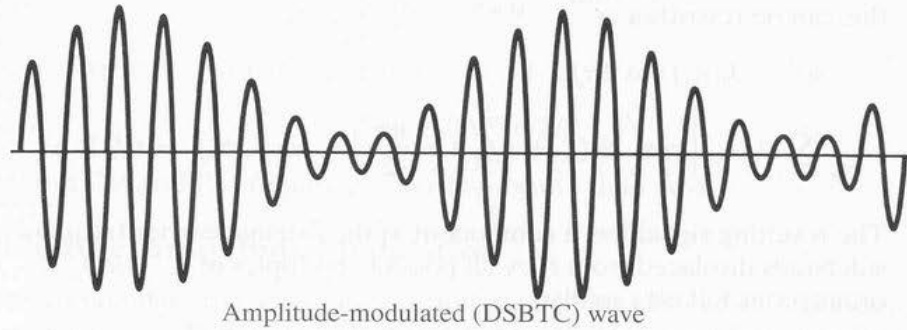
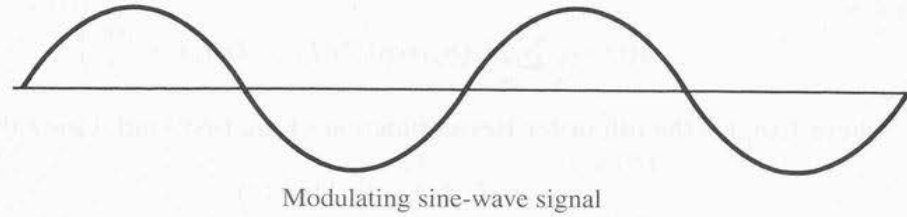
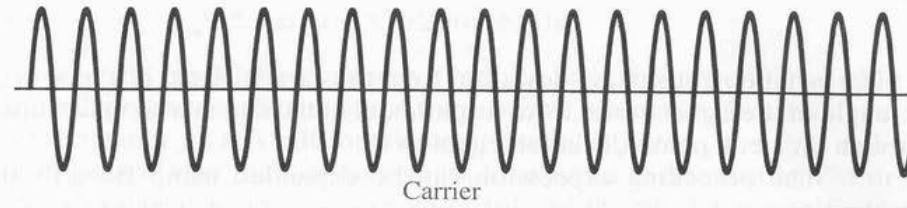


# Delta modulation





# Analog data – Analog Signal





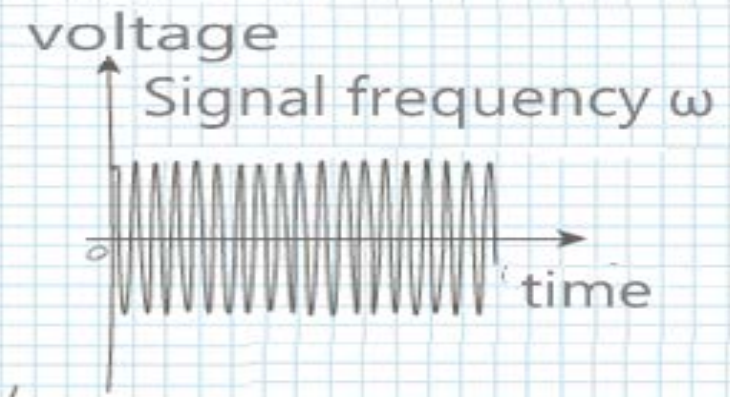
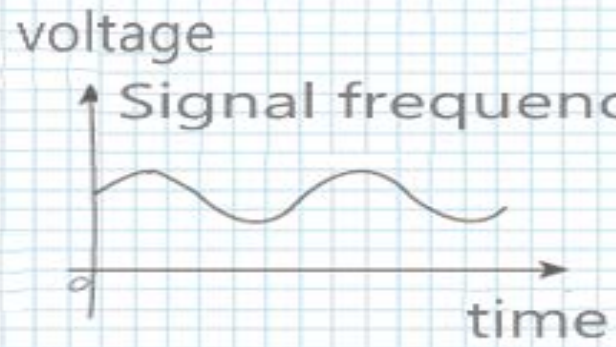


# Modulation

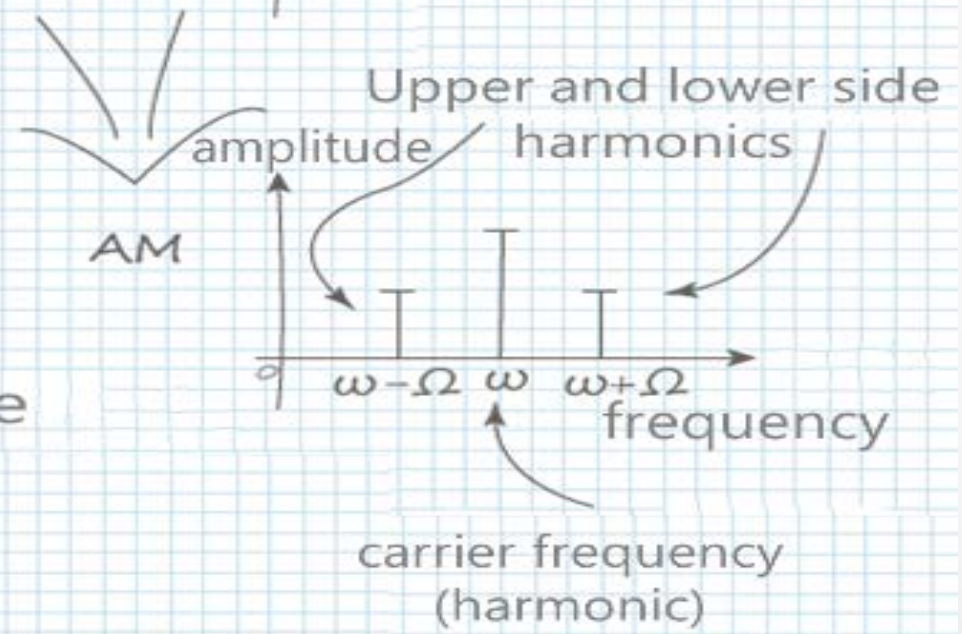
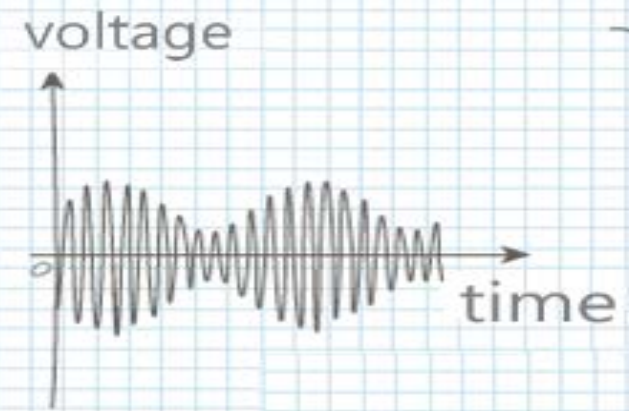


$$S(t) = A \cdot \sin(\omega t + \Psi)$$

amplitude      frequency      phase  
modulations



+





# Analog data – Analog Signal

- The Quadrature Amplitude Modulation (QAM) method is a combination of amplitude and phase modulation.
- The idea of this method is simultaneously send two different signals along the same line with the same carrier frequency, but  $90^\circ$  shifted in phase relative to each other. Each signal is generated by amplitude modulation.
- It is used in ADSL technology.

