

# Fundamentals of Digital Communications and Data Transmission

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# Overview

- Introduction
  - Communication systems
  - Digital communication system
  - Importance of Digital transmission
- Basic Concepts in Signals
  - Sampling
  - Quantization
  - Coding

# What is Communication?

- Communication is transferring data **reliably** from one point to another
  - Data could be: voice, video, codes etc...
- It is important to receive the same information that was sent from the transmitter.
- Communication system
  - A system that allows transfer of information reliably

# Communication Systems



Information  
Source



Transmitt  
er



Channel



Receiver



Information  
Sink

Block Diagram of a typical communication system

- Information Source

- The source of data

- Data could be: human voice, data storage device CD, video etc..

- Data types:

- Discrete: Finite set of outcomes “Digital”
    - Continuous : Infinite set of outcomes “Analog”

- Transmitter

- Converts the source data into a suitable form for transmission through signal processing

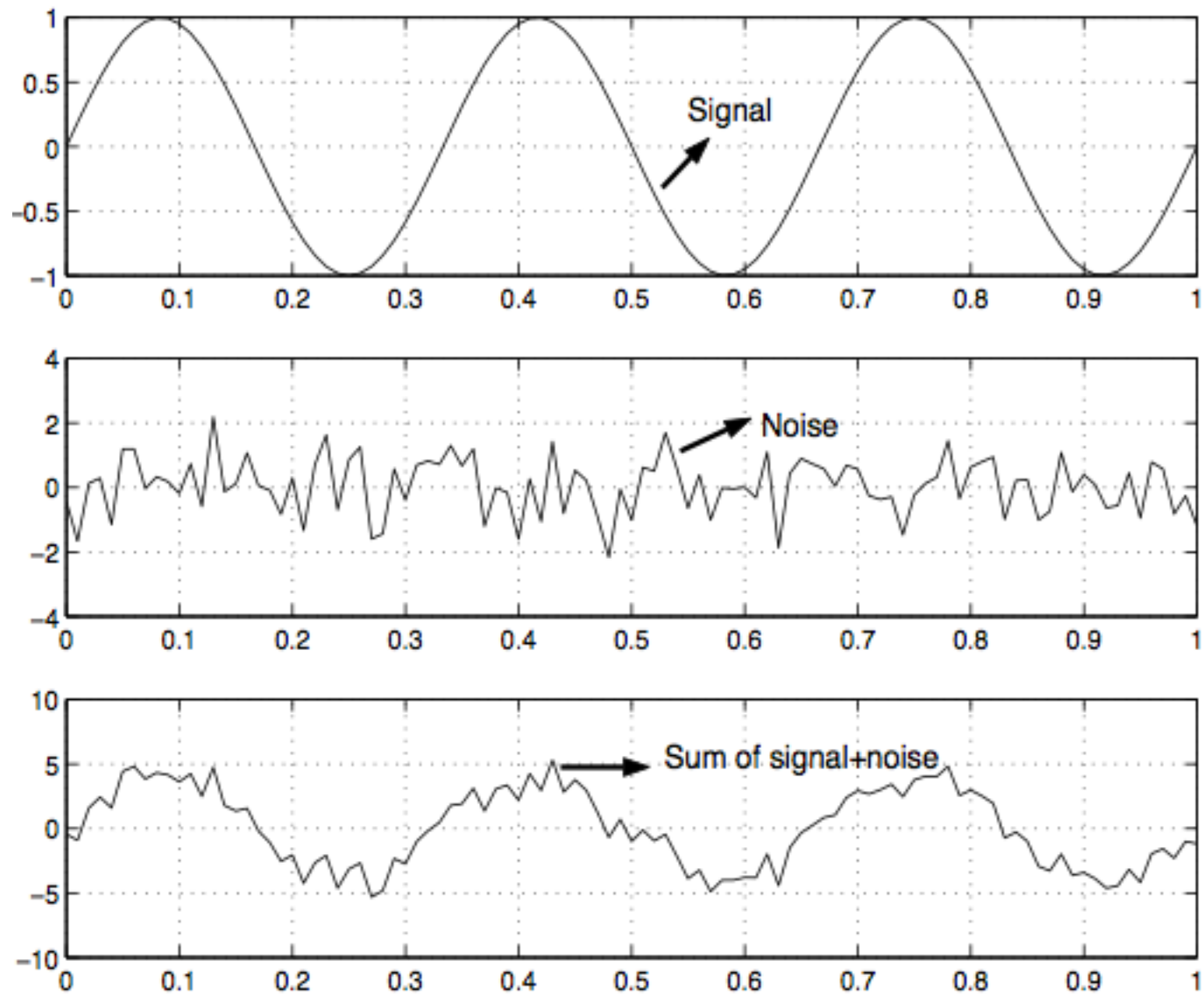
- Data form depends on the channel

- Channel:
  - The physical medium used to send the signal
  - The medium where the signal propagates till arriving to the receiver
  - Physical Mediums (Channels):
    - Wired : twisted pairs, coaxial cable, fiber optics
    - Wireless: Air, vacuum and water
  - Each physical channel has a certain limited range of frequencies ,( $f_{min} \rightarrow f_{max}$ ), that is called the channel bandwidth
  - Physical channels have another important limitation which is the **NOISE**

- Channel:
  - Noise is undesired random signal that corrupts the original signal and degrades it
  - Noise sources:
    - » Electronic equipments in the communication system
    - » Thermal noise
    - » Atmospheric electromagnetic noise (Interference with another signals that are being transmitted at the same channel)
- Another Limitation of noise is the attenuation
  - Weakens the signal strength as it travels over the transmission medium
  - Attenuation increases as frequency increases
- One Last important limitation is the delay distortion
  - Mainly in the wired transmission
  - Delays the transmitted signals → Violates the reliability of the communication system



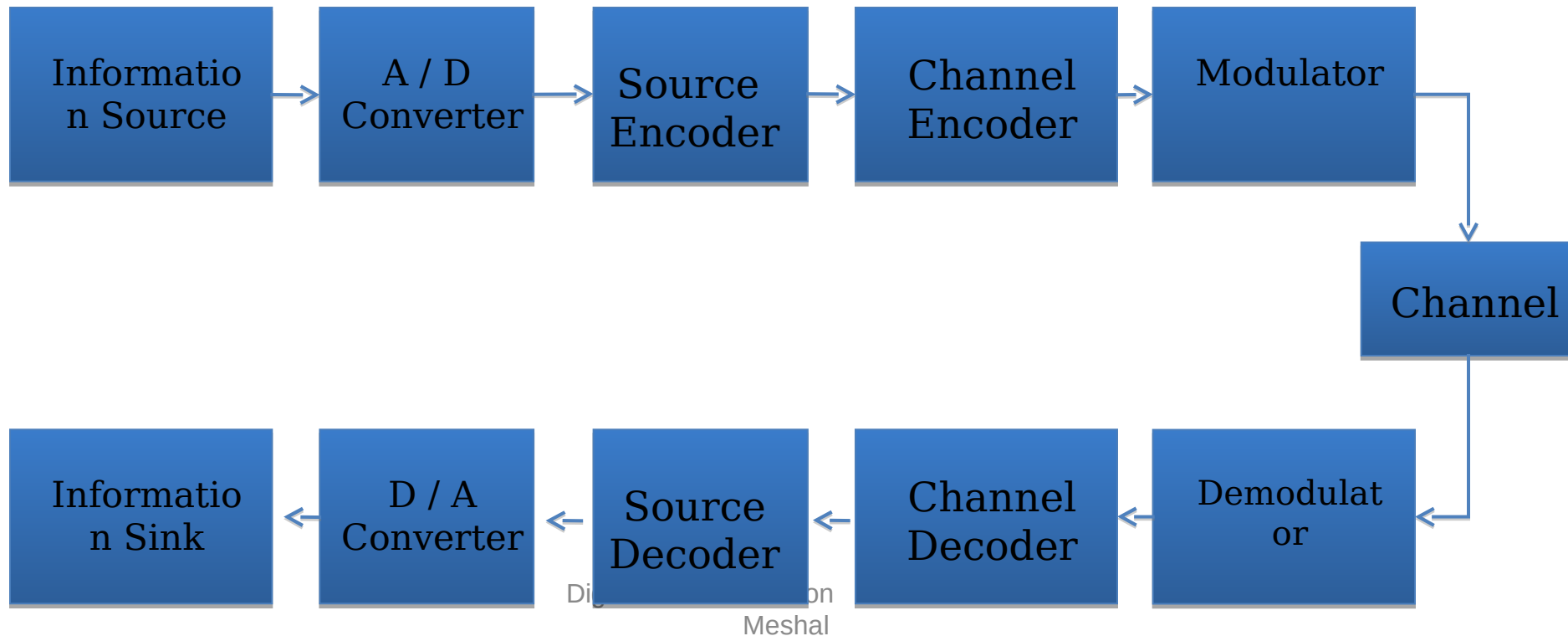
- Receiver
  - Extracting the message/code in the received signal
    - Example
      - Speech signal at transmitter is converted into electromagnetic waves to travel over the channel
      - Once the electromagnetic waves are received properly, the receiver converts it back to a speech form
  - Information Sink
    - The final stage
    - The user



Digital Communication Abdullah Al-  
Effect of Noise On a transmitted signal  
mesnani

# Digital Communication System

- Data of a digital format “i.e binary numbers”



- Information source
  - Analog Data: Microphone, speech signal, image, video etc...
  - Discrete (Digital) Data: keyboard, binary numbers, hex numbers, etc...
- Analog to Digital Converter (A/D)
  - Sampling:
    - Converting continuous time signal to a digital signal
  - Quantization:
    - Converting the amplitude of the analog signal to a digital value
  - Coding:
    - Assigning a binary code to each finite amplitude in the analog signal

- Source encoder
  - Represent the transmitted data more efficiently and remove redundant information
    - How? “write Vs. rite”
    - Speech signals frequency and human ear “20 kHz”
  - Two types of encoding:
  - Lossless data compression (encoding)
    - Data can be recovered without any missing information
  - Lossy data compression (encoding)
    - Smaller size of data
    - Data removed in encoding can not be recovered again

- Channel encoder:
  - To control the noise and to detect and correct the errors that can occur in the transmitted data due the noise.
- Modulator:
  - Represent the data in a form to make it compatible with the channel
    - Carrier signal “high frequency signal”
- Demodulator:
  - Removes the carrier signal and reverse the process of the Modulator

- Channel decoder:
  - Detects and corrects the errors in the signal gained from the channel
- Source decoder:
  - Decompresses the data into it's original format.
- Digital to Analog Converter:
  - Reverses the operation of the A/D
  - Needs techniques and knowledge about sampling, quantization, and coding methods.
- Information Sink
  - The User

# Why should we use digital communication?

- Ease of regeneration
  - Pulses “ 0 , 1”
  - Easy to use repeaters
- Noise immunity
  - Better noise handling when using repeaters that repeats the original signal
  - Easy to differentiate between the values “either 0 or 1”
- Ease of Transmission
  - Less errors
  - Faster !
  - Better productivity



# Why should we use digital communication?

- Ease of multiplexing
  - Transmitting several signals simultaneously
- Use of modern technology
  - Less cost !
- Ease of encryption
  - Security and privacy guarantee
  - Handles most of the encryption techniques

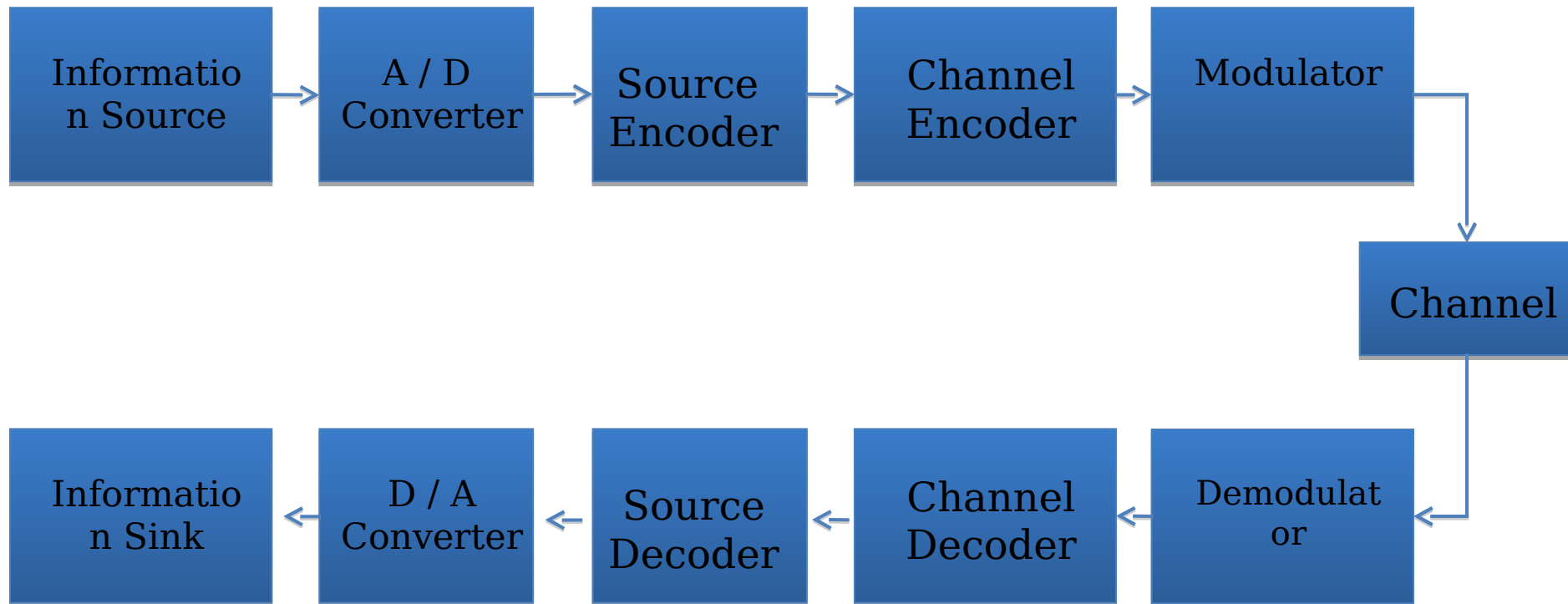
# Disadvantage !

- The major disadvantage of digital transmission is that it requires a greater transmission bandwidth or channel bandwidth to communicate the same information in digital format as compared to analog format.
- Another disadvantage of digital transmission is that digital detection requires system synchronization, whereas analog signals generally have no such requirement.

# Chapter 2: Analog to Digital Conversion (A/D)

Abdullah Al-Meshal

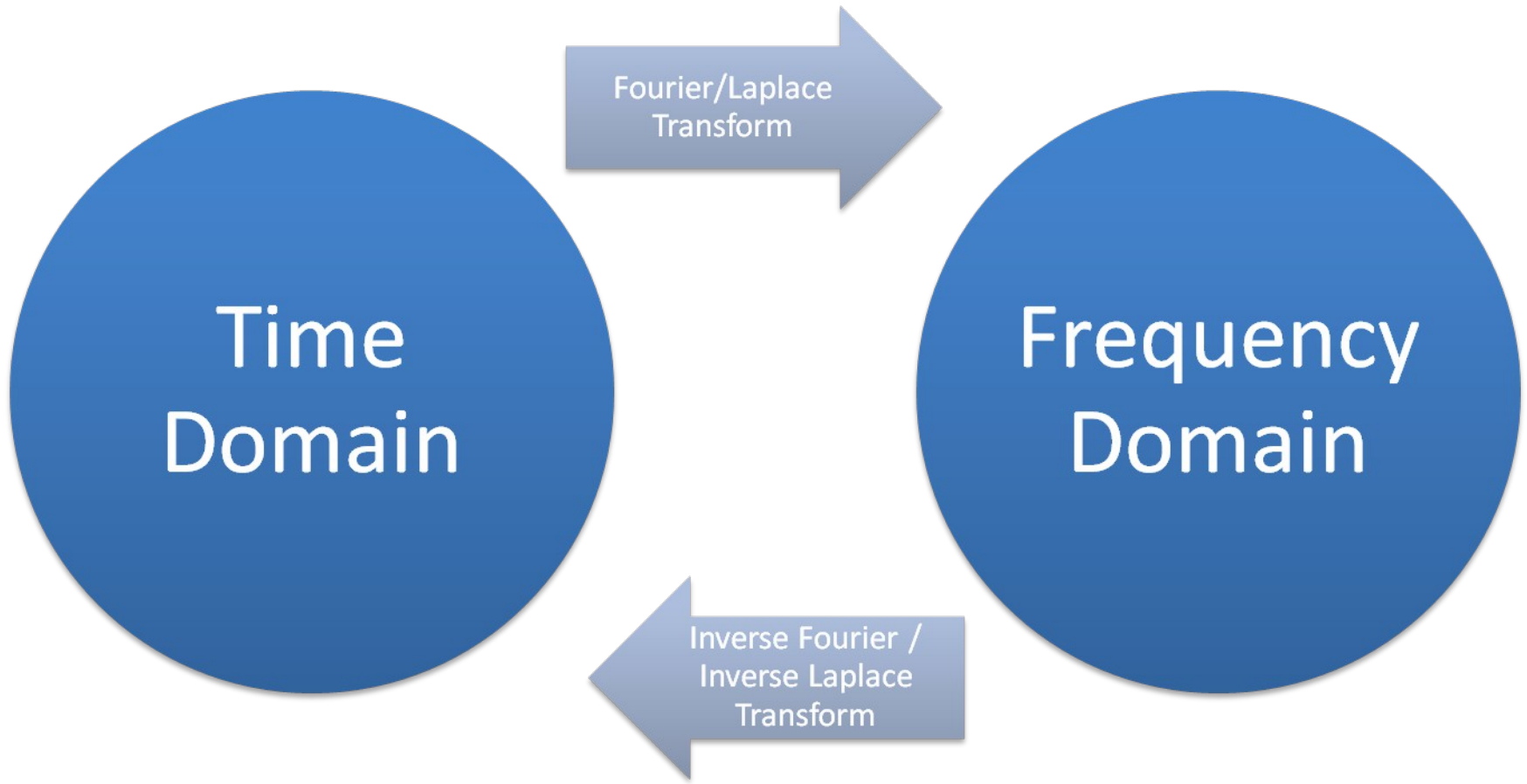
# Digital Communication System



## 2.1 Basic Concepts in Signals

- A/D is the process of converting an analog signal to digital signal, in order to transmit it through a digital communication system.
- Electric Signals can be represented either in Time domain or frequency domain.
  - Time domain i.e  $v(t) = 2\sin(2\pi 1000t + 45)$
  - We can get the value of that signal at any time (t) by substituting in the  $v(t)$  equation.

# Time domain Vs. Frequency domain



# Time domain Vs. Frequency domain

- Consider taking two types of images of a person:
  - Passport image
  - X-Ray image
- Two different domains, spatial domain “passport image” and “X-Ray domain”.
- Doctors are taking the image in the X-Ray domain to extract more information about the patient.
- Different domains helps fetching and gaining knowledge about an object.
  - An Object : Electric signal, human body, etc...