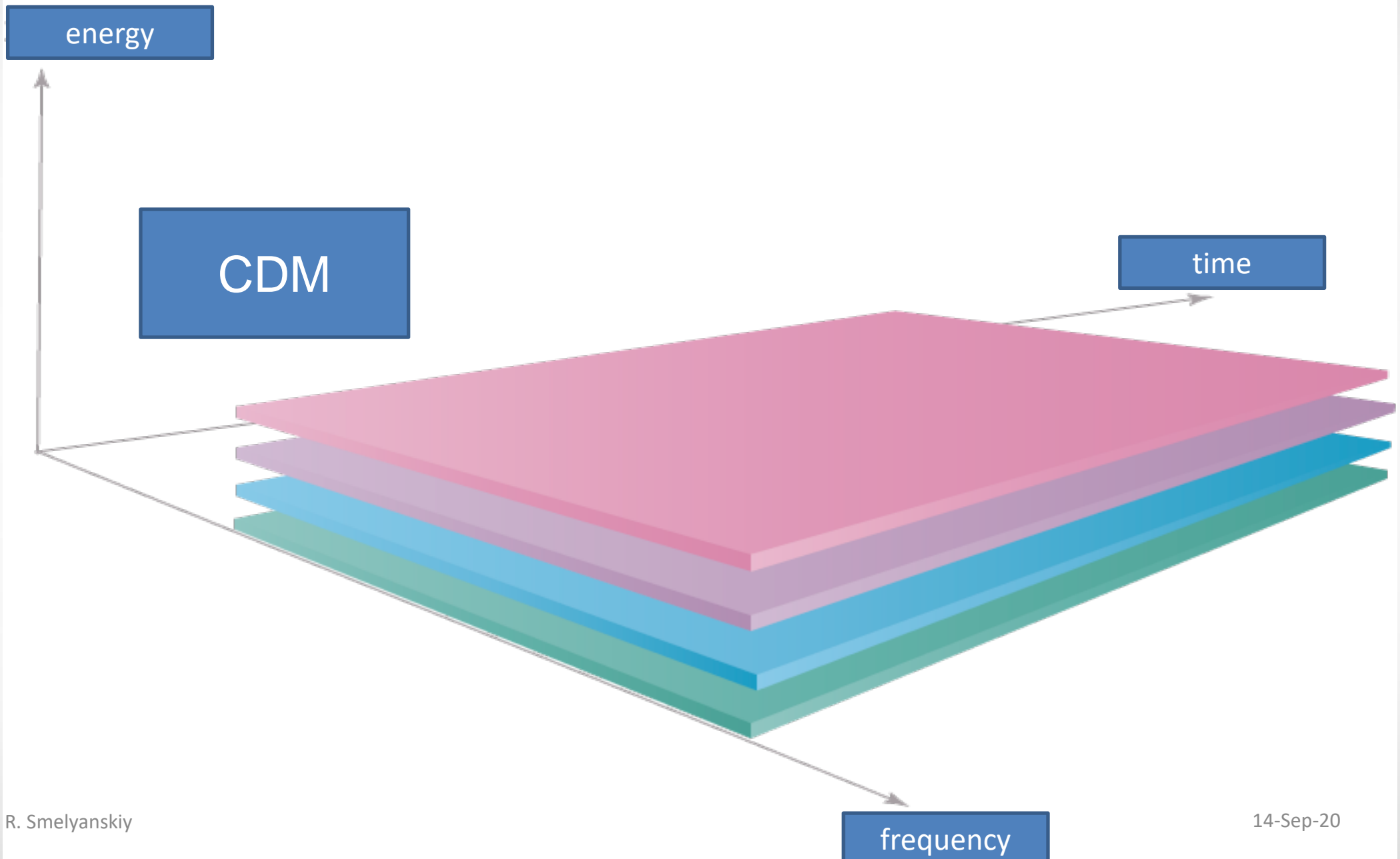


# Multiplexing

- Multiplexing - transmission of several data streams over the same physical channel. Each stream receives only a fraction of the total channel capacity. The transmission speed goes down.
  - Frequency (FDM)
  - Spectral (WDM)
  - Timing (TDM)
  - Plesiochronous Digital Hierarchy (PDH)
  - Code Division (CDMA)
  - Orthogonal Frequency (OFDM)
  - Spatial (MIMO)
  - Synchronous Digital Hierarchy (SDH)



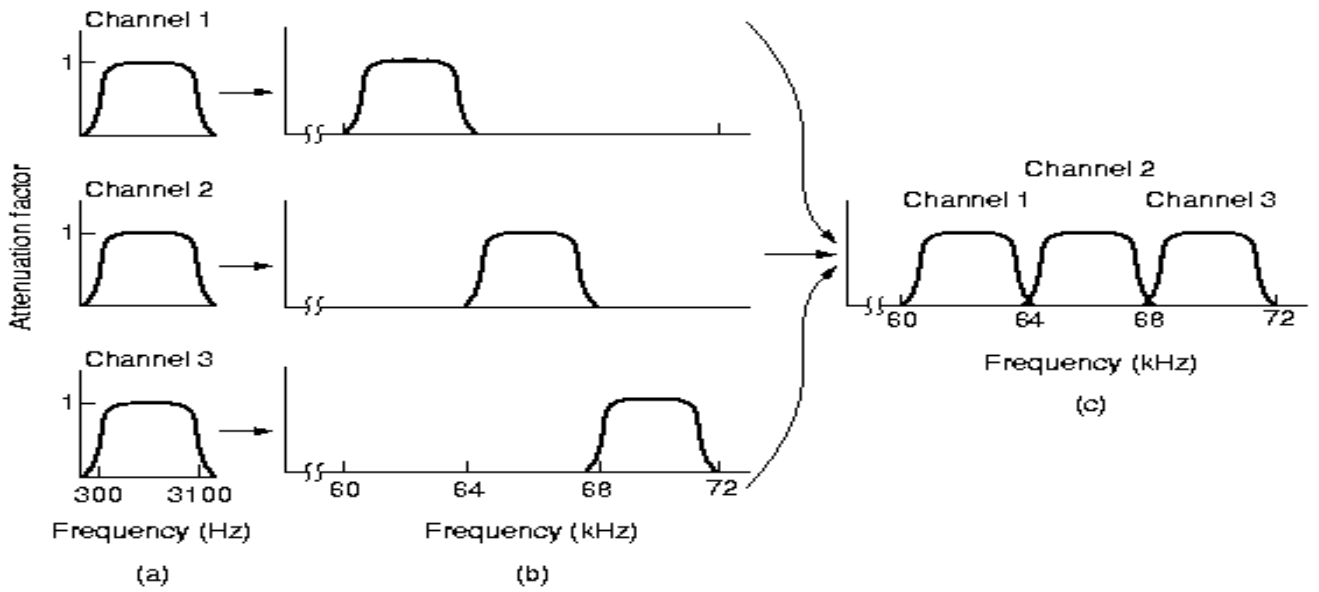
# Code Division Multiplexing (CDM)







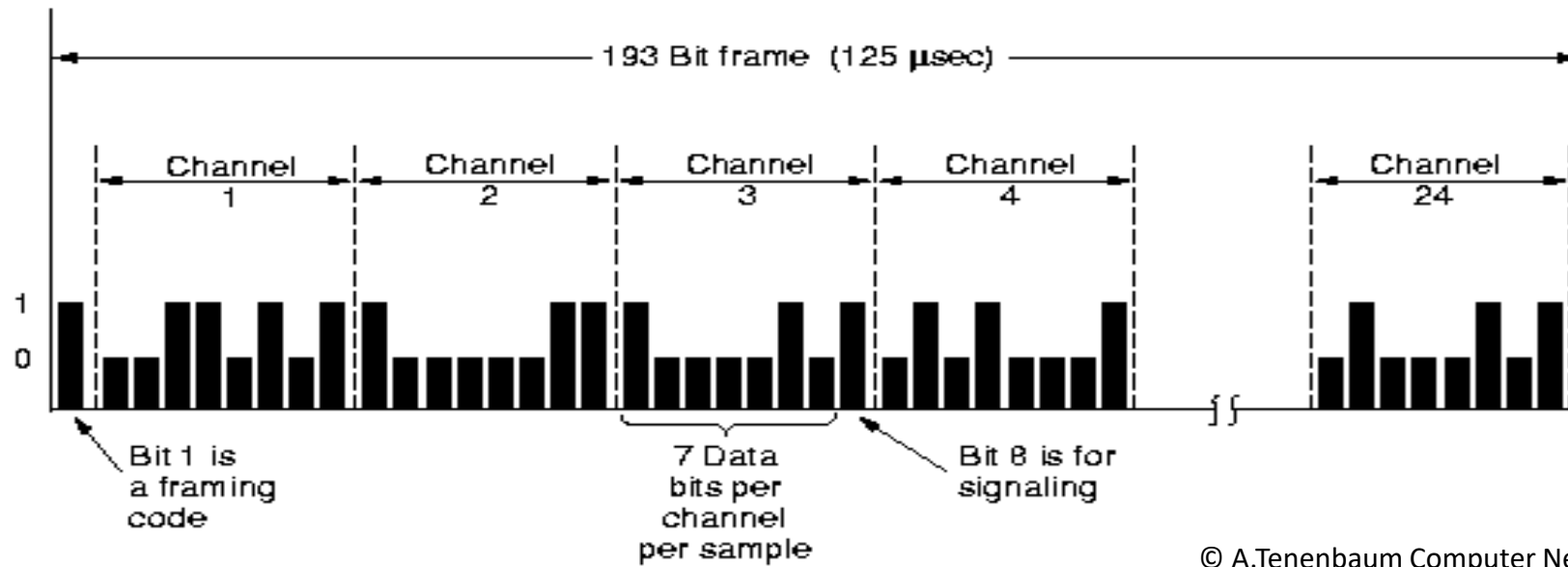
# Frequency Division Multiplexing (FDM)