# Time domain Vs Frequency domain

- We deal with communication system in the time domain.
  - Lack of information about the signal
  - Complex analysis
- Frequency domain gives us the ability to extract more information about the signal while simplifying the mathematical analysis.



# Frequency Domain

- To study the signal in the frequency domain, we need to transfer the original signal from the time domain into the frequency domain.
  - Using Fourier Transform

$$X(f) = \int_{-\infty}^{\infty} x(t) e^{-j2\pi ft} dt$$

Fourier Transform  
Time domain  $\rightarrow$  Frequency Domain

$$x(t) = \int_{-\infty}^{\infty} X(f) e^{j2\pi ft} df$$

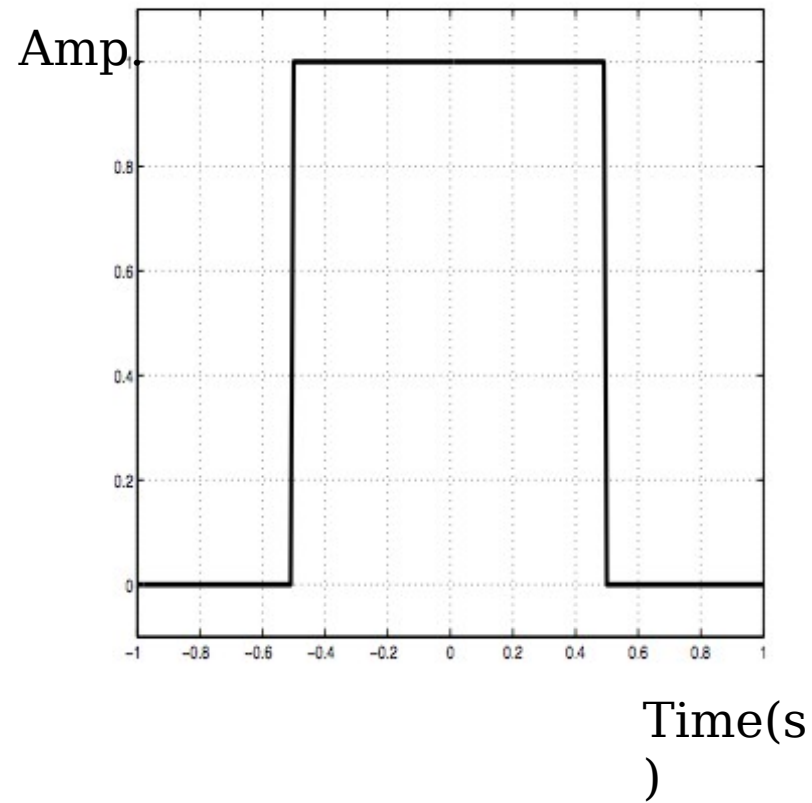
Inverse Fourier Transform  
Frequency domain  $\rightarrow$  Time Domain

# Spectrum

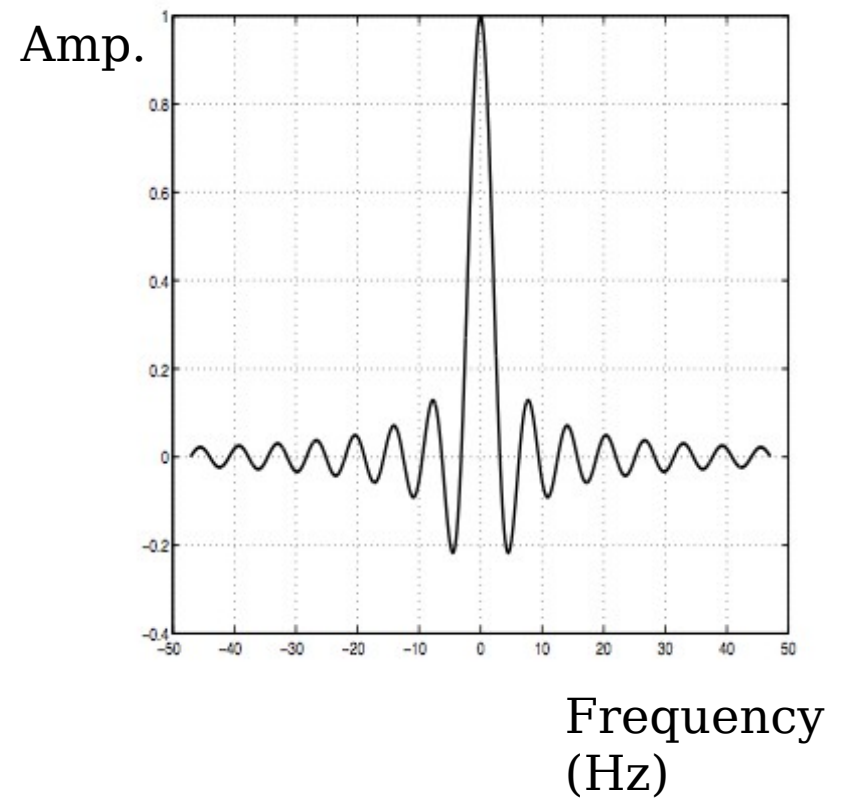
- The spectrum of a signal is a plot which shows how the signal amplitude or power is distributed as a function of frequency.

$$X(f) = \int_{-\infty}^{\infty} x(t) e^{-j2\pi ft} dt = \int_{-0.5}^{0.5} e^{-j2\pi ft} dt = \frac{1}{-j2\pi f} \left[ e^{-j0.5\pi ft} - e^{j0.5\pi ft} \right] = \frac{\sin(\pi f)}{\pi f}$$

## Time Domain

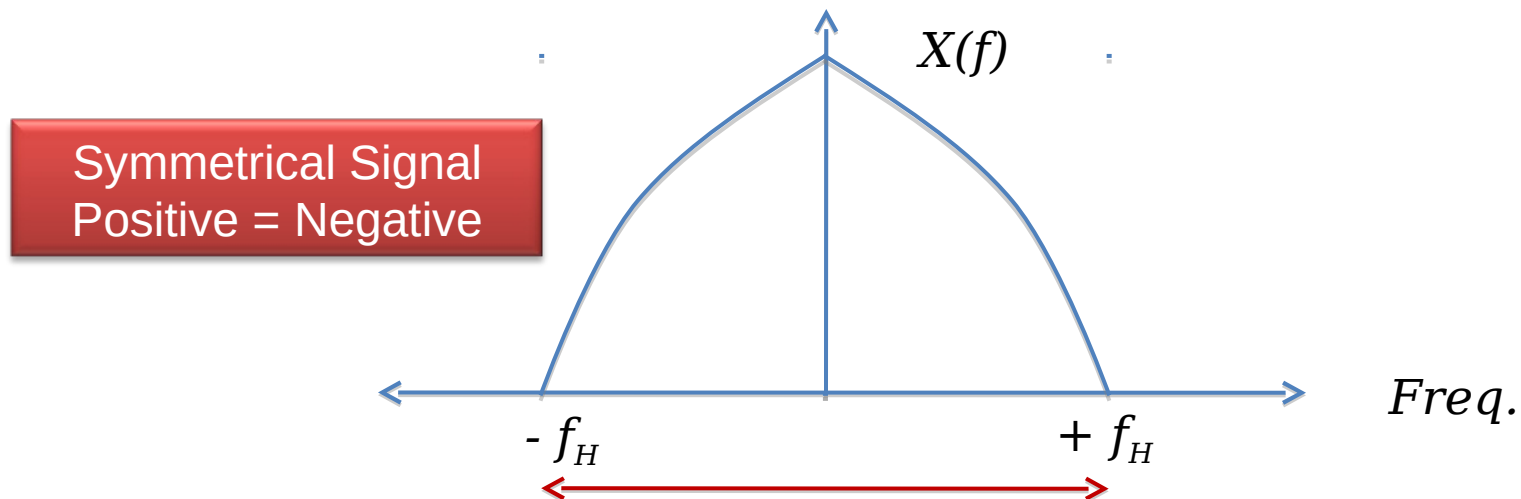


## Frequency Domain



# Band limited signals

- A band limited signal is a signal who has a finite spectrum.
- Most of signal energy in the spectrum is contained in a finite range of frequencies.
- After that range, the signal power is almost zero or negligible value.



# Converting an Analog Signal to a Discrete Signal (A/D)

- Can be done through three basic steps:

1- Sampling

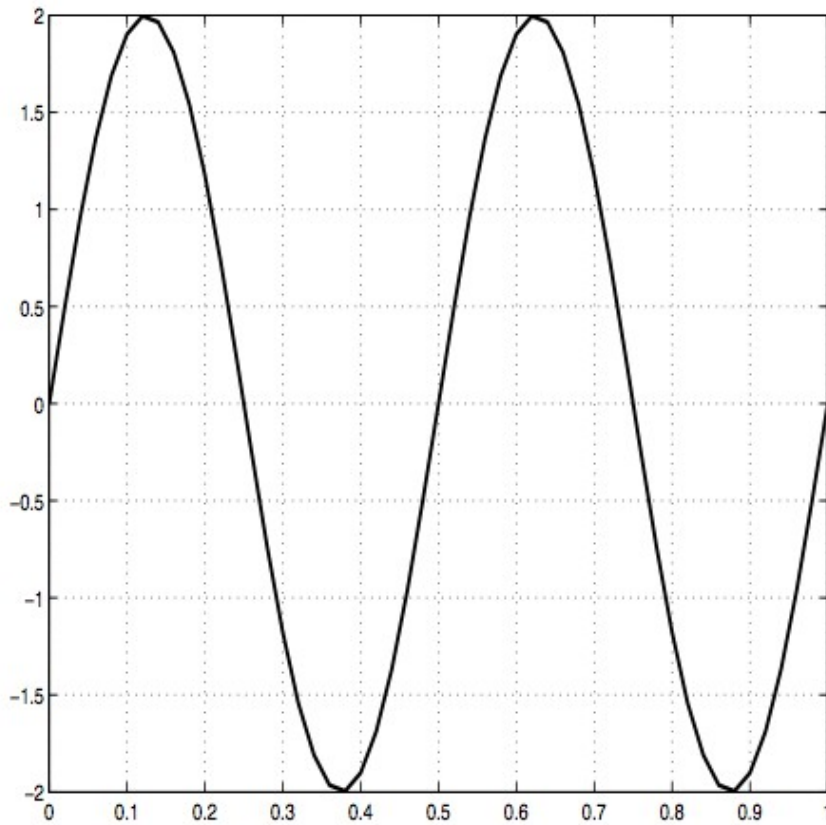
2- Quantization

3- Coding

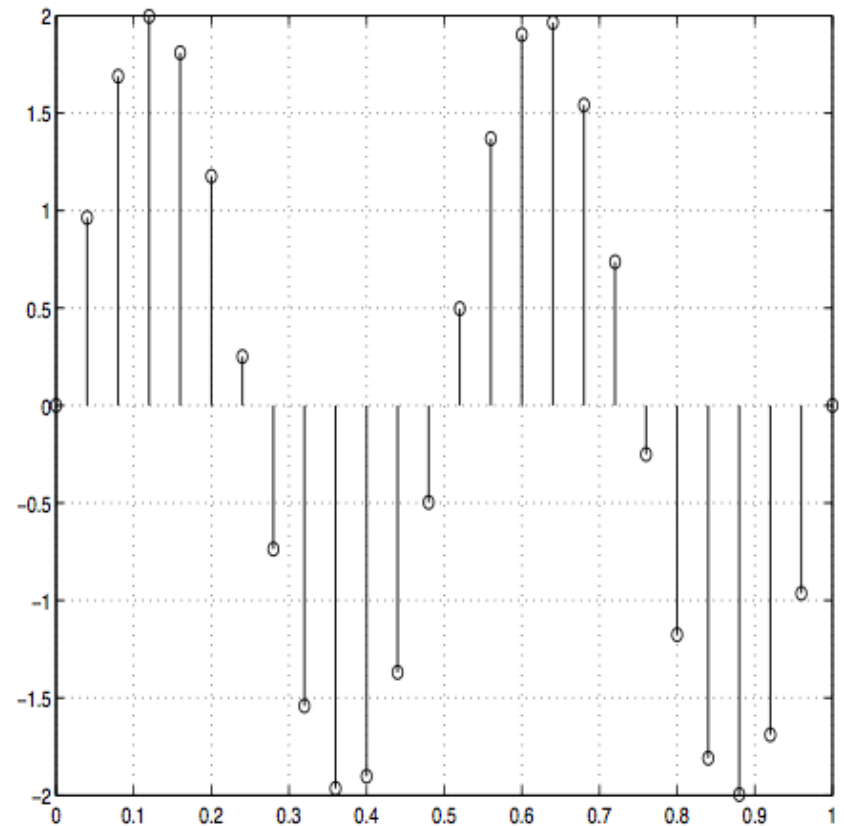
# Sampling

- Process of converting the continuous time signal to a discrete time signal.
- Sampling is done by taking “Samples” at specific times spaced regularly.
  - $V(t)$  is an analog signal
  - $V(nT_s)$  is the sampled signal
    - $T_s$  = positive real number that represent the spacing of the sampling time
    - $n$  = sample number integer

# Sampling



Original Analog Signal  
"Before Sampling"



Sampled Analog Signal  
"After Sampling"

# Sampling

- The closer the  $T_s$  value, the closer the sampled signal resemble the original signal.
- Note that we have lost some values of the original signal, the parts between each successive samples.
- **Can we recover these values? And How?**
- **Can we go back from the discrete signal to the original continuous signal?**



# Sampling Theorem

- A bandlimited signal having no spectral components above  $f_{max}$  (Hz), can be determined uniquely by values sampled at uniform intervals of  $T_s$  seconds, where
- An analog signal can be reconstructed from a sampled signal without any loss of information if and only if it is:
  - Band limited signal
  - The sampling frequency is at least twice the signal bandwidth

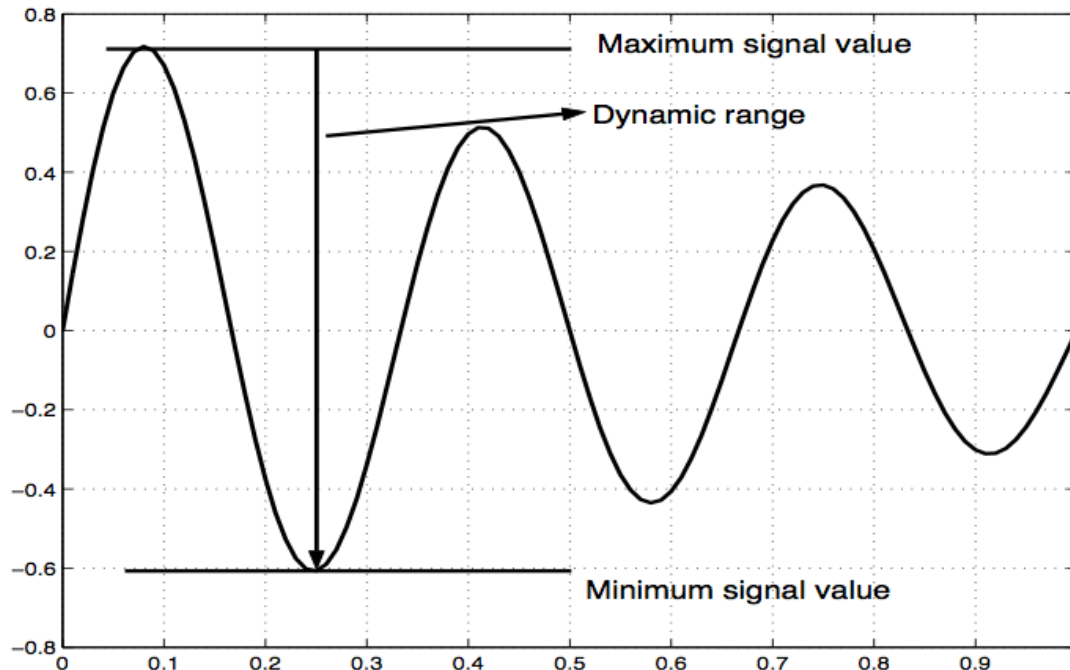
$$T_s \leq \frac{1}{2 f_{\max}}$$

# Quantization

- Quantization is a process of approximating a continuous range of values, very large set of possible discrete values, by a relatively small range of values, small set of discrete values.
- Continuous range  $\rightarrow$  infinite set of values
- Discrete range  $\rightarrow$  finite set of values

# Quantization

- **Dynamic range of a signal**
  - The difference between the highest to lowest value the signal can takes.



# Quantization

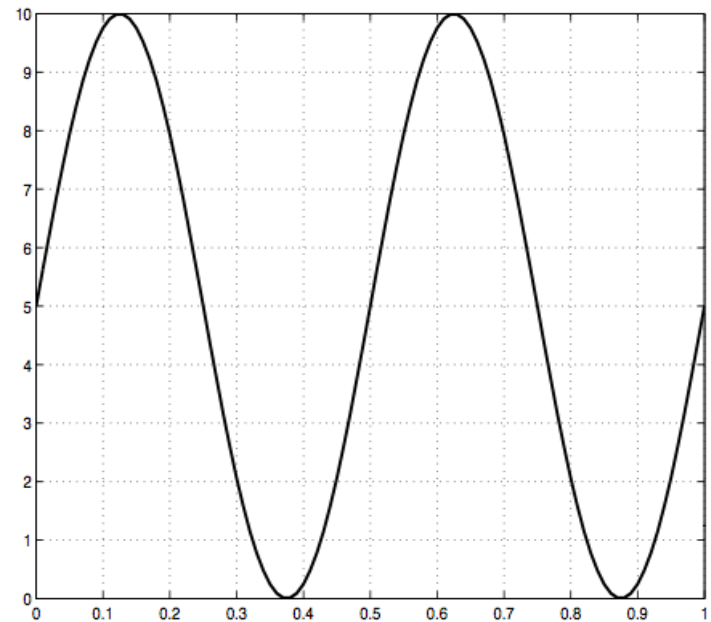
- In the Quantization process, the dynamic range of a signal is divided into  $L$  amplitude levels denoted by  $m_k$ , where  $k = 1, 2, 3, \dots, L$
- $L$  is an integer power of 2
  - $L = 2^k$
  - $k$  is the number of bits needed to represent the amplitude level.
- For example:
  - If we divide the dynamic range into 8 levels,
    - $L = 8 = 2^3$
  - We need 3 bits to represent each level.

# Quantization

- Example:

- Suppose we have an analog signal with the values between  $[0, 10]$ . If we divide the signal into four levels. We have

- $m_1 \rightarrow [0, 2.5]$
    - $m_2 \rightarrow [2.5, 5]$
    - $m_3 \rightarrow [5, 7.5]$
    - $m_4 \rightarrow [7.5, 10]$

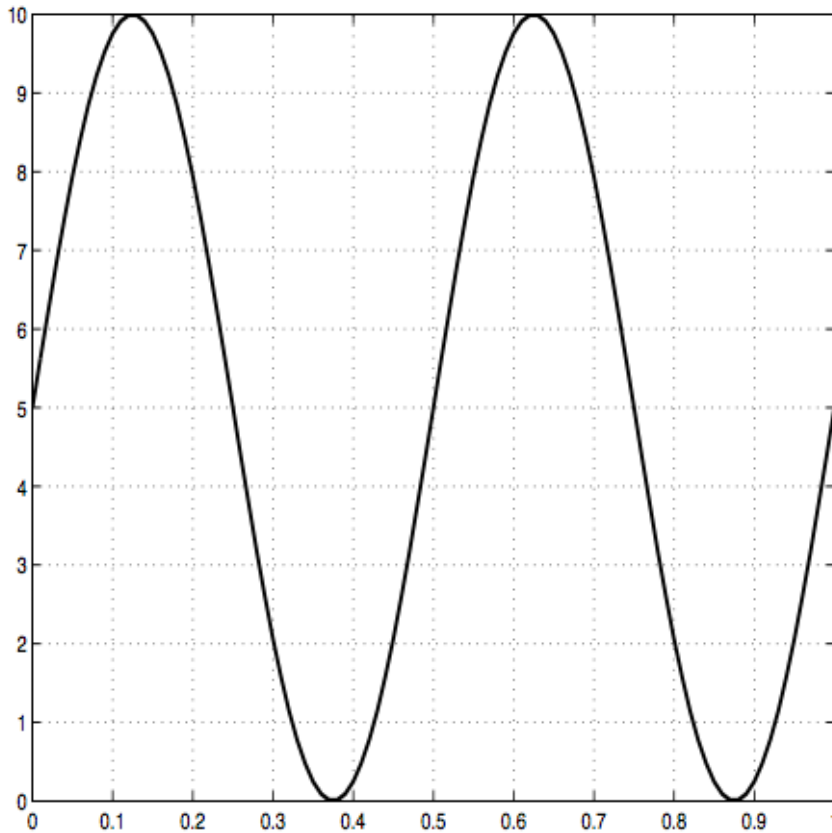


# Quantization

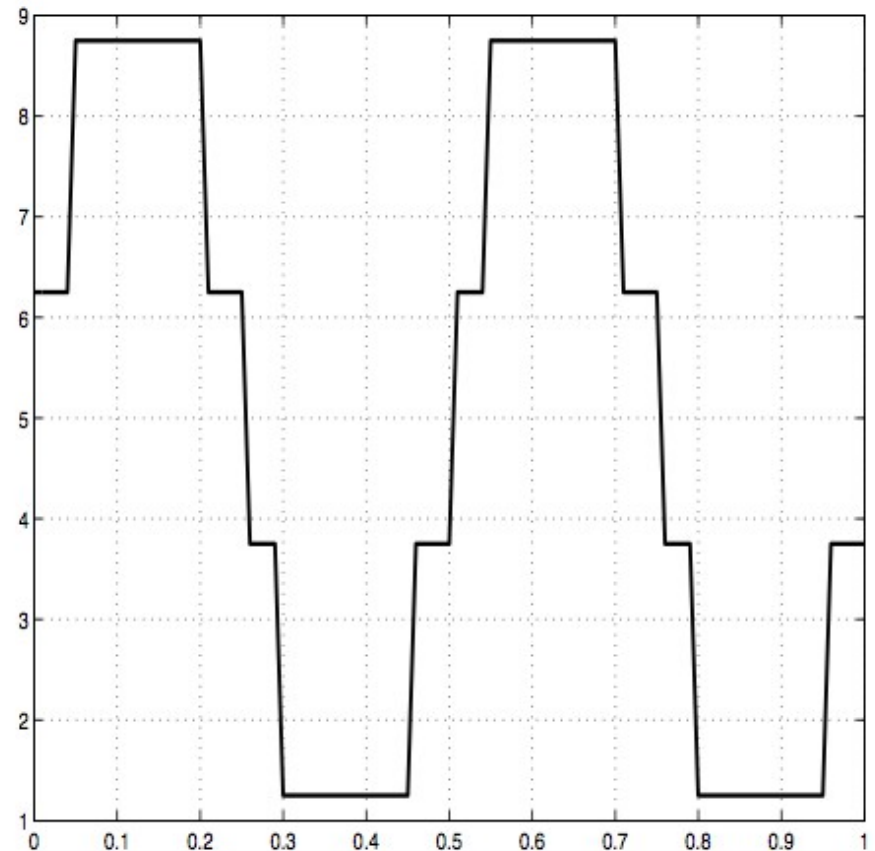
- For every level, we assign a value for the signal if it falls within the same level.

$$Q[v(t)] = \begin{cases} M1 = 1.25 & \text{if the signal in } m1 \\ M2 = 3.75 & \text{if the signal in } m2 \\ M3 = 6.25 & \text{if the signal in } m3 \\ M4 = 8.75 & \text{if the signal in } m4 \end{cases}$$

# Quantization

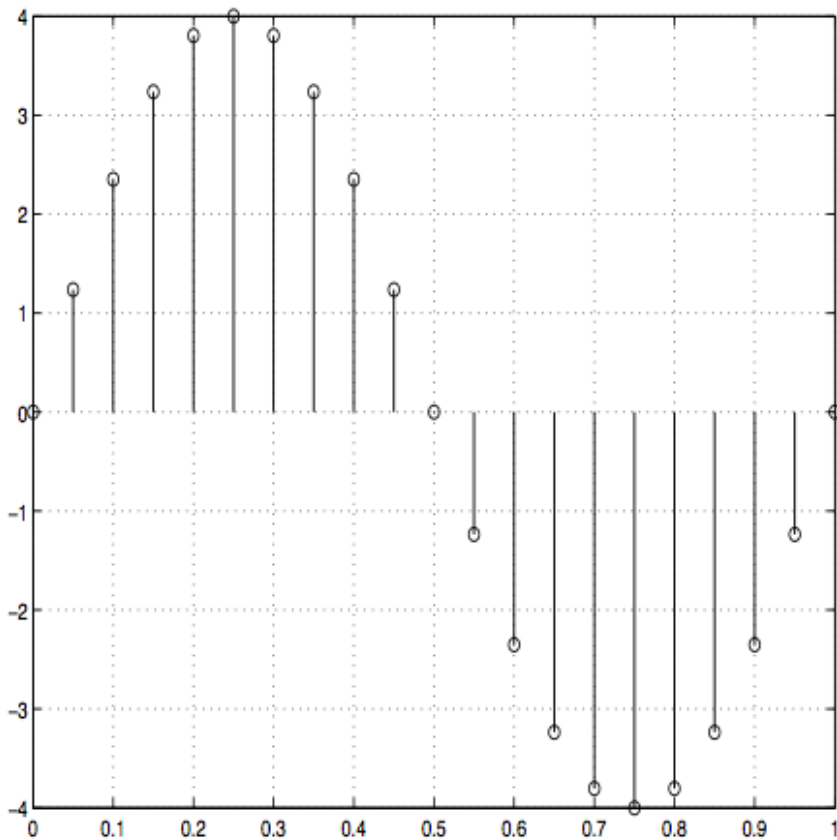


Original Analog Signal  
“Before Quantization”

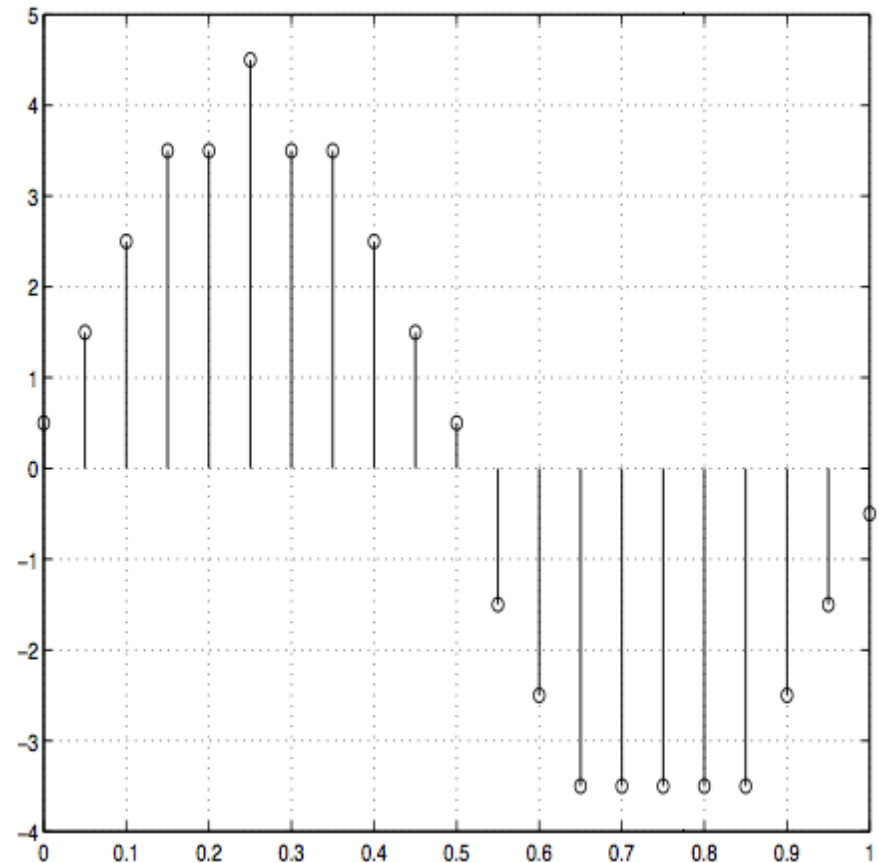


Quantized Analog Signal  
“After Quantization”

# Quantization



Original Discrete Signal  
“Before Quantization”



Quantized Discrete Signal  
“After Quantization”



# Quantization

- The more quantization levels we take the smaller the error between the original and quantized signal.
- Quantization step

$$\Delta = \frac{\text{Dynamic Range}}{\text{No. of Quantization levels}} = \frac{S_{\max} - S_{\min}}{L}$$

- The smaller the  $\Delta$  the smaller the error.

# Coding

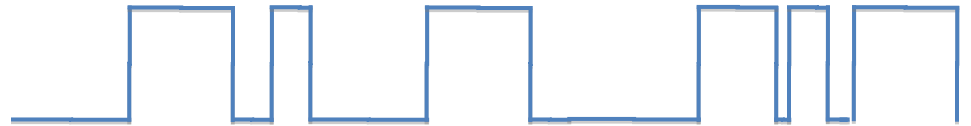
- Assigning a binary code to each quantization level.
- For example, if we have quantized a signal into 16 levels, the coding process is done as the following:

| Step | Code | Step | Code | Step | Code | Step | Code |
|------|------|------|------|------|------|------|------|
| 0    | 0000 | 4    | 0100 | 8    | 1000 | 12   | 1100 |
| 1    | 0001 | 5    | 0101 | 9    | 1001 | 13   | 1101 |
| 2    | 0010 | 6    | 0110 | 10   | 1010 | 14   | 1110 |
| 3    | 0011 | 7    | 0111 | 11   | 1011 | 15   | 1111 |

# Coding

- The binary codes are represented as pulses

- Pulse means 1
- No pulse means 0



- After coding process, the signal is ready to be transmitted through the channel. And Therefore, completing the A/D conversion of an analog signal.

# Chapter 3: Source Coding

12<sup>th</sup> November 2008

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# 3.1 Measure of Information

- What is the definition of “Information” ?
  - News, text data, images, videos, sound etc..
- In Information Theory
  - Information is linked with the element of surprise or uncertainty
  - In terms of probability
  - Information
    - The more probable some event to occur the less information related to its occurrence.
    - The less probable some event to occur the more information we get when it occurs.

# Example1:

- The rush hour in Kuwait is between 7.00 am – 8.00 am
  - A person leaving his home to work at 7.30 will NOT be surprised about the traffic jam → almost no information is gained here
  - A person leaving his home to work at 7.30 will BE surprised if THERE IS NO traffic jam:
    - He will start asking people / family / friends
    - Unusual experience
    - Gaining more information

# Example 2

- The weather temperature in Kuwait at summer season is usually above 30°
- It is known that from the historical data of the weather, the chance that it rains in summer is very rare chance.
  - A person who lives in Kuwait will not be surprised by this fact about the weather
  - A person who lived in Kuwait will BE SURPRISED if it rains during summer, therefore asking about the phenomena. Therefore gaining more knowledge “information”

# How can we measure information?

- Measure of Information
  - Given a digital source with N possible outcomes “messages”, the information sent from the digital source when the  $j^{\text{th}}$  message is transmitted is given by the following equation:

$$I_j = \log_2\left(\frac{1}{p_j}\right) \quad [ \text{ Bits } ]$$



# Example 1

- Find the information content of a message that takes on one of four possible outcomes equally likely

- **Solution**

The probability of each outcome =  $P = \frac{1}{0.25}$   
Therefore,

$$I = \log_2\left(\frac{1}{0.25}\right) = \frac{\log\left(\frac{1}{0.25}\right)}{\log(2)} = 2 \text{ bits}$$

## Example 2

- Suppose we have a digital source that generates binary bits. The probability that it generates “0” is 0.25, while the probability that it generates “1” is 0.75. Calculate the amount of information conveyed by every bit.

## Example 2 (Solution)

- For the binary “0” :  $I = \log_2\left(\frac{1}{0.25}\right) = 2 \text{ bits}$
- For the binary “1”:  $I = \log_2\left(\frac{1}{0.75}\right) = 0.42 \text{ bits}$
- Information conveyed by the “0” is more than the information conveyed by the “1”

## Example 3:

- A discrete source generates a sequence of  $(n)$  bits. How many possible messages can we receive from this source?
- Assuming all the messages are equally likely to occur, how much information is conveyed by each message?

## Example 3 (solution):

- The source generates a sequence of  $n$  bits, each bit takes one of two possible values
  - a discrete source generates either “0” or “1”
- Therefore:
  - We have  $2^n$  possible outcomes
- The Information Conveyed by each outcome

$$I = \log_2\left(\frac{1}{2^n}\right) = \frac{\log(2^n)}{\log(2)} = \frac{n \log(2)}{\log(2)} = n \text{ bits}$$

## 3.3 Entropy

- The entropy of a discrete source  $S$  is the average amount of information ( or uncertainty ) associated with that source.

$$H(s) = \sum_{j=1}^m p_j \log_2 \left( \frac{1}{p_j} \right) \quad [bits]$$

- $m$  = number of possible outcomes
- $P_j$  = probability of the  $j^{\text{th}}$  message

# Importance of Entropy

- Entropy is considered one of the most important quantities in information theory.
- There are two types of source coding:
  - Lossless coding “lossless data compression”
  - Lossy coding “lossy data compression”
- Entropy is the threshold quantity that separates lossy from lossless data compression.

# Example 4

- Consider an experiment of selecting a card at random from a cards deck of 52 cards. Suppose we're interested in the following events:
  - Getting a picture, with probability of :  $\frac{12}{52}$
  - Getting a number less than 3, with probability of:  $\frac{8}{52}$
  - Getting a number between 3 and 10, with a probability of:  $\frac{32}{52}$
- Calculate the Entropy of this random experiment.



## Example 4 (solution) :

- The entropy is given by :

$$H(s) = \sum_{j=1}^3 p_j \log_2 \left( \frac{1}{p_j} \right) \quad [bits]$$

- Therefore,

$$H(s) = \frac{12}{52} \log_2 \left( \frac{52}{12} \right) + \frac{8}{52} \log_2 \left( \frac{52}{8} \right) + \frac{32}{52} \log_2 \left( \frac{52}{32} \right) = 1.335 \quad bits$$

# Source Coding Theorem

- First discovered by Claude Shannon.
- Source coding theorem
  - “A discrete source with entropy rate  $H$  can be encoded with arbitrarily small error probability at any rate  $L$  bits per source output as long as  $L > H$ ”

Where

$H$  = Entropy rate

$L$  = codeword length

If we encode the source with  $L > H \rightarrow$  Trivial Amount of errors

If we encode the source with  $L < H \rightarrow$  we're certain that an error will occur

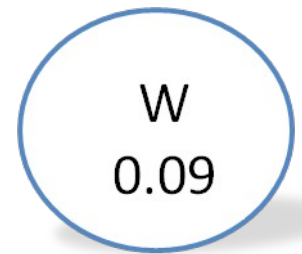
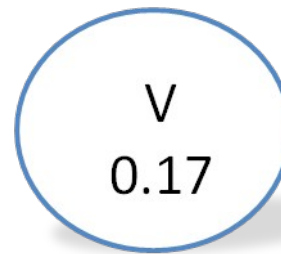
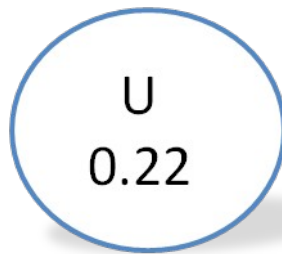
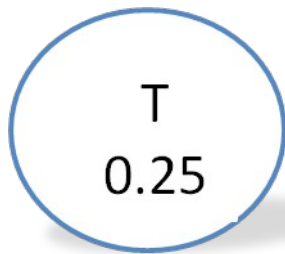
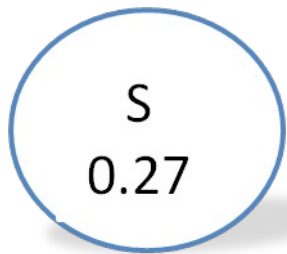
# 3.4 Lossless data compression

- Data compression
  - Encoding information in a relatively smaller size than their original size
    - Like ZIP files (WinZIP), RAR files (WinRAR), TAR files etc..
- Data compression:
  - Lossless: the compressed data are an exact copy of the original data
  - Lossy: the compressed data may be different than the original data
- Lossless data compression techniques:
  - Huffman coding algorithm
  - Lempel-Ziv Source coding algorithm

## 3.4.1 Huffman Coding Algorithm

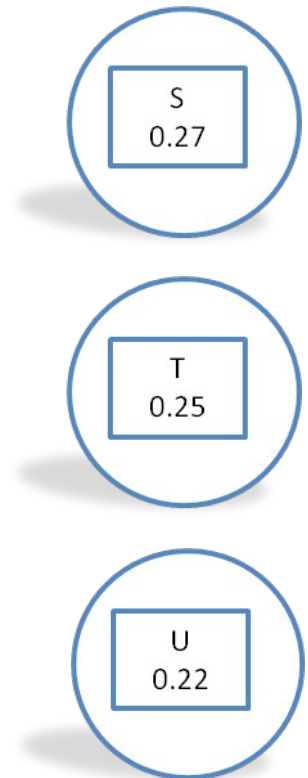
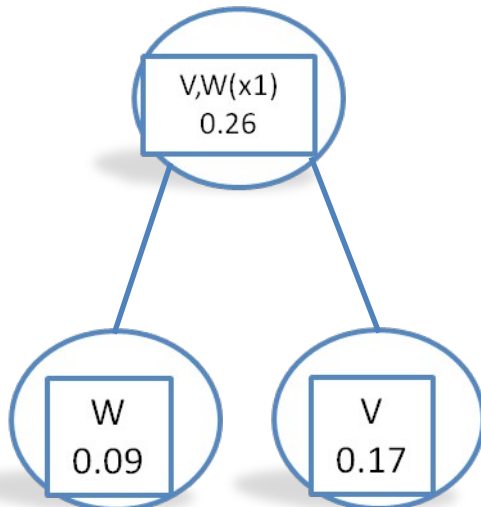
- A digital source generates five symbols with the following probabilities:
  - S ,  $P(s)=0.27$
  - T,  $P(t)=0.25$
  - U,  $P(t)=0.22$
  - V, $P(t)=0.17$
  - W, $P(t)=0.09$
- Use Huffman Coding algorithm to compress this source