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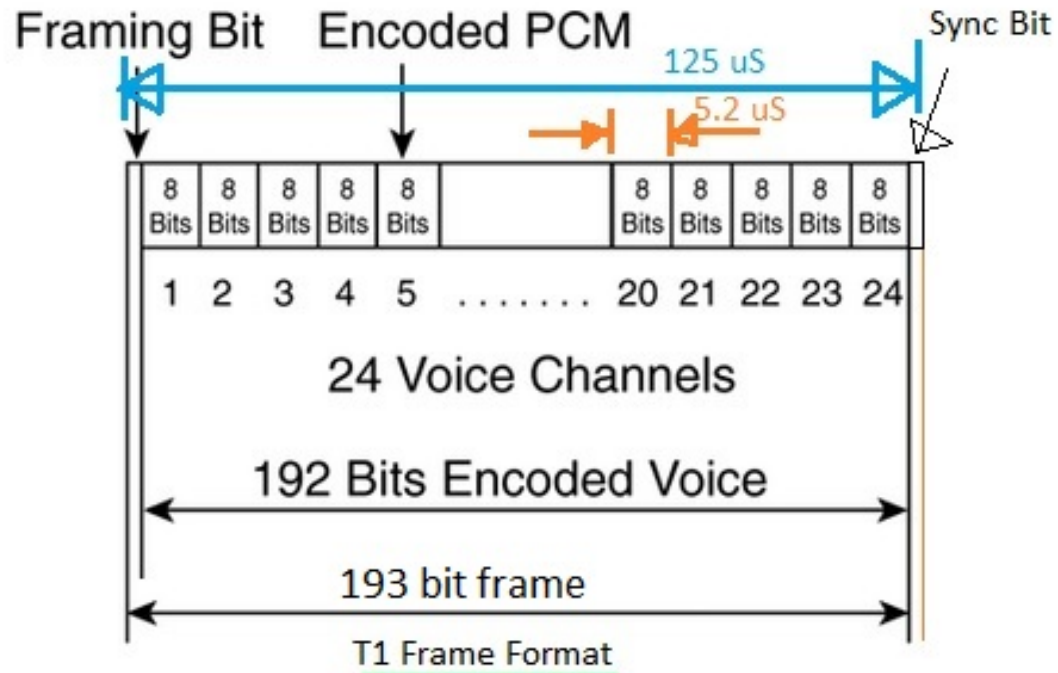
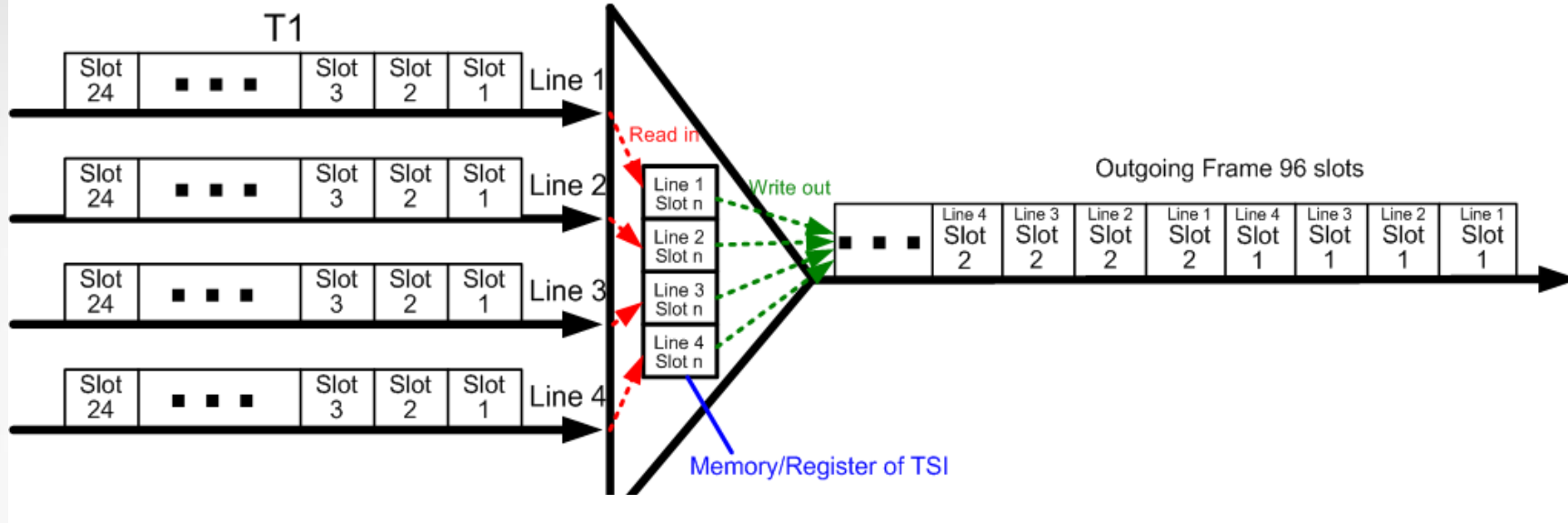
# Time Division Multiplexing (TDM)



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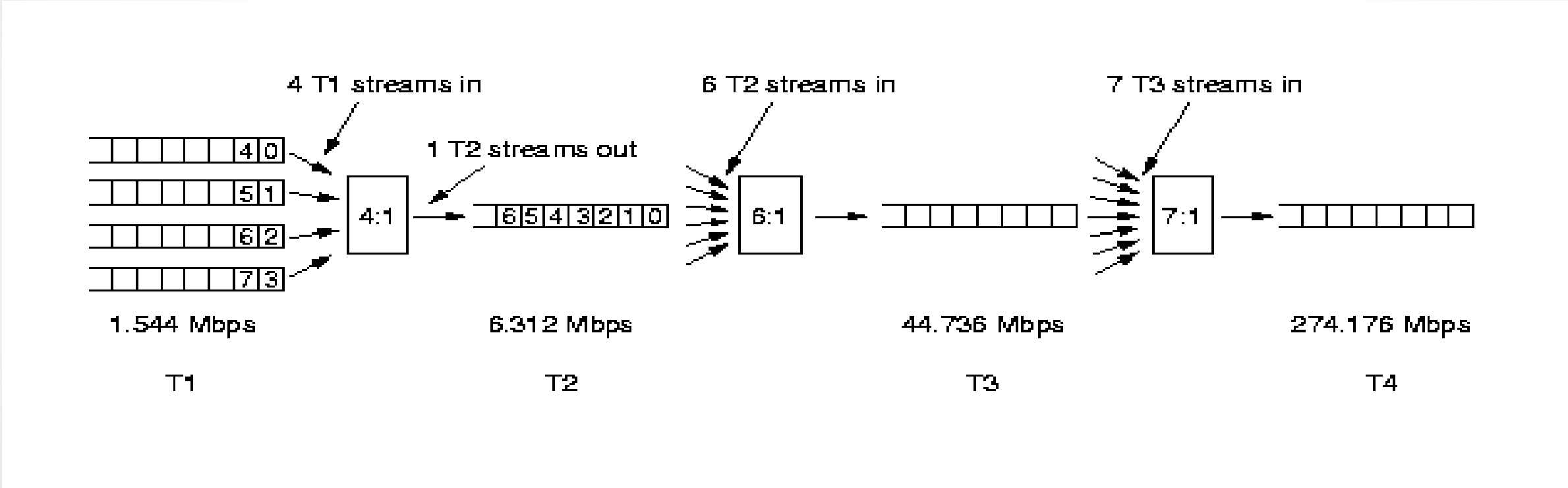


# Time Division Multiplexing (NDM)





# Plesiochronous multiplexing according to T standard series

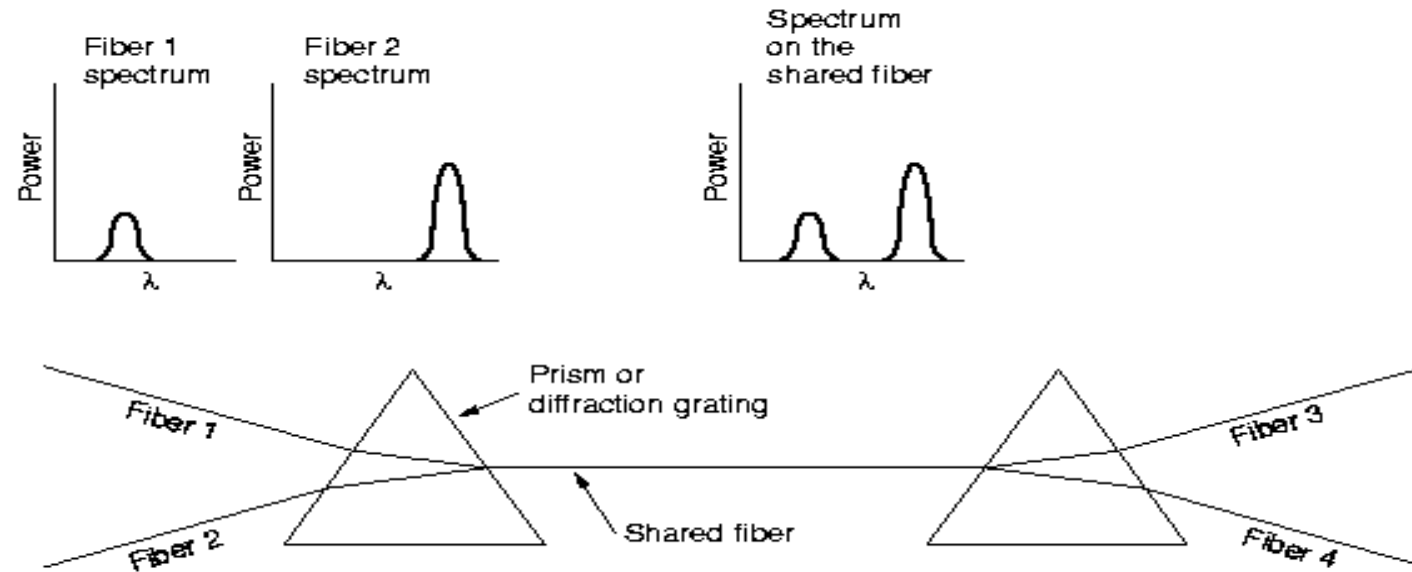


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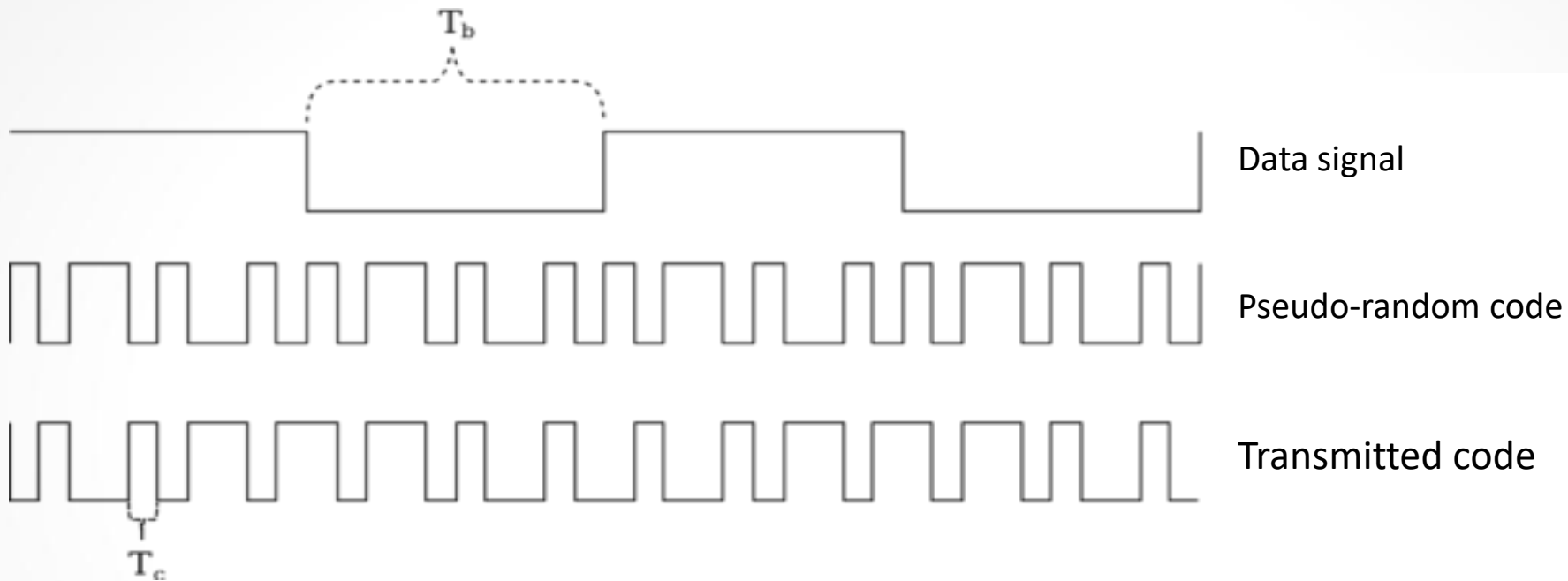
# Wavelength Division Multiplexing (WDM)



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# CDMA – code division



- Flexible allocation of resources. With code division, there is no strict limit on the number of channels. With the increase in the number of subscribers, the probability of decoding errors gradually increases, which leads to a decrease in the quality of the channel, but not to denial of service.
- Higher channel security. It is very difficult to select the desired channel without knowing its code. The entire frequency band is uniformly filled with a noise-like signal.
- CDMA phones have a lower peak radiation power and therefore allow more economical use of the battery.



# CDMA: Spread Spectrum Method

*one*

0	0	1	1	0	0	1	1	0	0	1
1	1	0	0	1	1	0	0	1	0	0

*zero*

*Original chip sequence*

0	0	1	1	0	0	1	1	0	0	1
---	---	---	---	---	---	---	---	---	---	---

*Resulting chip sequence*

0	1	1	1	1	0	1	1	0	0	1
---	---	---	---	---	---	---	---	---	---	---





# Walsh Functions



- Walsh functions are a family of functions that form an system of orthogonal functions, taking values only +1 and -1 in the entire domain.

$$\text{wal}(x) = \text{sign}(\sin(2^n \pi x))$$

- A group of Walsh functions forms a Hadamard matrix.
- Hadamard matrix can be formed recursively by using building block matrices by the following general formula:

$$H_{2^n} = \begin{bmatrix} H_{2^{n-1}} & H_{2^{n-1}} \\ H_{2^{n-1}} & -H_{2^{n-1}} \end{bmatrix}, \quad H_1 = [1], \quad H_2 = \begin{bmatrix} 1 & 1 \\ 1 & -1 \end{bmatrix}, \quad H_4 = \begin{bmatrix} 1 & 1 & 1 & 1 \\ 1 & -1 & 1 & -1 \\ 1 & 1 & -1 & -1 \\ 1 & -1 & -1 & 1 \end{bmatrix},$$

- The system of Walsh functions is an orthonormal basis and, as a result, allows us to decompose signals of arbitrary shape into a generalized Fourier series.
- Walsh function is defined on the interval [0, T]; outside this interval, the function repeats periodically.
- Walsh function numbered k is denoted as wal(k, θ), where θ = t / T is the dimensionless time.
- Regarding the moment θ, the Walsh functions can be divided into even and odd. They are denoted by cal(k, θ) = wal(2k, θ) and sal(k, θ) = wal(2k-1, θ) respectively.
- These functions are similar to trigonometric sines and cosines.