*Step1: Arrange the symbols in a descending order according to their probabilities*

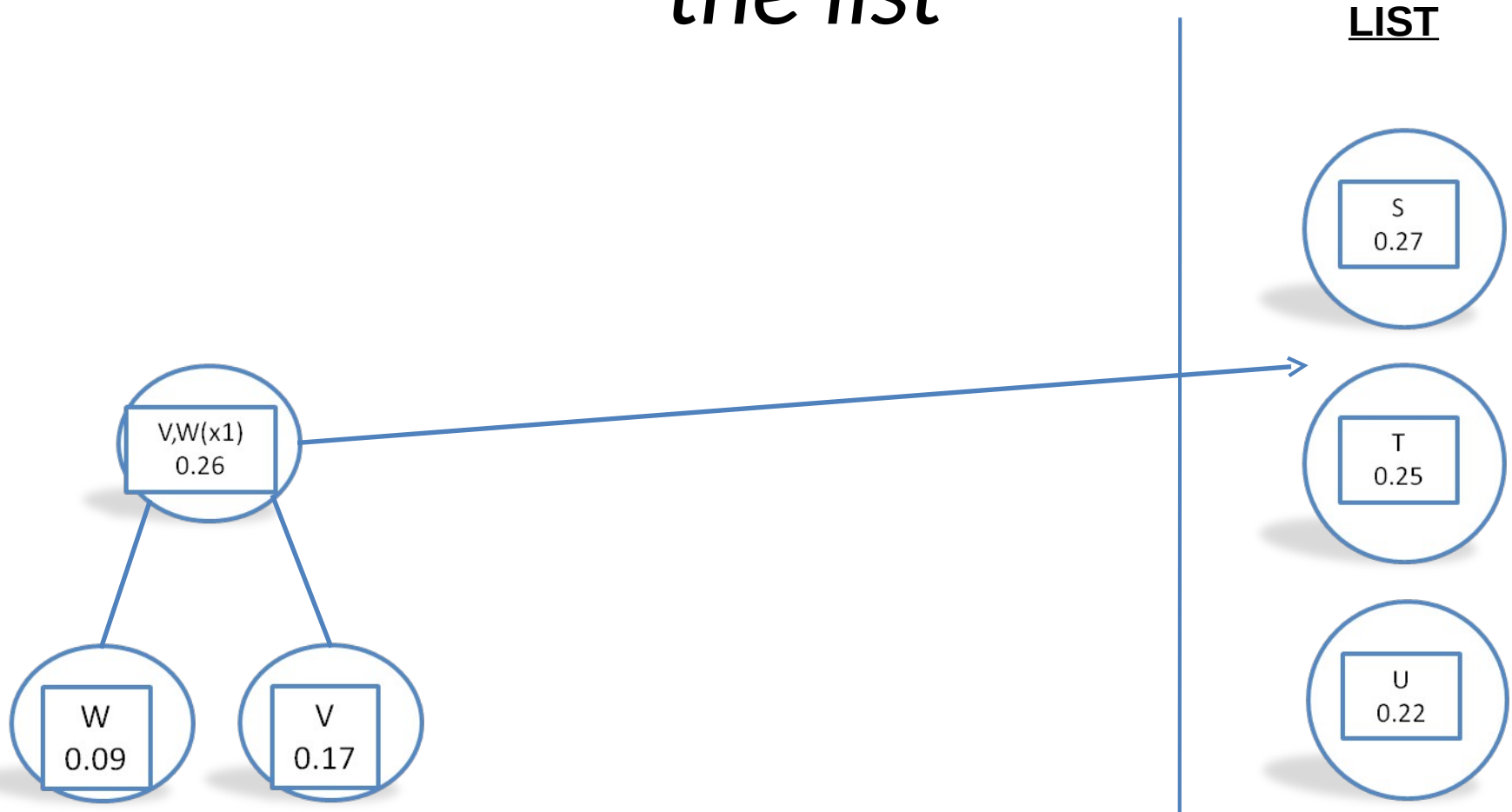


*Step 2: take the symbols with the lowest probabilities and form a leaf*

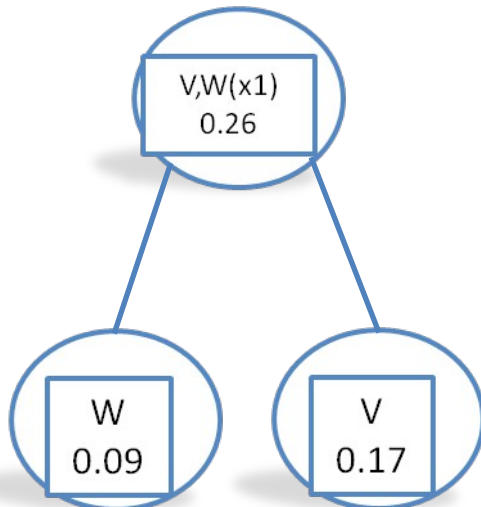
LIST



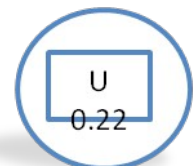
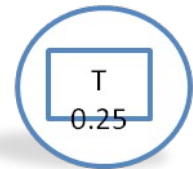
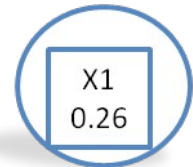
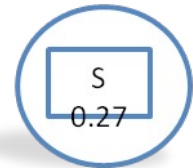
# *Step 3: Insert the parent node to the list*



# Step 3: Insert the parent node to the list

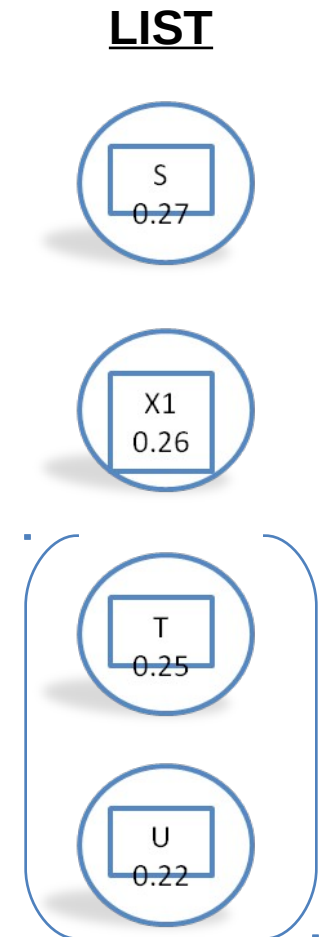
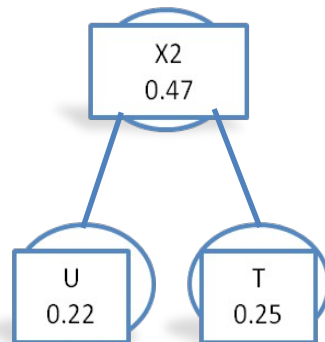
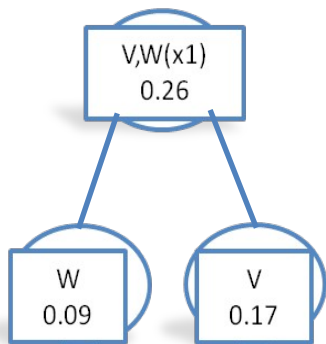


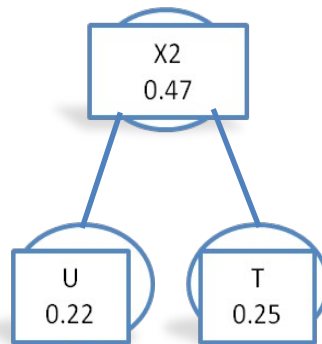
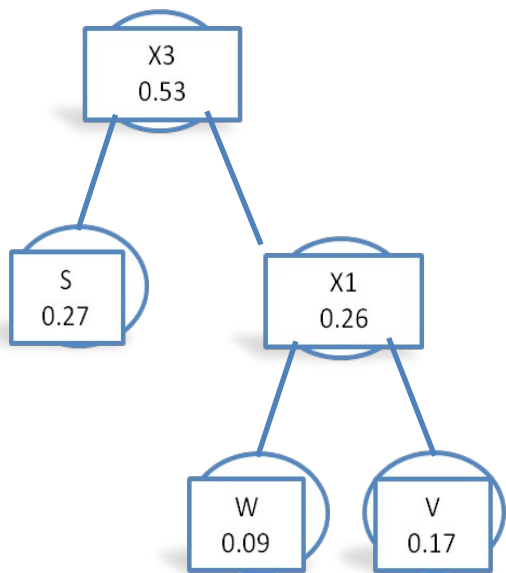
## LIST



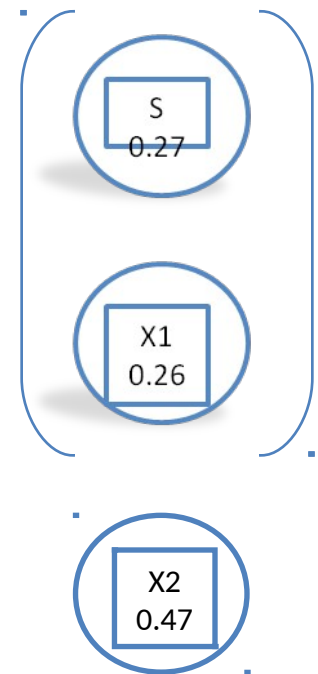


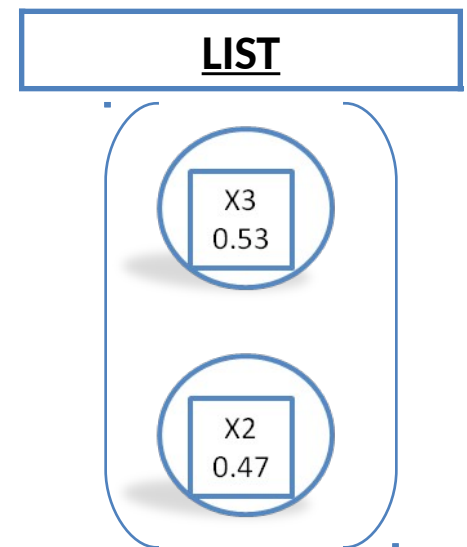
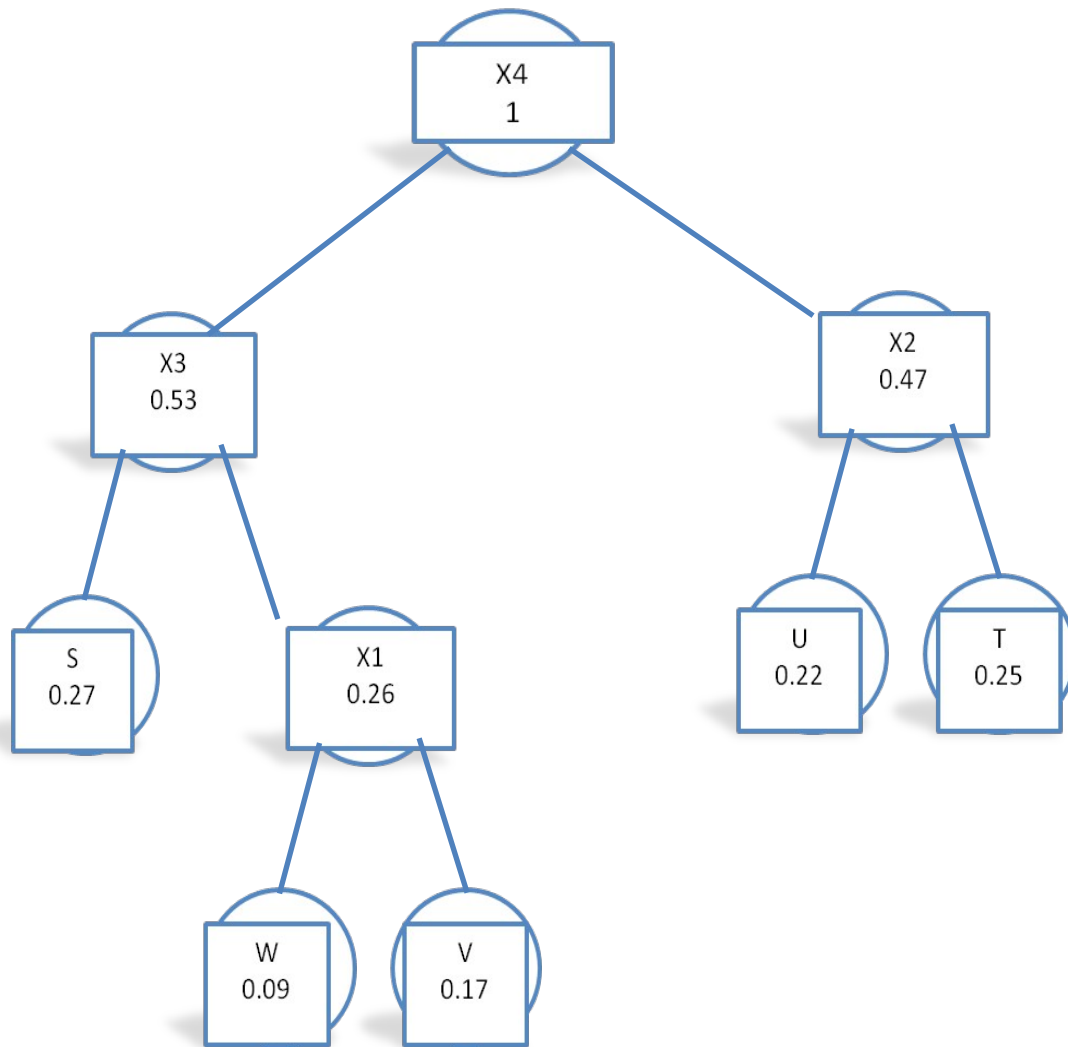
*Step 4: Repeat the same procedure on the updated list till we have only one node*



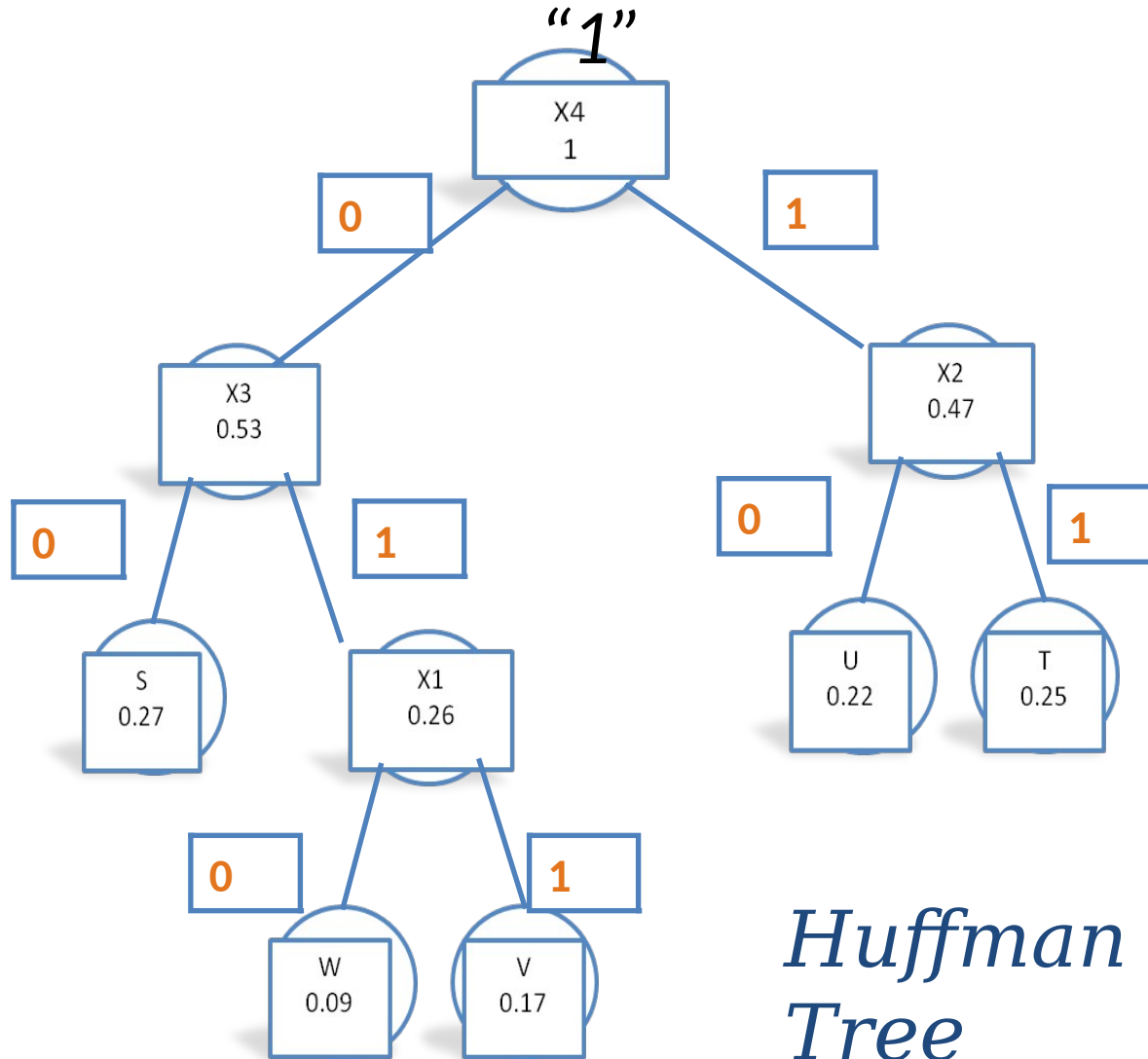


## LIST



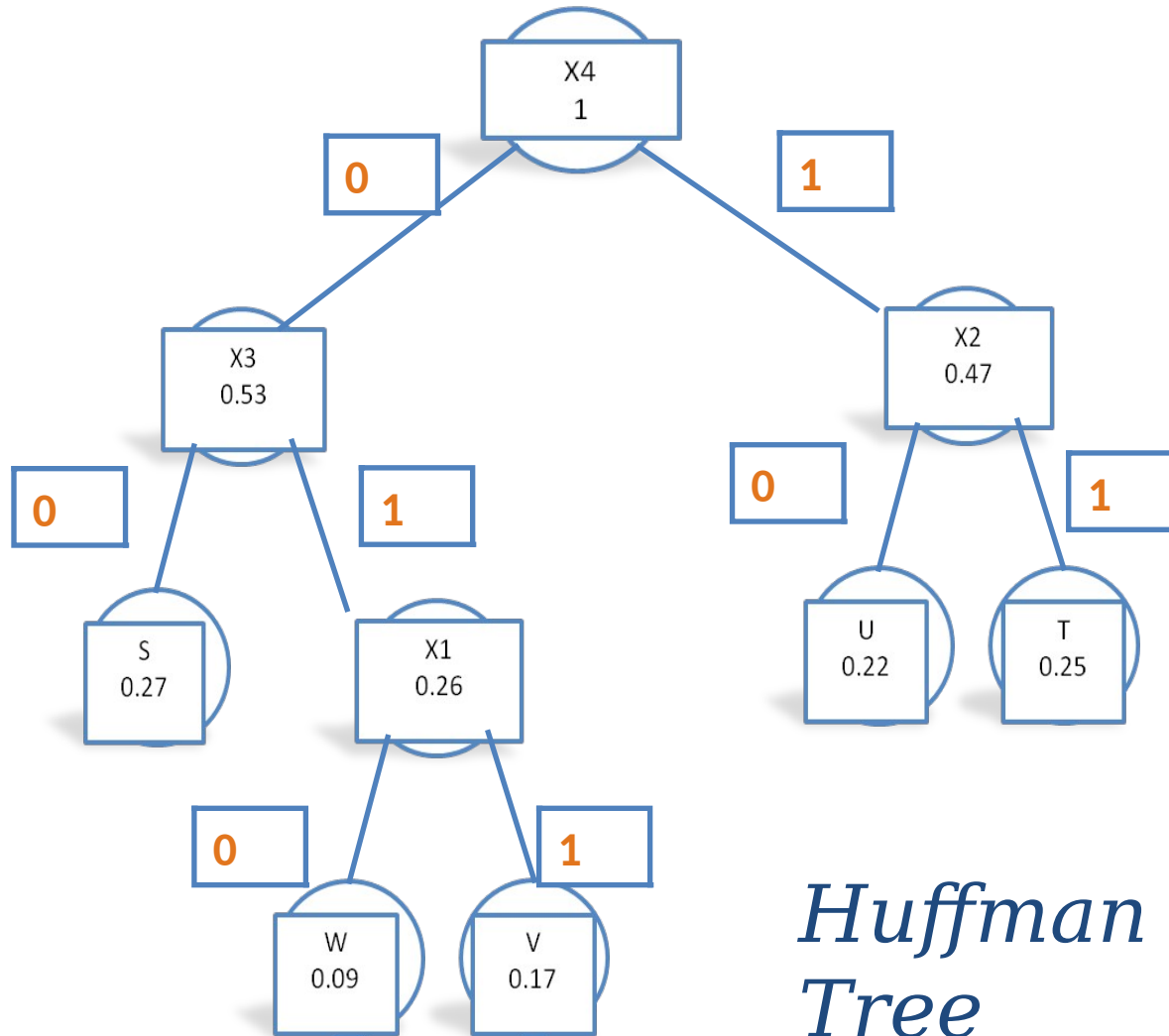


*Step 5: Label each branch of the tree with “0” and*



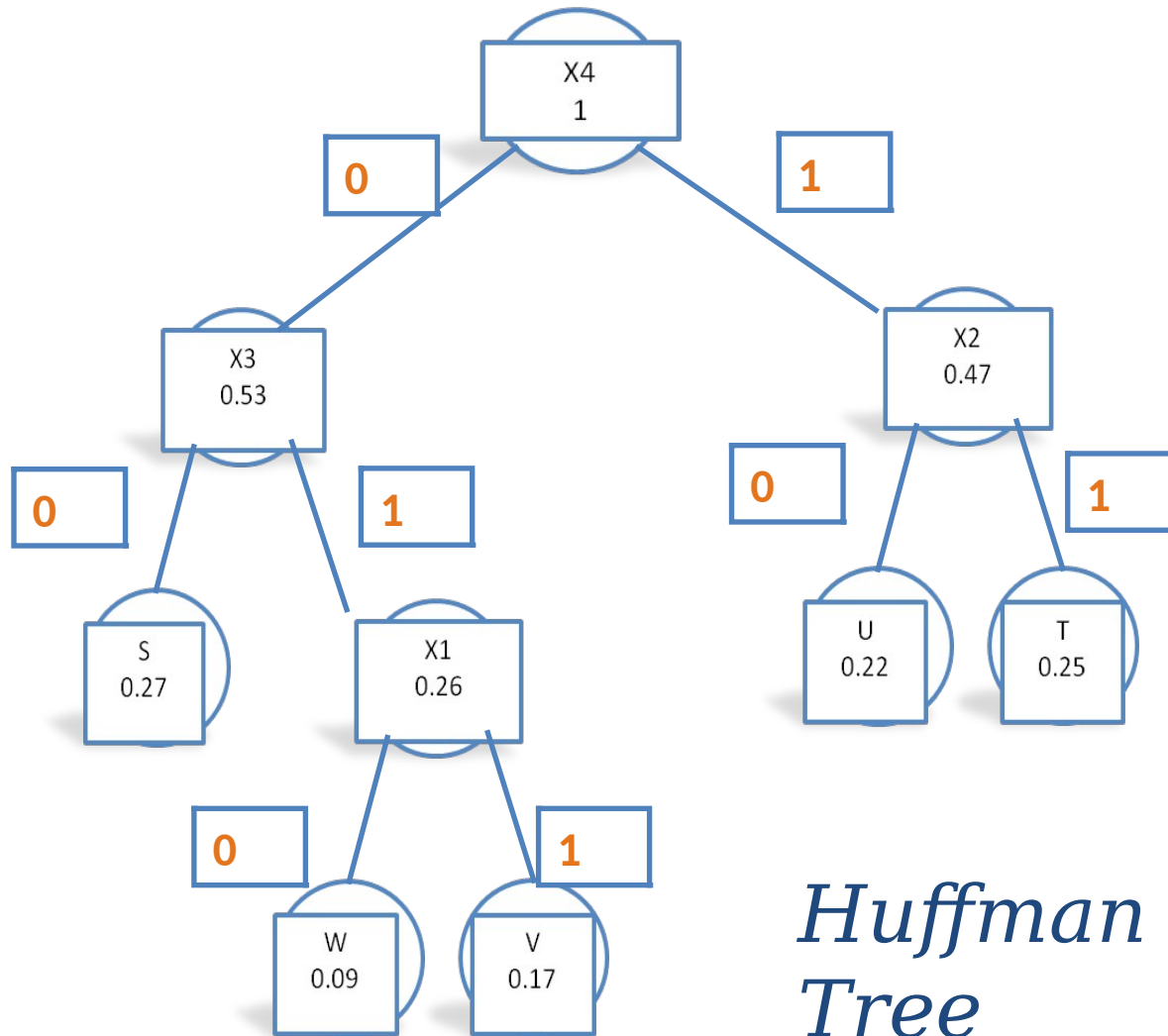
*Huffman Code  
Tree*

*Codeword of  $w = 010$*



*Huffman Code  
Tree*

*Codeword of  $u=10$*



*Huffman Code  
Tree*

# As a result:

| Symbol | Probability | Codeword |
|--------|-------------|----------|
| S      | 0.27        | 00       |
| T      | 0.25        | 11       |
| U      | 0.22        | 10       |
| V      | 0.17        | 011      |
| W      | 0.09        | 010      |

Symbols with higher probability of occurrence have a shorter codeword length, while symbols with lower probability of occurrence have longer codeword length.

# Average codeword length

- The Average codeword length can be calculated by:

$$L = \sum_{j=1}^m P_j l_j$$

- For the previous example we have the average codeword length as follows:

$$L = (0.27 \times 2) + (0.25 \times 2) + (0.22 \times 2) + (0.17 \times 3) + (0.09 \times 3)$$

$$L = 2.26 \text{ bits}$$



# The Importance of Huffman Coding Algorithm

- As seen by the previous example, the average codeword length calculated was 2.26 bits
  - Five different symbols “S,T,U,V,W”
- Without coding, we need three bits to represent all of the symbols
- By using Huffman coding, we’ve reduced the amount of bits to 2.26 bits
  - Imagine transmitting 1000 symbols
    - Without coding, we need 3000 bits to represent them
    - With coding, we need only 2260
  - That is almost 25% reduction “25% compression”

# Chapter 4: Channel Encoding

Abdullah Al-Meshal

# Overview

- Channel encoding definition and importance
- Error Handling techniques
- Error Detection techniques
- Error Correction techniques

# Channel Encoding - Definition

- In digital communication systems an optimum system might be defined as one that minimizes the probability of bit error.
- Error occurs in the transmitted signal due to the transmission in a non-ideal channel
  - Noise exists in channels
  - Noise signals corrupt the transmitted data

# Channel Encoding - Importance

- Channel encoding
  - Techniques used to protect the transmitted signal from the noise effect
- Two basic approaches of channel encoding
  - Automatic Repeat Request (ARQ)
  - Forward Error Correction (FEC)

# Automatic Repeat Request (ARQ)

- Whenever the receiver detects an error in the transmitted block of data, it requests the transmitter to send the block again to overcome the error.
- The request continue “repeats” until the block is received correctly
- ARQ is used in two-way communication systems
  - Transmitter  $\longleftrightarrow$  Receiver

# Automatic Repeat Request (ARQ)

- Advantages:
  - Error detection is simple and requires much simpler decoding equipments than the other techniques
- Disadvantages:
  - If we have a channel with high error rate, the information must be sent too frequently.
  - This results in sending less information thus producing a less efficient system

# Forward Error Correction (FEC)

- The transmitted data are encoded so that the receiver can detect **AND** correct any errors.
- Commonly known as Channel Encoding
- Can be Used in both two-way or one-way transmission.
- FEC is the most common technique used in the digital communication because of its improved performance in correcting the errors.

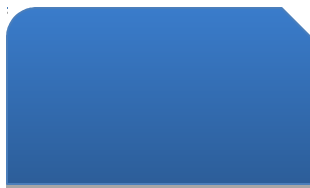


# Forward Error Correction (FEC)

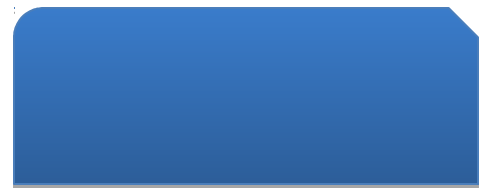
- Improved performance because:
  - It introduces redundancy in the transmitted data in a controlled way
  - Noise averaging : the receiver can average out the noise over long time of periods.

# Error Control Coding

- There are two basic categories for error control coding
  - Block codes
  - Tree Codes
- Block Codes:
  - A block of  $k$  bits is mapped into a block of  $n$  bits



Block of K bits



Block of n bits

# Error Control Coding

- tree codes are also known as codes with memory, in this type of codes the encoder operates on the incoming message sequence continuously in a serial manner.
- Protecting data from noise can be done through:
  - Error Detection
  - Error Correction

# Error Control Coding

- Error Detection
  - We basically check if we have an error in the received data or not.
- There are many techniques for the detection stage
  - Parity Check
  - Cyclic Redundancy Check (CRC)

# Error Control Coding

- Error Correction
  - If we have detected an error “or more” in the received data and we can correct them, then we proceed in the correction phase
- There are many techniques for error correction as well:
  - Repetition Code
  - Hamming Code

# Error Detection Techniques

- Parity Check
  - Very simple technique used to detect errors
- In Parity check, a parity bit is added to the data block
  - Assume a data block of size  $k$  bits
  - Adding a parity bit will result in a block of size  $k+1$  bits
- The value of the parity bit depends on the number of “1”s in the  $k$  bits data block

# Parity Check

- Suppose we want to make the number of 1's in the transmitted data block odd, in this case the value of the parity bit depends on the number of 1's in the original data
  - if we a message = 1010111
    - $k = 7$  bits
  - Adding a parity check so that the number of 1's is even
    - The message would be : 1010111 $1$
    - $k+1 = 8$  bits
- At the reciever ,if one bit changes its values, then an error can be detected

# Example - 1

- At the transmitter, we need to send the message  $M = 1011100$ .
  - We need to make the number of one's odd
- Transmitter:
  - $k=7$  bits ,  $M = 1011100$
  - $k+1=8$  bits ,  $M' = 10111001$
- Receiver:
  - If we receive  $M' = 10111001 \rightarrow$  no error is detected
  - If we receive  $M' = 10111000 \rightarrow$  an Error is detected



# Parity Check

- If an odd number of errors occurred, then the error still can be detected “assuming a parity bit that makes an odd number of 1’s”
- Disadvantage:
  - If an even number of errors occurred, the error can NOT be detected “assuming a parity bit that makes an odd number of 1’s”

# Cyclic Redundancy Check (CRC)

- A more powerful technique used for error detection.
- Can detect the errors with very high probability.
- Procedure:
  - $M$  = original data message (  $m$  bits)
  - $P$  = Predefined pattern
  - $MX^n$  =  $M$  concatenated with  $n$  zeros
  - $R$  = remainder of dividing (  $MX^n / P$  )

# Sender Operation

- Sender
  - The transmitter performs the division  $M / P$
  - The transmitter then computes the remainder  $R$
  - It then concatenates the remainder with the message “M:R”
  - Then it sends the encoded message over the channel “M:R”.
  - The channel transforms the message  $M:R$  into  $M':R'$

# Receiver Operation

- Receiver:
  - The receiver receives the message  $M':R'$
  - It then performs the division of the message by the predetermined pattern  $P$ , “  $M':R' / P$  ”
    - If the remainder is zero, then it assumes the message is not corrupted “Does not have any error”. Although it may have some.
    - If the remainder is NON-zero, then for sure the message is corrupted and contain error/s.

# Division process

- The division used in the CRC is a modulo-2 arithmetic division.
- Exactly like ordinary long division, only simpler, because at each stage we just need to check whether the leading bit of the current three bits is 0 or 1.
  - If it's 0, we place a 0 in the quotient and exclusively OR the current bits with zeros.
  - If it's 1, we place a 1 in the quotient and exclusively OR the current bits with the divisor.

## Example - 2

- Using CRC for error detection and given a message  $M = 10110$  with  $P = 110$ , compute the following
  - Frame check Sum (FCS)
  - Transmitted frame
  - Received frame and check if there is any error in the data

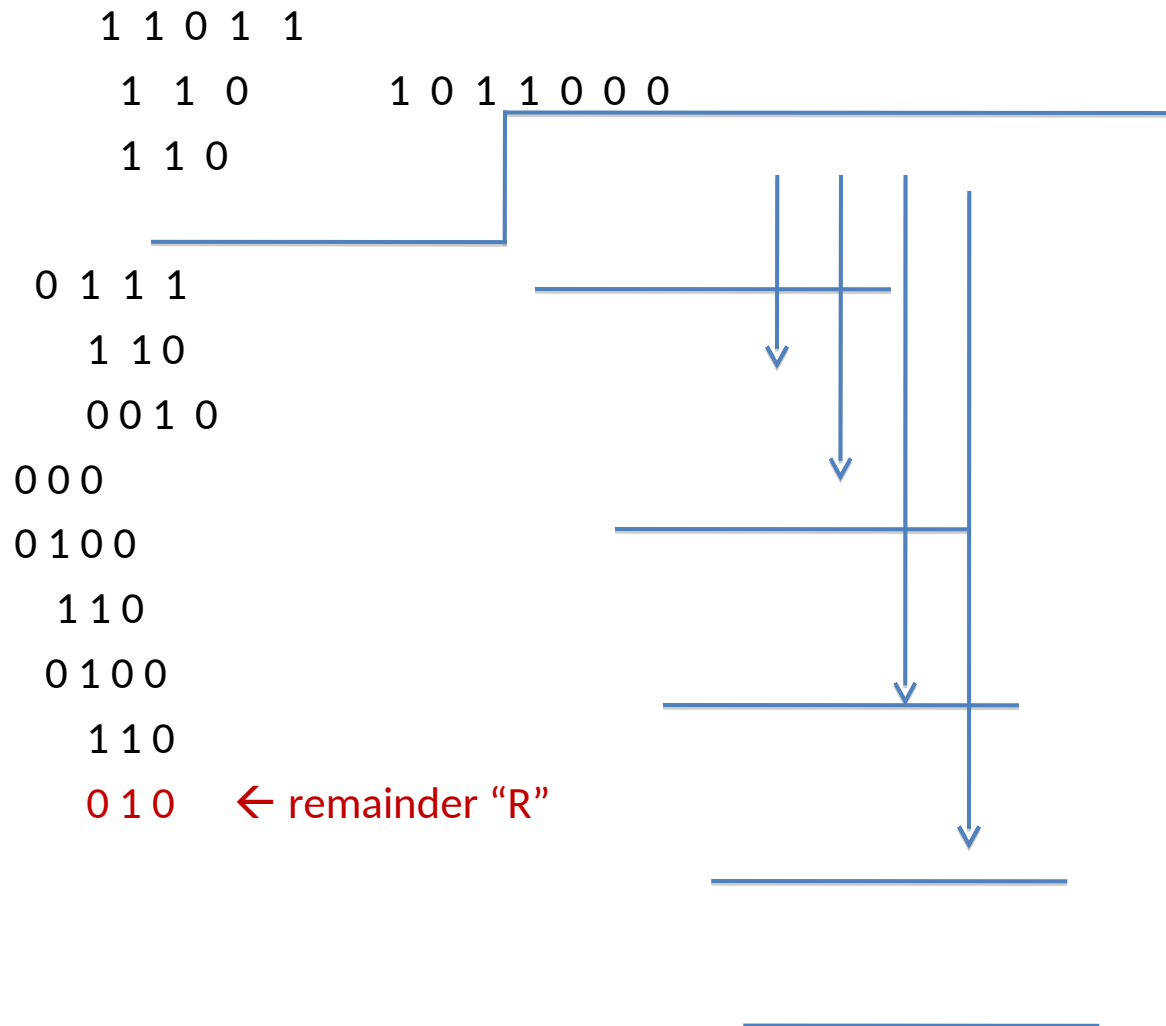
$$M = 10110$$

$$P = 110 \quad "n+1 = 3 \text{ bits}" \rightarrow n = 2 \text{ bits}$$

Hence, Frame check sum has a length = 2 bits.