# Walsh Functions

The main properties of the Walsh system.

- 1) Walsh functions are orthogonal and normalized
  - 2) The average value of the Walsh functions for all is zero
  - 3) The product of two Walsh functions is equal to the new Walsh function from the same system
  - 4) Even with respect to the middle of the interval (0.5), functions correspond to
- Walsh Functions - A Combination of Rectangular Pulses Easily Implemented by Digital Devices
  - Signal decomposition into Walsh functions

$$s(t) = \sum_{k=0}^{\infty} C_k \varphi_k(t) = C_0 \text{wal}_0\left(\frac{t}{T_H}\right) + C_1 \text{wal}_1\left(\frac{t}{T_H}\right) + C_2 \text{wal}_2\left(\frac{t}{T_H}\right) + \dots$$



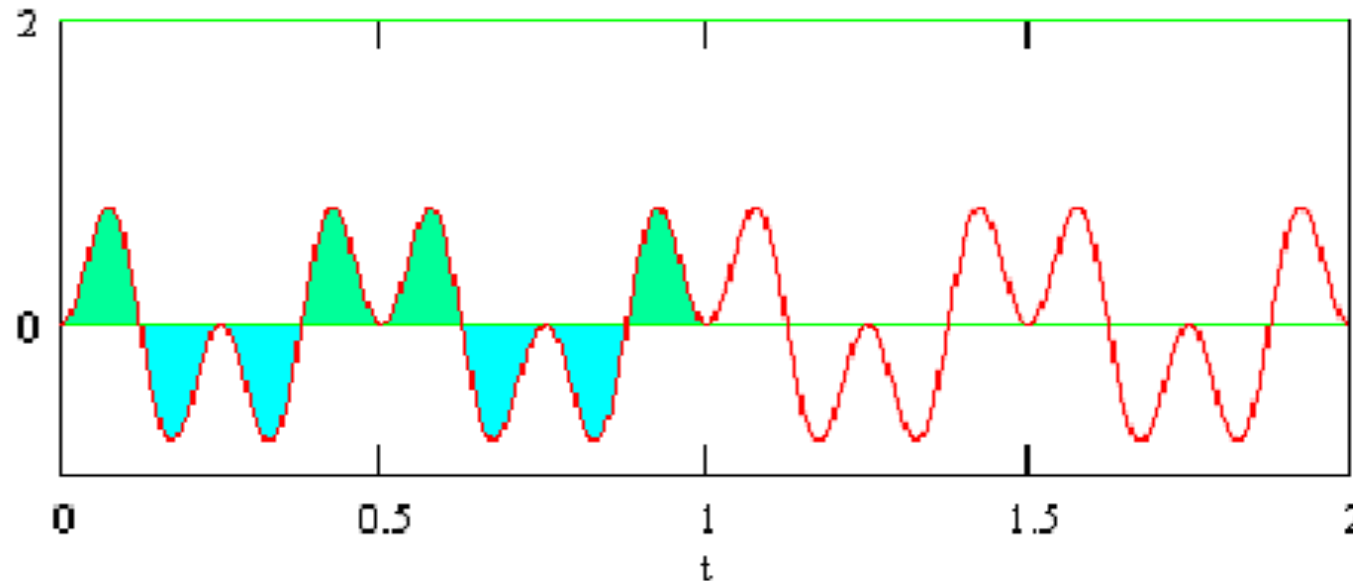


# Orthogonal Frequency Division Multiplexing



# Orthogonal Subcarriers

$$f(t) = \sin wt * \sin nwt$$



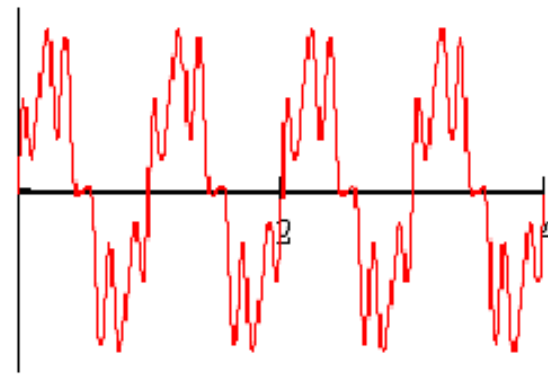
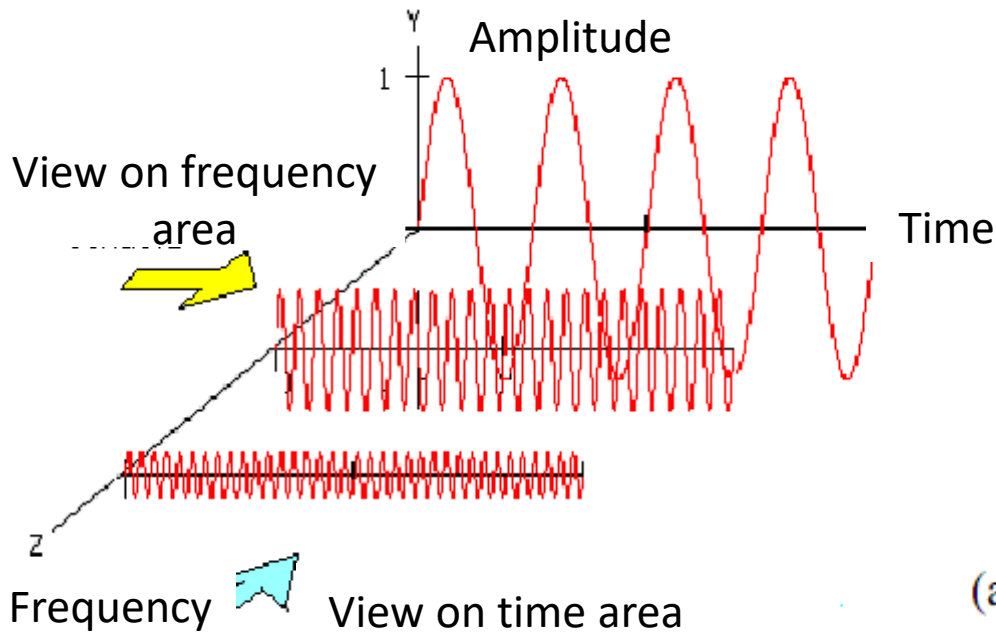
The product of sinusoids of different harmonics