$$M 2^n = M 2^2 = 10110\textcolor{red}{00}$$

At the Transmitter





# Now,

We concatenate M with R

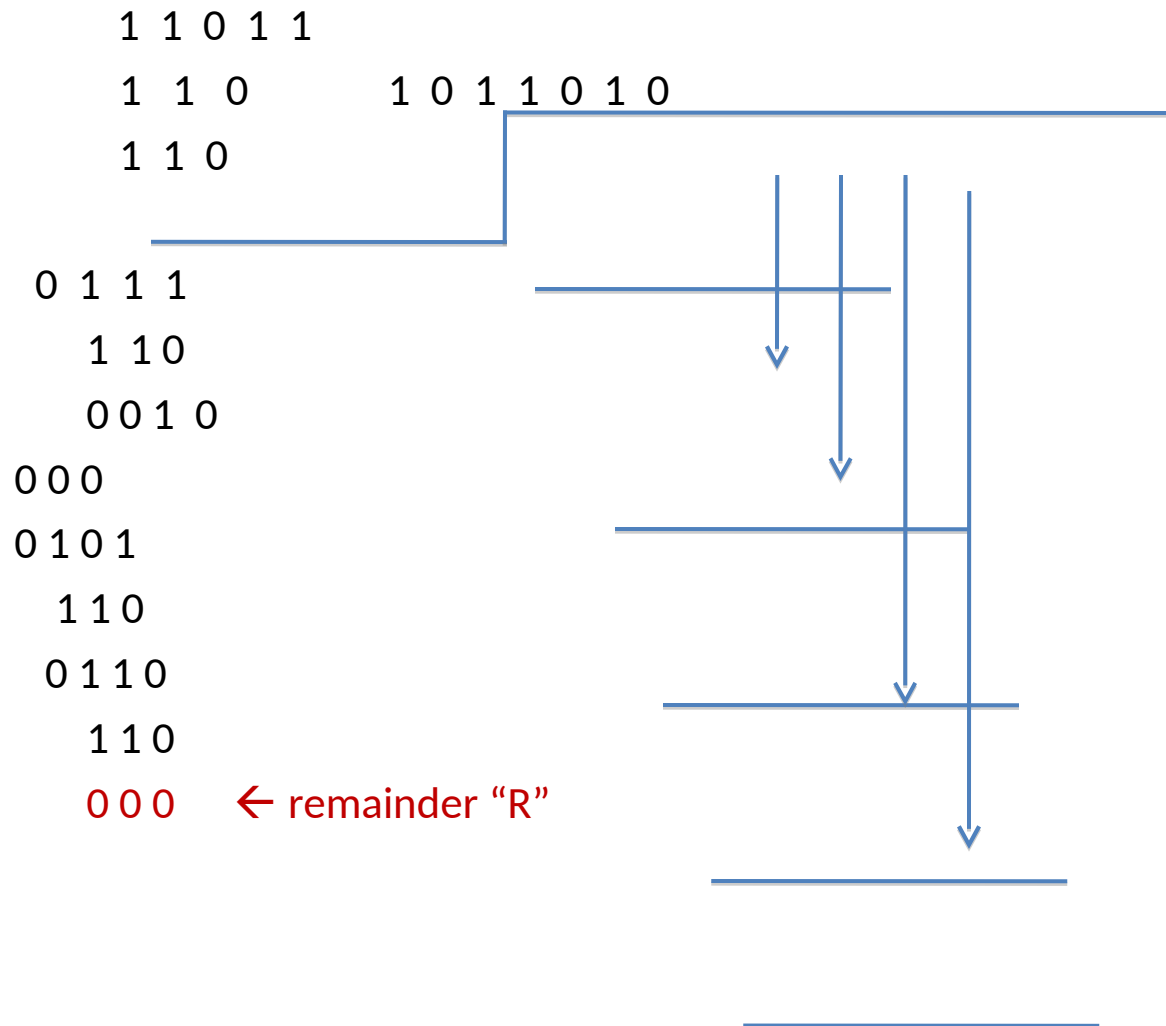
$M = 10110$

$R = 10$

→  $M:R = 1011010$

→  $M:R$  is the transmitted message

At the Receiver



- Since there is no remainder at the receiver, then we can say that the message is not corrupted “i.e. does not contain any errors”
- If the remainder is not zero, then we are sure that the message is corrupted.

# Example - 3

- Let  $M = 111001$  and  $P = 11001$
- Compute the following:
  - Frame check Sum (FCS)
  - Transmitted frame
  - Received frame and check if there is any error in the data

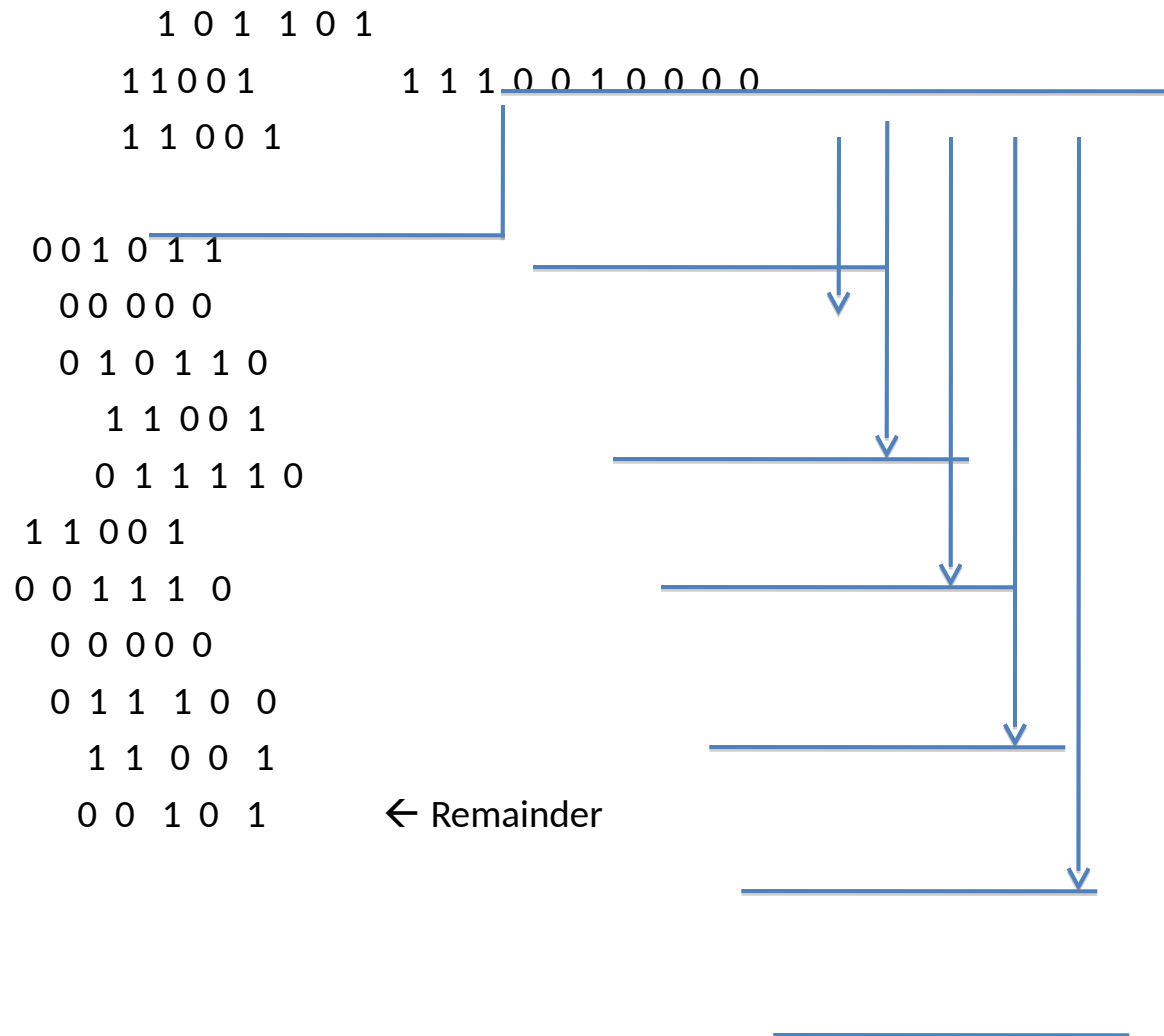
$$M = 111001$$

$$P = 11001 \text{ “ } n+1 = 5 \text{ bits” } \rightarrow n = 4 \text{ bits}$$

Hence, Frame check sum has a length = 4 bits.

$$M 2^n = M 2^4 = 1110010000$$

At the Transmitter



Now,

We concatenate M with R

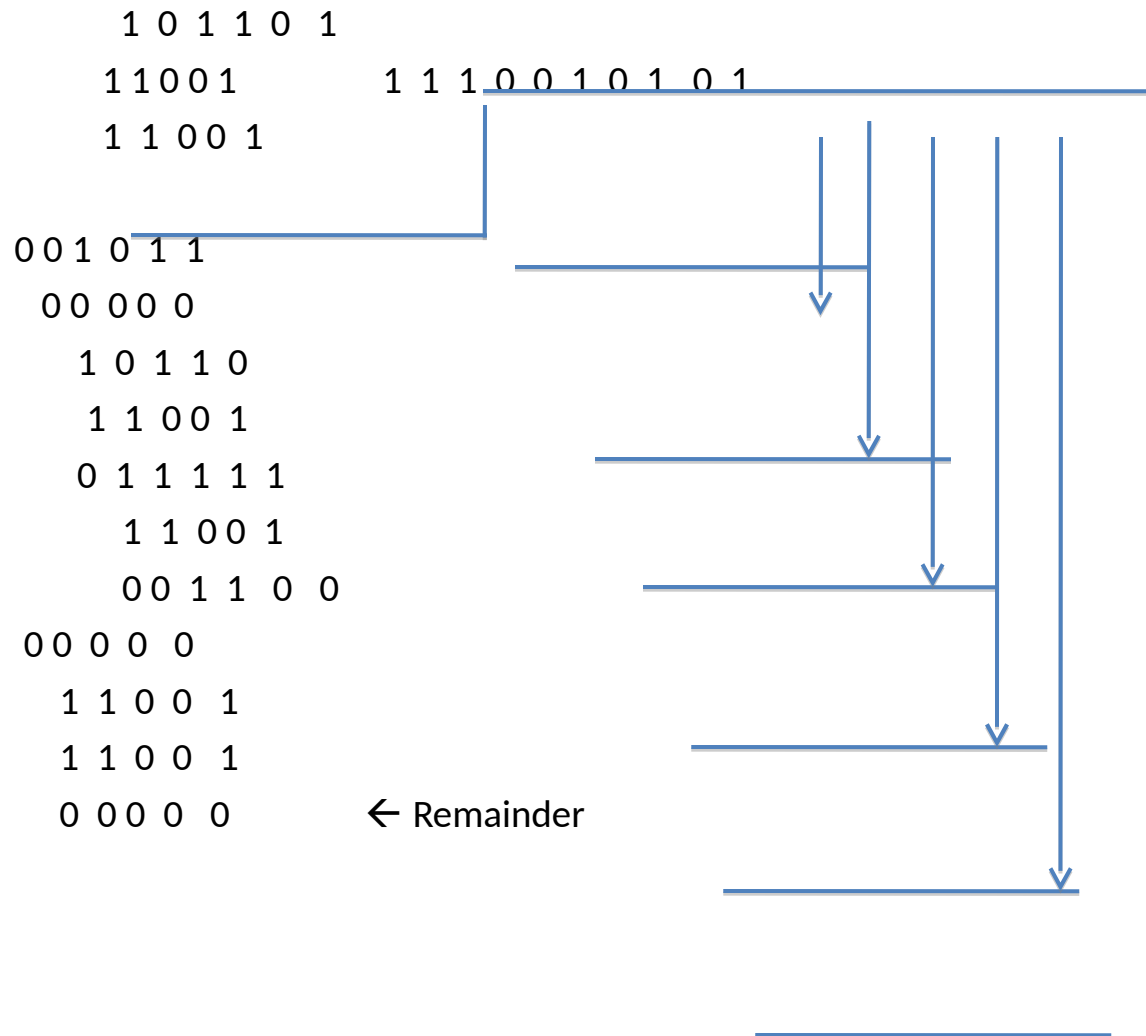
$M = 111001$

$R = 0101$

$\rightarrow M:R = 111001101$

$\rightarrow M:R$  is the transmitted message

At the Receiver





# Chapter 5: Modulation Techniques

Abdullah Al-Meshal

# Introduction

- After encoding the binary data, the data is now ready to be transmitted through the physical channel
- In order to transmit the data in the physical channel we must convert the data back to an electrical signal
  - Convert it back to an analog form
- This process is called modulation

# Modulation - Definition

- Modulation is the process of changing a parameter of a signal using another signal.
- The most commonly used signal type is the sinusoidal signal that has the form of :
  - $V(t) = A \sin ( \omega t + \theta )$
- A : amplitude of the signal
- $\omega$  : radian frequency
- $\theta$  : Phase shift

# Modulation

- In modulation process, we need to use two types of signals:
  - Information, message or transmitted signal
  - Carrier signal
- Let's assume the carrier signal is of a sinusoidal type of the form  $x(t) = A \sin (\omega t + \theta )$
- Modulation is letting the message signal to change one of the carrier signal parameters

# Modulation

- If we let the carrier signal amplitude changes in accordance with the message signal then we call the process **amplitude modulation**
- If we let the carrier signal frequency changes in accordance with the message signal then we call this process **frequency modulation**

# Digital Data Transmission

- There are two types of Digital Data Transmission:
  - 1) Base-Band data transmission
    - Uses low frequency carrier signal to transmit the data
  - 2) Band-Pass data transmission
    - Uses high frequency carrier signal to transmit the data

# Base-Band Data Transmission

- Base-Band data transmission = Line coding
- The binary data is converted into an electrical signal in order to transmit them in the channel
- Binary data are represented using amplitudes for the 1's and 0's
- We will presenting some of the common base-band signaling techniques used to transmit the information

# Line Coding Techniques

- Non-Return to Zero (NRZ)
- Unipolar Return to Zero (Unipolar-RZ)
- Bi-Polar Return to Zero (Bi-polar RZ)
- Return to Zero Alternate Mark Inversion (RZ-AMI)
- Non-Return to Zero – Mark (NRZ-Mark)
- Manchester coding (Biphase)

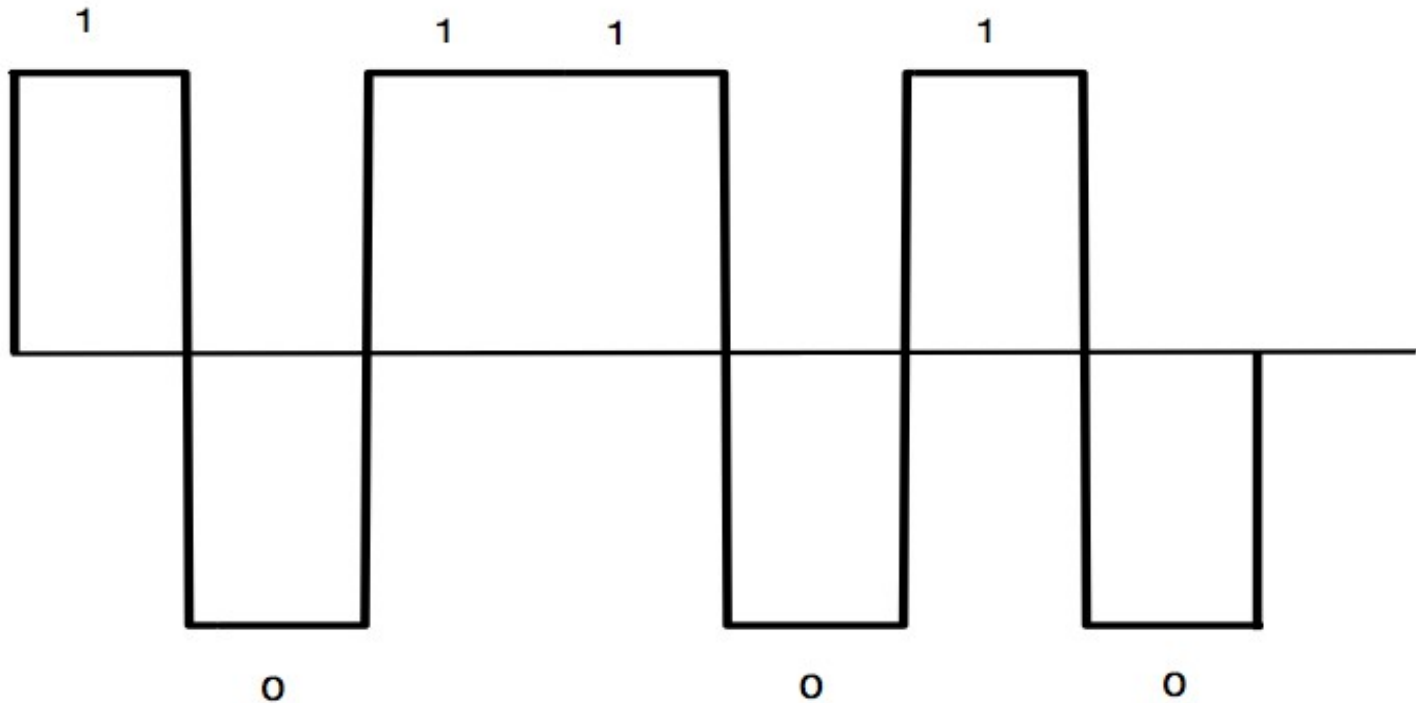


# Non-Return to Zero (NRZ)

- The “1” is represented by some level
- The “0” is represented by the opposite
- The term non-return to zero means the signal switched from one level to another without taking the zero value at any time during transmission.