$$= \frac{1}{2} \cos(m - n) - \frac{1}{2} \cos(m + n)$$

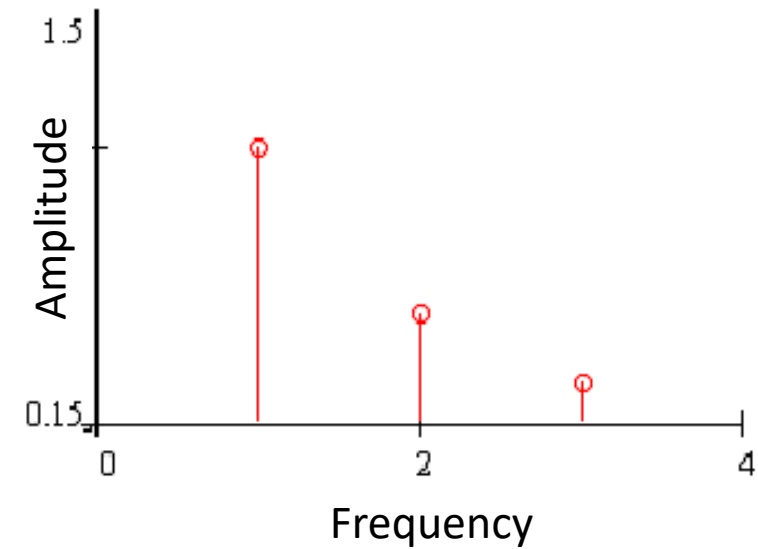
$$\int_0^{2\pi} \frac{1}{2} \cos(m - n)\omega t - \int_0^{2\pi} \frac{1}{2} \cos(m + n)\omega t = 0$$



# Relationships OFDM with DFT/IFT



(a) View in time area  
- DFT



(b) View in frequency area  
- IFT

$$X(k) = \sum_{n=0}^{N-1} e^{-i\frac{2\pi}{T}kn} x(n) = \sum_{n=0}^{N-1} \left[ \cos\left(\frac{2\pi}{T}kn\right) - i \sin\left(\frac{2\pi}{T}kn\right) \right] x(n)$$



# OFDM Multiplexing

## Advantages

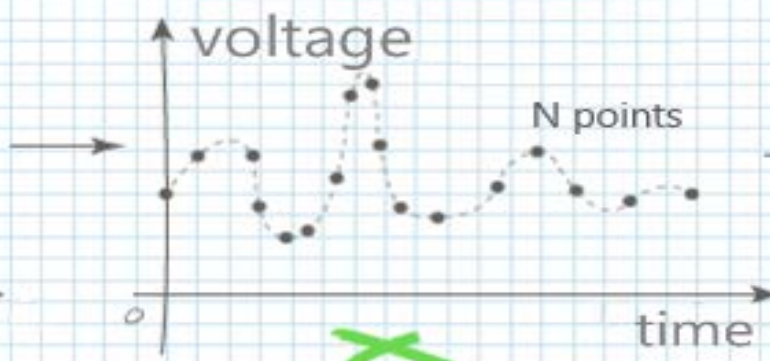
- High efficiency of the radio frequency spectrum usage, due to the almost rectangular shape of the envelope of the spectrum with a large number of subcarriers.
- Simple hardware implementation: basic operations are implemented using digital processing methods.
- Good opposition to intersymbol interference and interference between subcarriers. As a result - good adaptability to multipath propagation.
- The possibility of using different modulation schemes for each subcarrier, which allows adaptively varying noise immunity and information transfer rate.

## Disadvantages

- High synchronization of frequency and time is required.
- Sensitivity to the Doppler effect, limiting the use of OFDM in mobile systems.
- The high level of phase noise of modern receivers and transmitters limits system performance.
- The guard interval used in OFDM to combat multipath propagation reduces the spectral efficiency of the signal.

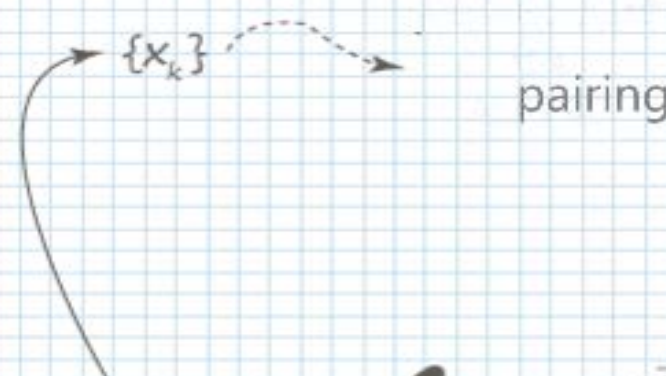


# Fast Fourier Transformation (FFT)

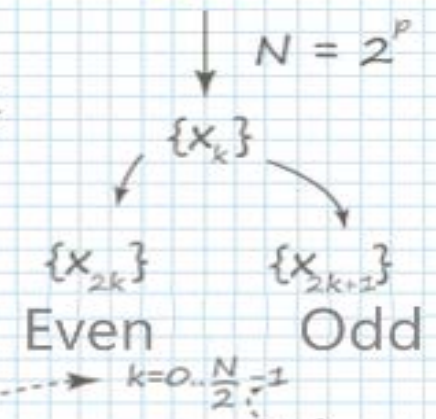


$\{x_k\}$  - Signal values  
 Discrete Fourier Transformation  
 $\{c_n\}$  - Spectrum value

$$c_n = \frac{1}{N} \sum_{k=0}^{+\infty} x_k e^{-\frac{j2\pi nk}{N}}$$



pairing → FFT (N=2) → Union →  $\{c_n\}$



For all values

$$c_n = \begin{cases} c_n + c_n e^{-\frac{j2\pi n}{N}} & (n \leq N/2-1) \\ c_{n-N/2} - c_{n-N/2} e^{-\frac{j2\pi(n-N/2)}{N}} & (n \geq N/2) \end{cases}$$