# NRZ - Example

- We want to transmit  $m=1011010$

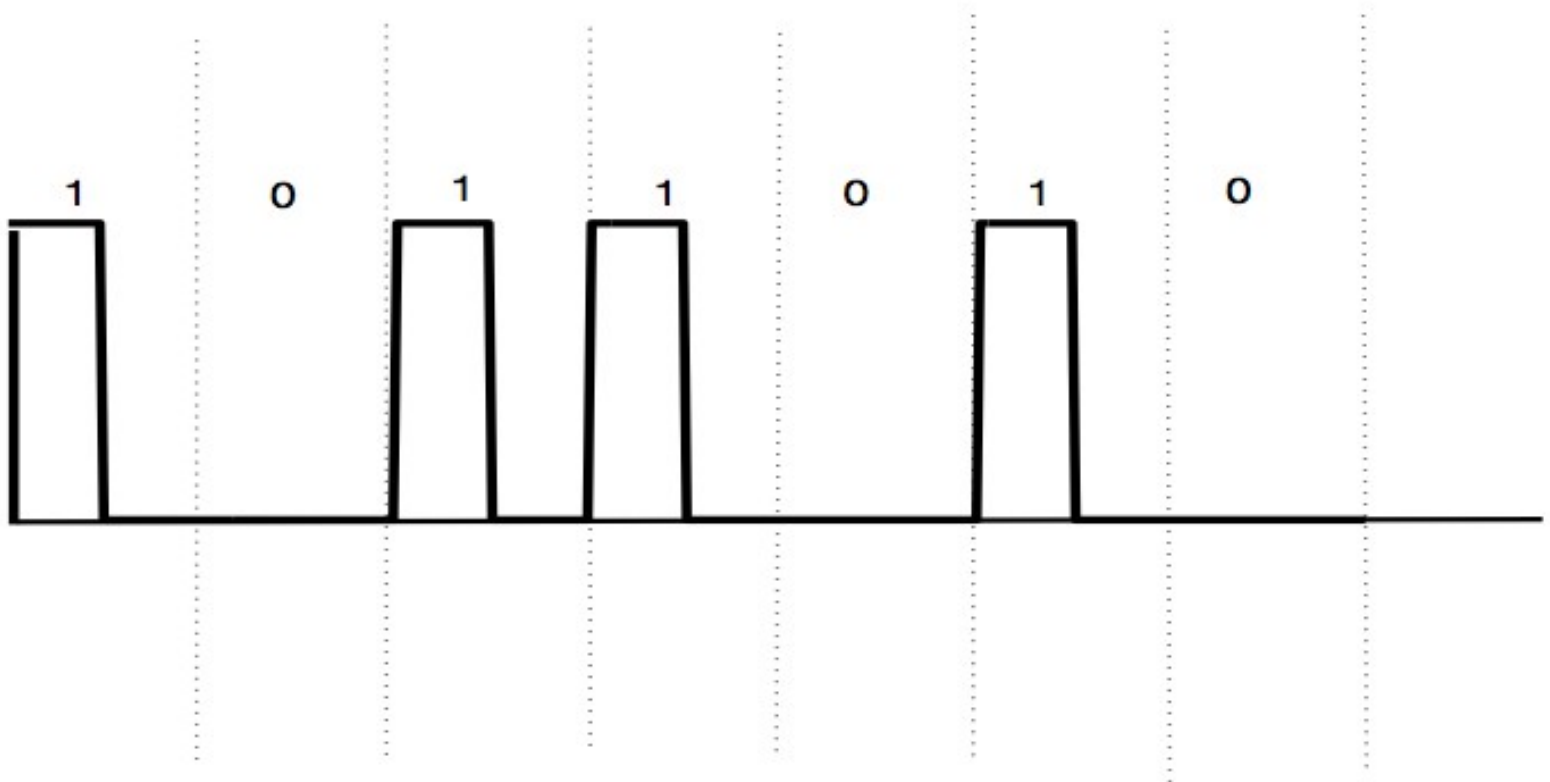


# Unipolar Return to Zero (Unipolar RZ)

- Binary “1” is represented by some level that is half the width of the signal
- Binary “0” is represented by the absence of the pulse

# Unipolar RZ - Example

- We want to transmit  $m=1011010$

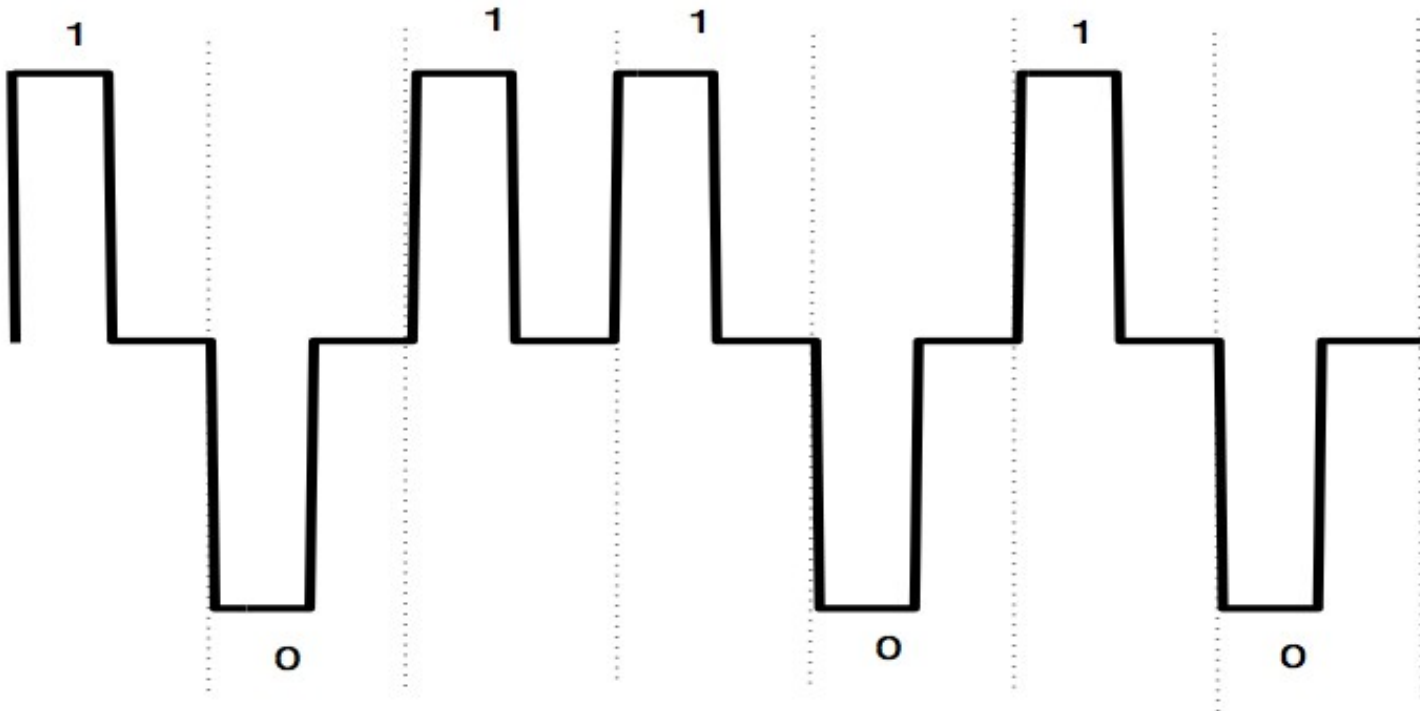


# Bipolar Return to Zero (Bipolar RZ)

- Binary “1” is represented by some level that is half the width of the signal
- Binary “0” is represented a pulse that is half width the signal but with the opposite sign

# Bipolar RZ - Example

- We want to transmit  $m=1011010$

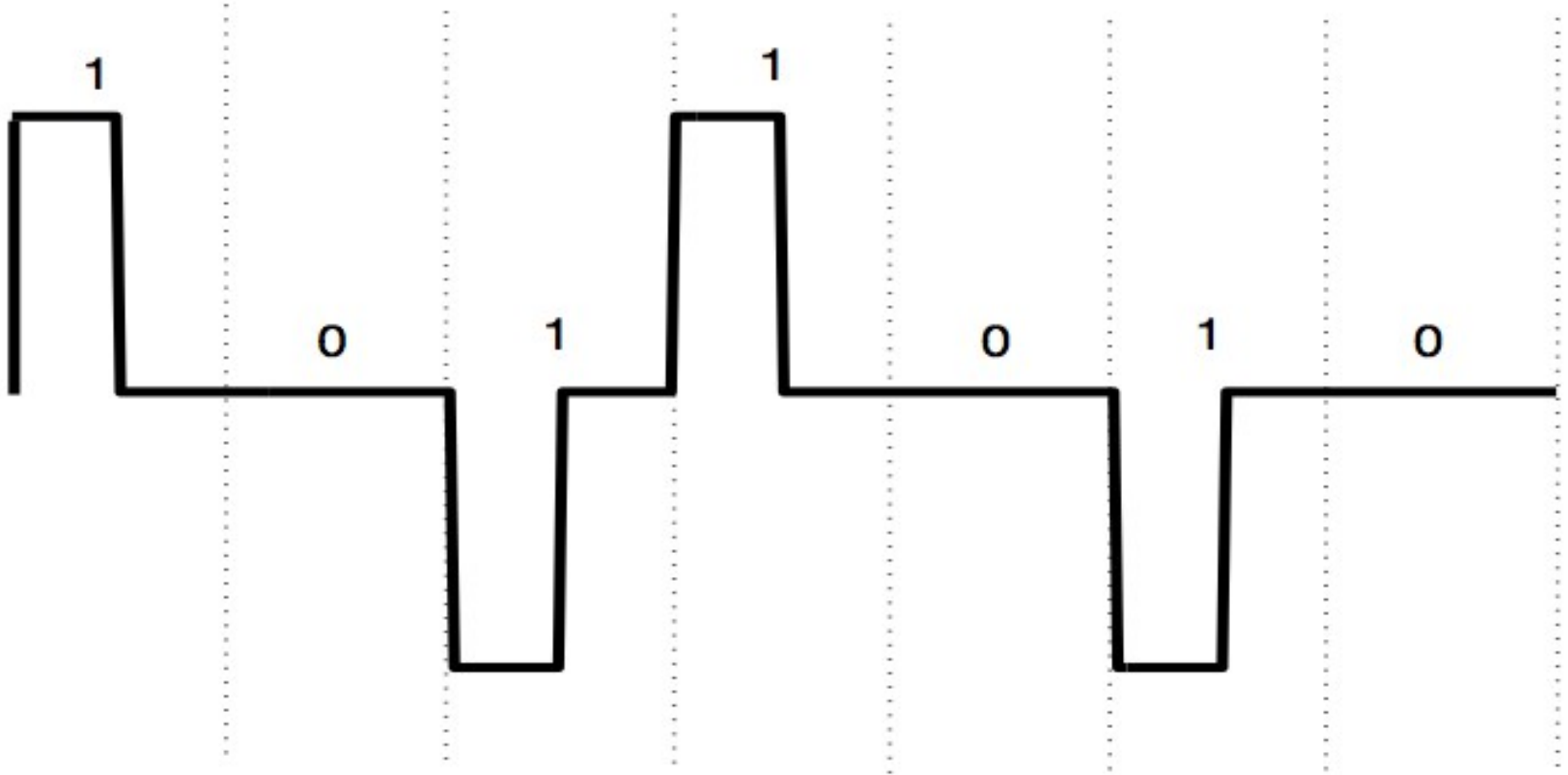


# Return to Zero Alternate Mark Inversion (RZ-AMI)

- Binary “1” is represented by a pulse alternating in sign
- Binary “0” is represented with the absence of the pulse

# RZ-AMI - Example

- We want to transmit  $m=1011010$



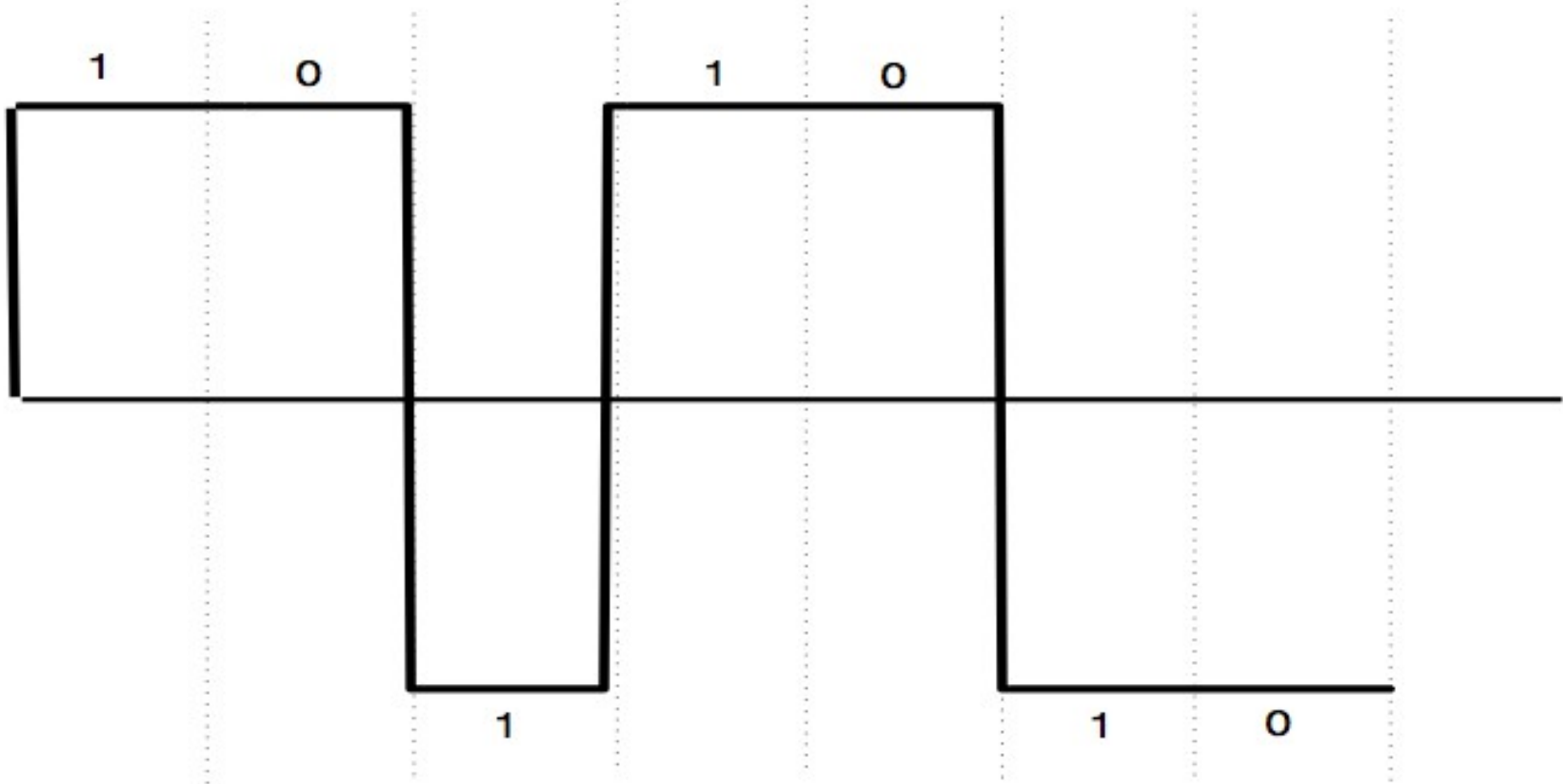


# Non-Return to Zero – Mark (NRZ-Mark)

- Also known as differential encoding
- Binary “1” represented in the change of the level
  - High to low
  - Low to high
- Binary “0” represents no change in the level

# NRZ-Mark - Example

- We want to transmit  $m=1011010$

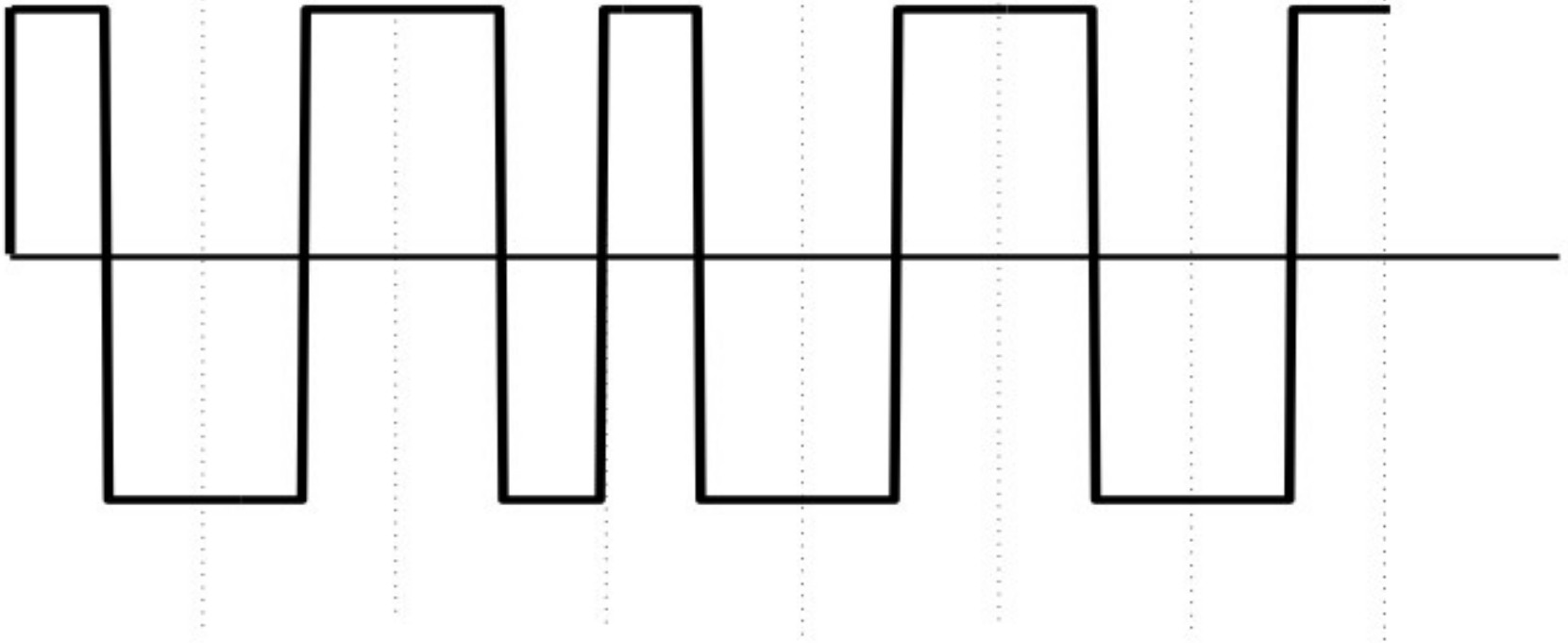


# Manchester coding (Biphase)

- Binary “1” is represented by a positive pulse half width the signal followed by a negative pulse
- Binary “0” is represented by a negative pulse half width the signal followed by a positive pulse

# Manchester coding - Example

- We want to transmit  $m=1011010$



# Scrambling Techniques

- The idea of data scrambling is to replace a sequence of bits with another sequence to achieve certain goals.
  - For example, a long sequence of zeros or long sequence of ones.
- This long sequence of zeros or ones can cause some synchronization problem at the receiver.
- To solve this problem, we replace these sequences by special codes which provides sufficient transmissions for the receiver's clock to maintain synchronization.