#6

↑ Even    ↑ Odd

For half the values

$$c_n = c_n + c_n e^{-\frac{j2\pi n}{N}}$$

#5

↑ Even    ↑ Odd

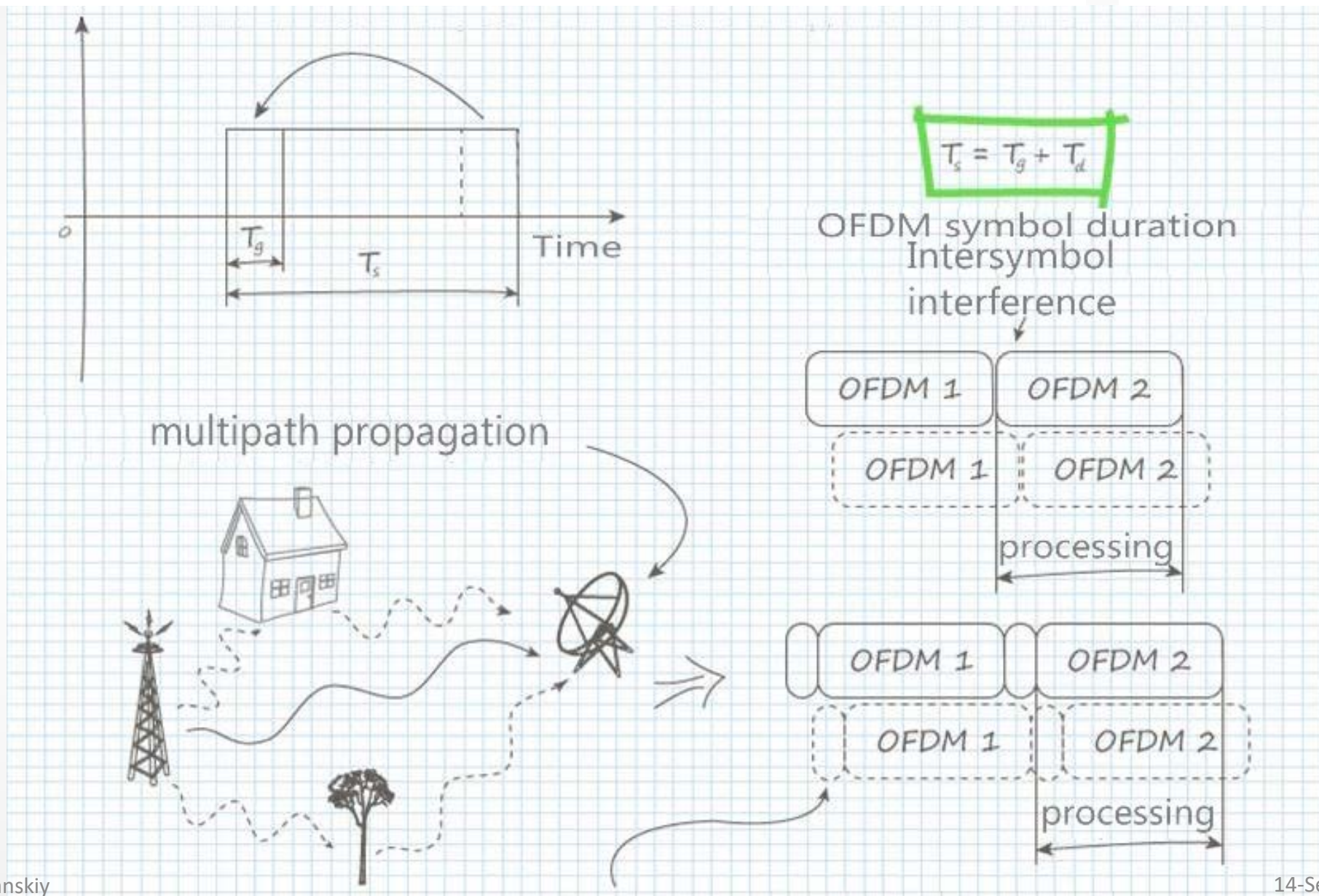
because count from

#4



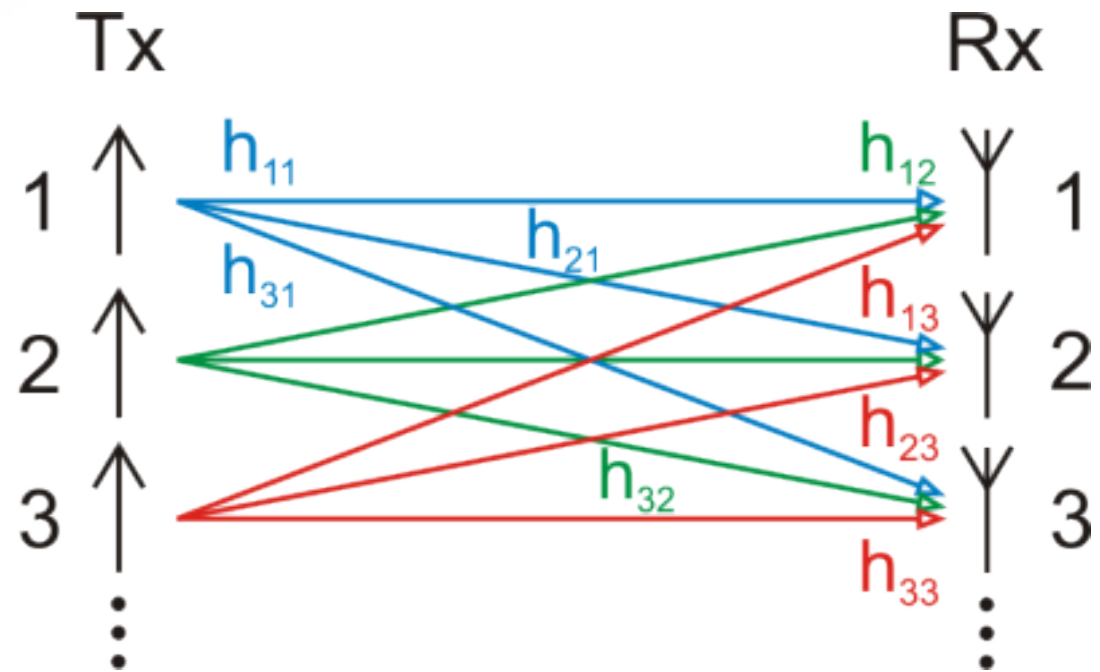


# The guard interval





# MIMO Spatial Multiplexing





# Conclusion

- Mobility growth
- Growth as traffic amount as traffic QoS requirements (CDN)
- Change Organization and Computation Model
- Virtualization and Convergence of resources
- Data Center Hierarchy (Edge computing)
- Network structure telecom operator
- Channel throughput growth:
  - Physical environments
  - Data Encoding Methods
  - Digital transmission
  - Multiplexing



# Thank You

## Questions?