# Scrambling techniques

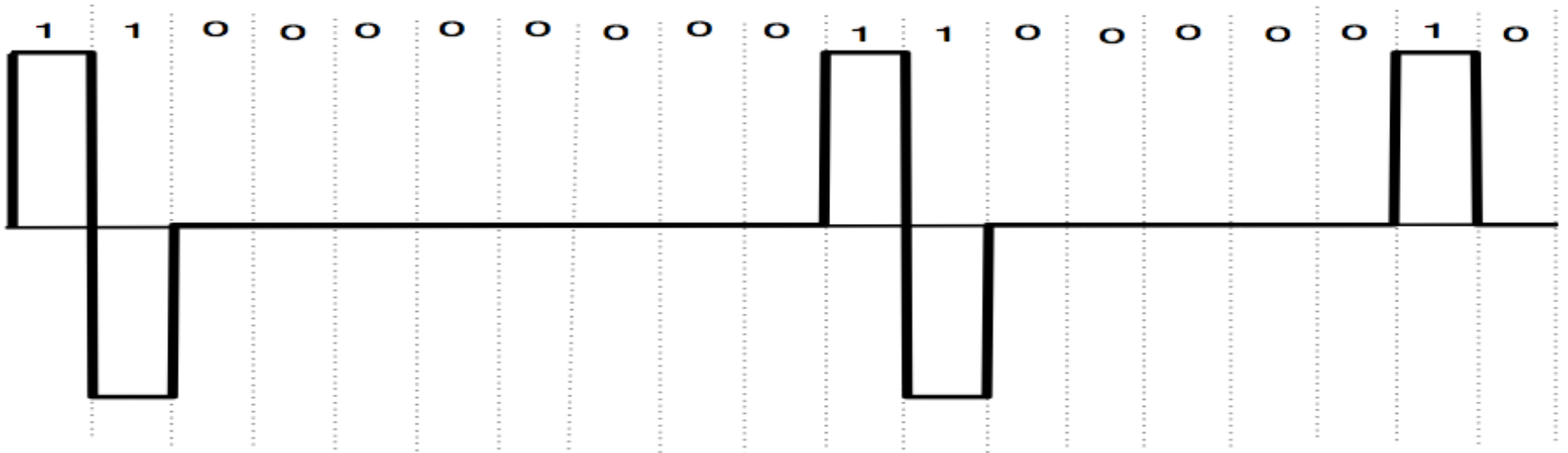
- We present two techniques used to replace a long sequence of zeros by some special type of sequences
  - Bipolar 8 Zero substitution (B8ZS)
  - High Density bipolar 3 Zeros (HDB3)

# Bipolar 8 Zero substitution (B8ZS)

- Used in North America to replace sequences with 8 zeros with a special sequence according to the following rules:
  - If an octet (8) of all zeros occurs and the last voltage pulse preceding this octet was positive, then 000+-0-+
  - If an octet of all zeros occurs and the last voltage pulse preceding this octet was negative, then 000-+0+-

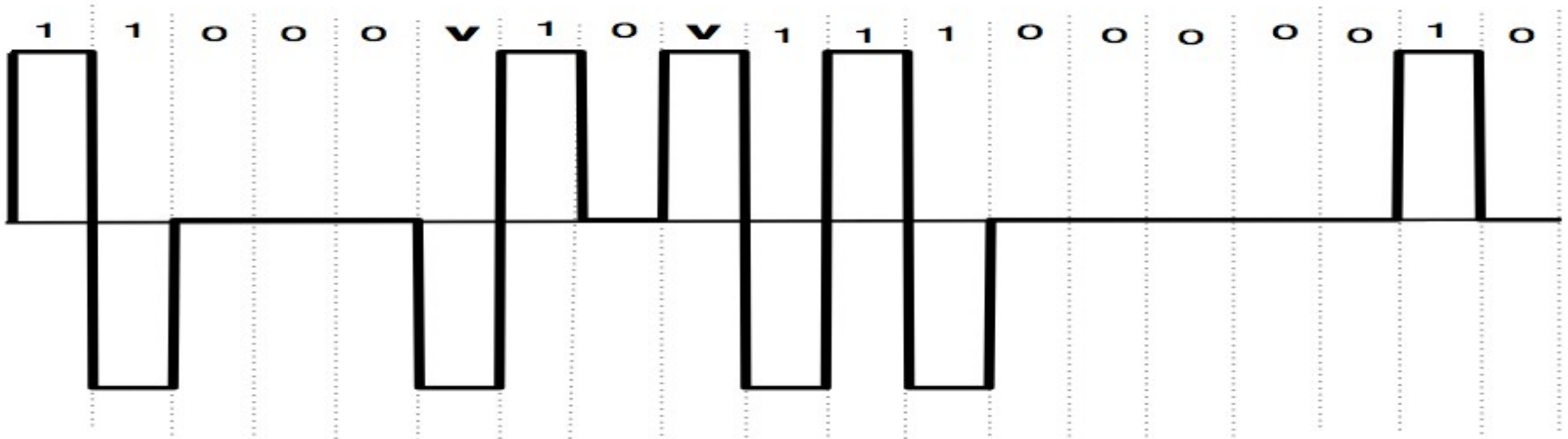
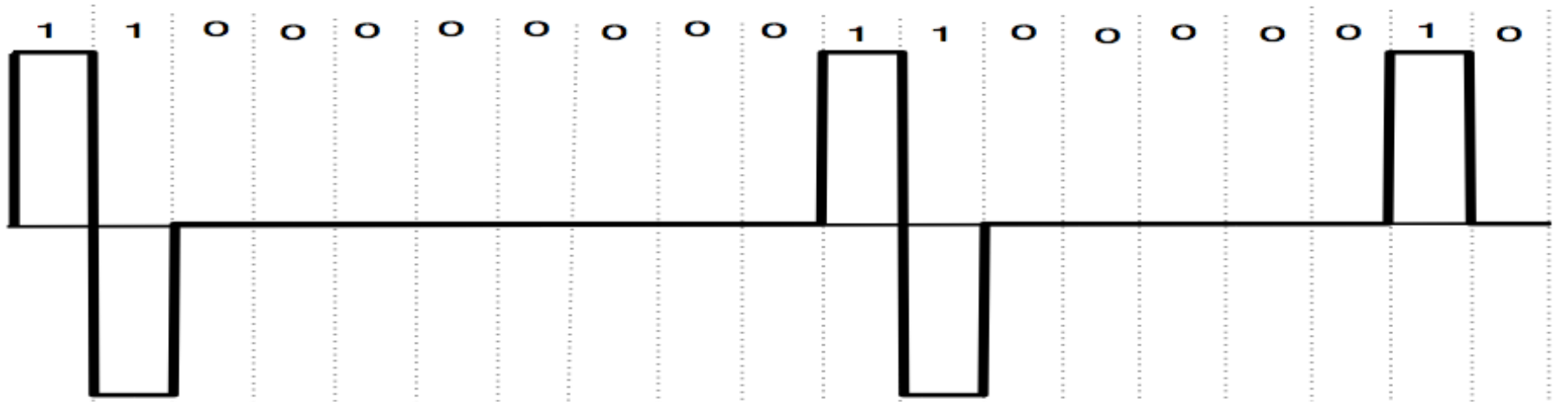
# B8ZS - Example

- Suppose that we want to encode the message  
 $m=1100000000110000010$





# B8ZS – Example (Continue)



# High Density bipolar 3 Zeros (HDB3)

- Used in Europe and Japan to replace a sequence of 4 zeros according to the following rules:

| Sign of preceding pulse | Number of ones (pulses) since the last substitution |         |
|-------------------------|-----------------------------------------------------|---------|
|                         | Odd                                                 | Even    |
| Negative                | 0 0 0 -                                             | + 0 0 + |
| Positive                | 0 0 0 +                                             | - 0 0 - |

# Transmission

- **Transmission bandwidth:** the transmission bandwidth of a communication system is the band of frequencies allowed for signal transmission, in another word it is the band of frequencies at which we are allowed to use to transmit the data.

# Bit Rate

- **Bit Rate** : is the number of bits transferred between devices per second
- If each bit is represented by a pulse of width  $T_b$ , then the bit rate

$$R_b = \frac{1}{T_b} \quad \text{bits/sec}$$

# Example – Bit rate calculation

- Suppose that we have a binary data source that generates bits. Each bit is represented by a pulse of width  $T_b = 0.1$  mSec
- Calculate the bit rate for the source

- **Solution**

$$R_b = \frac{1}{T_b} = \frac{1}{0.1 \times 10^{-3}} = 10000 \quad \text{bits/sec}$$

# Example – Bit rate calculation

- Suppose we have an image frame of size 200x200 pixels. Each pixel is represented by three primary colors red, green and blue (RGB). Each one of these colors is represented by 8 bits, if we transmit 1000 frames in 5 seconds what is the bit rate for this image?

# Example – Bit rate calculation

- We have a total size of  $200 \times 200 = 40000$  pixels
- Each pixel has three colors, RGB that each of them has 8 bits.
  - $3 \times 8 = 24$  bits ( for each pixel with RGB)
- Therefore, for the whole image we have a total size of  $24 \times 40000 = 960000$  bits
- Since we have 1000 frames in 5 seconds, then the total number of bits transmitted will be  $1000 \times 960000 = 960000000$  bits in 5 seconds
- Bit rate =  $960000000/5 = 192000000$  bits/second

# Baud rate (Symbol rate)

- The number of symbols transmitted per second through the communication channel.
- The symbol rate is related to the bit rate by the following equation:

- $R_b$  = bit rate
  - $R_s$  = symbol rate
  - $N$  = Number of bits per symbol
- $$R_s = \frac{R_b}{N}$$



# Baud rate (Symbol rate)

- We usually use symbols to transmit data when the transmission bandwidth is limited
- For example, we need to transmit a data at high rate and the bit duration  $T_b$  is very small; to overcome this problem we take a group of more than one bit, say 2, therefore :

$$T_b \rightarrow f_o = \frac{1}{T_b}$$

$$2T_b \rightarrow f = \frac{1}{2T_b} = \frac{1}{2} f_o$$

$$4T_b \rightarrow f = \frac{1}{4T_b} = \frac{1}{4} f_o$$

# Baud rate (Symbol rate)

- We notice that by transmitting symbols rather than bits we can reduce the spectrum of the transmitted signal.
- Hence, we can use symbol transmission rather than bit transmission when the transmission bandwidth is limited

# Example

- A binary data source transmits binary data, the bit duration is  $1\mu\text{sec}$ , Suppose we want to transmit symbols rather than bits, if each symbol is represented by four bits. what is the symbol rate?
- Each bit is represented by a pulse of duration  $1\mu$  second, hence the bit rate

$$R_b = \frac{1}{1 \times 10^{-6}} = 1000000 \text{ bits/sec}$$

## Example (Continue)

- Therefore, the symbol rate will be

$$R_s = \frac{R_b}{N} = \frac{1000000}{4} = 250000 \text{ symbols/sec}$$

# Chapter 5: Modulation Techniques (Part II)

Abdullah Al-Meshal

# Introduction

- Bandpass data transmission
- Amplitude Shift Keying (ASK)
- Phase Shift Keying (PSK)
- Frequency Shift Keying (FSK)
- Multilevel Signaling ( $M_{ary}$  Modulation)

# Bandpass Data Transmission

- In communication, we use modulation for several reasons in particular:
  - To transmit the message signal through the communication channel efficiently.
  - To transmit several signals at the same time over a communication link through the process of multiplexing or multiple access.
  - To simplify the design of the electronic systems used to transmit the message.
  - by using modulation we can easily transmit data with low loss

# Bandpass Digital Transmission

- Digital modulation is the process by which digital symbols are transformed into waveforms that are compatible with the characteristics of the channel.
- The following are the general steps used by the modulator to transmit data
  - 1. Accept incoming digital data
  - 2. Group the data into symbols
  - 3. Use these symbols to set or change the phase, frequency or amplitude of the reference carrier signal appropriately.



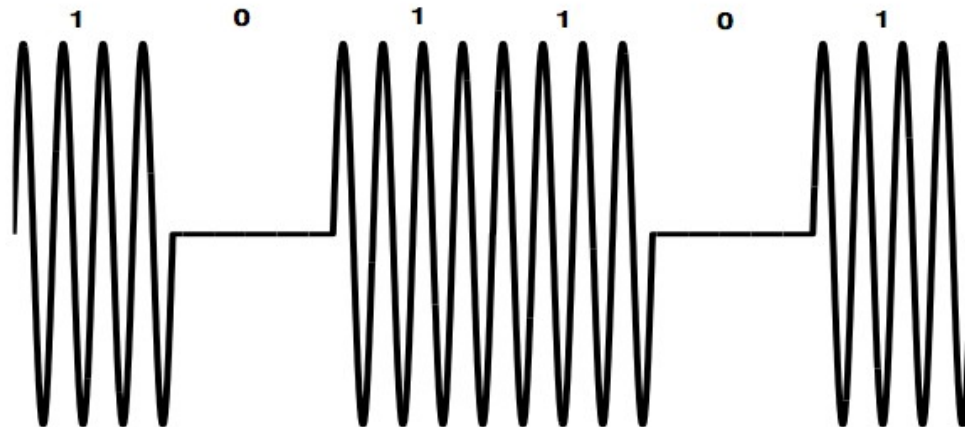
# Bandpass Modulation Techniques

- Amplitude Shift Keying (ASK)
- Phase Shift Keying (PSK)
- Frequency Shift Keying (FSK)
- Multilevel Signaling ( $M_{ary}$  Modulation)
  - $M_{ary}$  Amplitude Modulation
  - $M_{ary}$  Phase Shift Keying ( $M_{ary}$  PSK)
  - $M_{ary}$  Frequency Shift Keying ( $M_{ary}$  FSK)
- Quadrature Amplitude Modulation (QAM)

# Amplitude Shift Keying (ASK)

- In ASK the binary data modulates the amplitude of the carrier signal

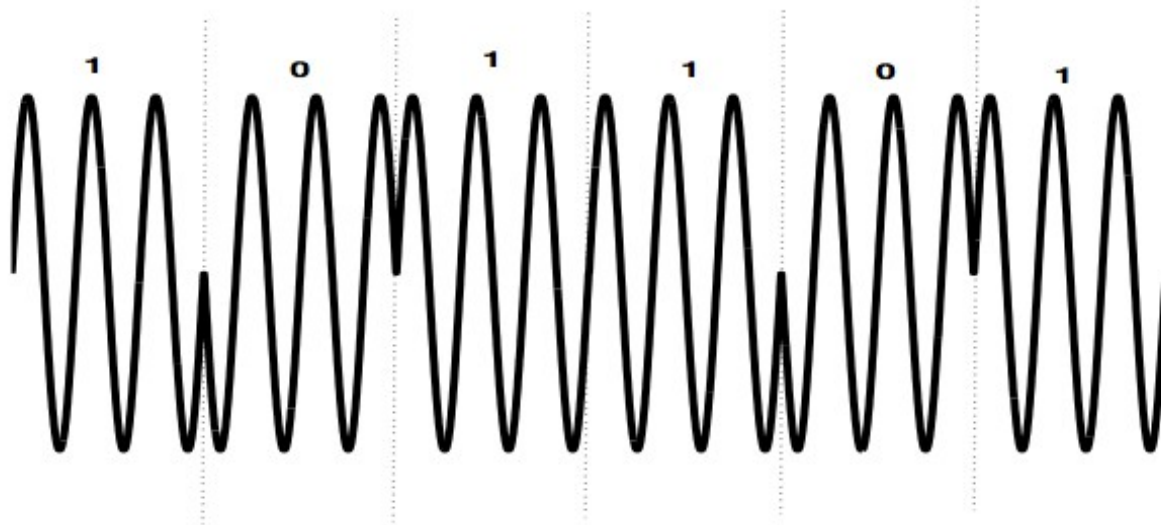
$$v(t) = \begin{cases} A.\cos(\omega_c t) & \text{if binary "1" is transmitted} \\ 0 & \text{if binary "0" is transmitted .} \end{cases}$$



# Phase Shift Keying (PSK)

- In PSK the binary data modulates the phase of the carrier signal

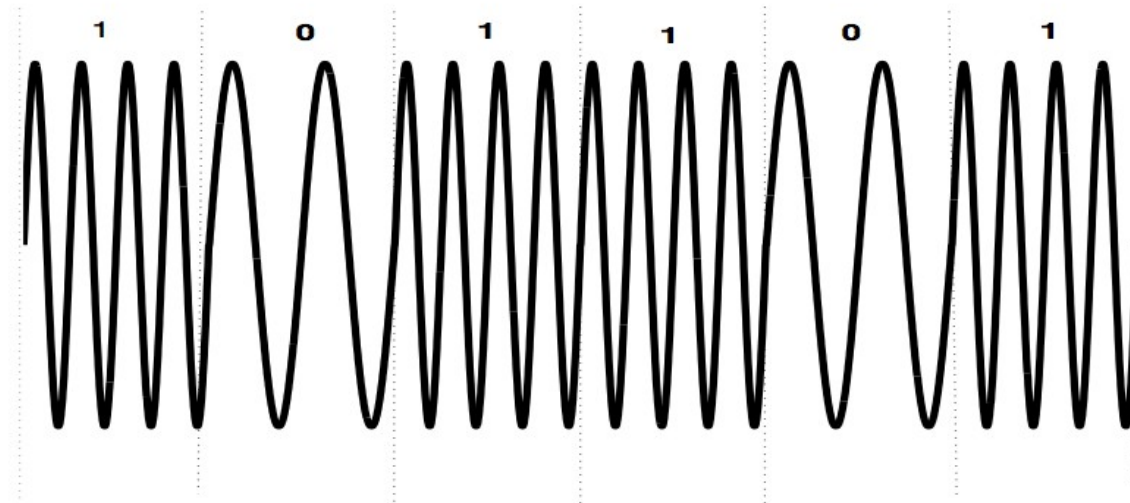
$$v(t) = \begin{cases} A.\cos(\omega_c t + 0) & \text{if binary "1" is transmitted} \\ A.\cos(\omega_c t + \pi) & \text{if binary "0" is transmitted .} \end{cases}$$



# Frequency Shift Keying (FSK)

- In FSK the binary data modulates the frequency of the carrier signal

$$s_i(t) = \begin{cases} A \cos((\omega_o + \Delta\omega)t) & \text{for binary "1"} \\ A \cos((\omega_o - \Delta\omega)t) & \text{for binary "0"}. \end{cases}$$



# Multilevel Signaling

## ( $M_{ary}$ Modulation)

- With multilevel signaling, digital inputs with more than two modulation levels are allowed on the transmitter input.
- The data is transmitted in the form of symbols, each symbol is represented by  $k$  bits
  - $\rightarrow$  We will have  $M=2^k$  different symbol
- There are many different  $M_{ary}$  modulation techniques, some of these techniques modulate one parameter like the amplitude, or phase, or frequency

# $M_{ary}$ Modulation

- Multilevel Signaling ( $M_{ary}$  Modulation)
  - $M_{ary}$  Amplitude Modulation
    - Changing the Amplitude using different levels
  - $M_{ary}$  Phase Shift Keying ( $M_{ary}$  PSK)
    - Changing the phase using different levels
  - $M_{ary}$  Frequency Shift Keying ( $M_{ary}$  FSK)
    - Changing the frequency using different levels

# $M_{ary}$ Amplitude Modulation

- In multi level amplitude modulation the amplitude of the transmitted (carrier) signal takes on  $M$  different levels.
- For a group of  $k$  bits we need  $M = 2^k$  different amplitude levels
- Used in both baseband and bandpass transmission
  - Baseband  $\rightarrow M_{ary}$  Pulse Amplitude Modulation (PAM)
  - Bandpass  $\rightarrow M_{ary}$  Amplitude Shift Keying (ASK)

# M<sub>ary</sub> Amplitude Modulation

- Suppose the maximum allowed value for the voltage is  $A$ , then all  $M$  possible values at baseband are in the range  $[-A, A]$  and they are given by:

$$v_i = \frac{2A}{M-1}i - A \quad ; \text{where } i = 0, 1, \dots, M-1$$

- And the difference between one symbol and another is given by

$$\delta = \frac{2A}{M-1}$$



# Example

- Show how to transmit the message

$m=100110001101010111$

Using  $8_{ary}$  Pulse Amplitude Modulation. Find the corresponding amplitudes of the transmitted signal and calculate the difference between the symbols. Given that the maximum amplitude is 4 Volts

# Example - Solution

- Since we will be using  $8_{ary}$  modulation then the signal must be divided into symbols each of 3 bits
  - Because  $2^3 = 8$
- Therefore

m = 100 110 001 101 010 111

$S_4$   $S_6$   $S_1$   $S_5$   $S_2$   $S_7$

# Example – Solution (Cont.)

- Amplitude calculations

$$v_i = \frac{2A}{M - 1} i - A$$

$$v_4 = \frac{2(4)}{8 - 1}(4) - 4 = 0.5714 \text{ volts}$$

$$v_6 = \frac{2(4)}{8 - 1}(6) - 4 = 2.8571 \text{ volts}$$

$$v_1 = \frac{2(4)}{8 - 1}(1) - 4 = -2.8571 \text{ volts}$$

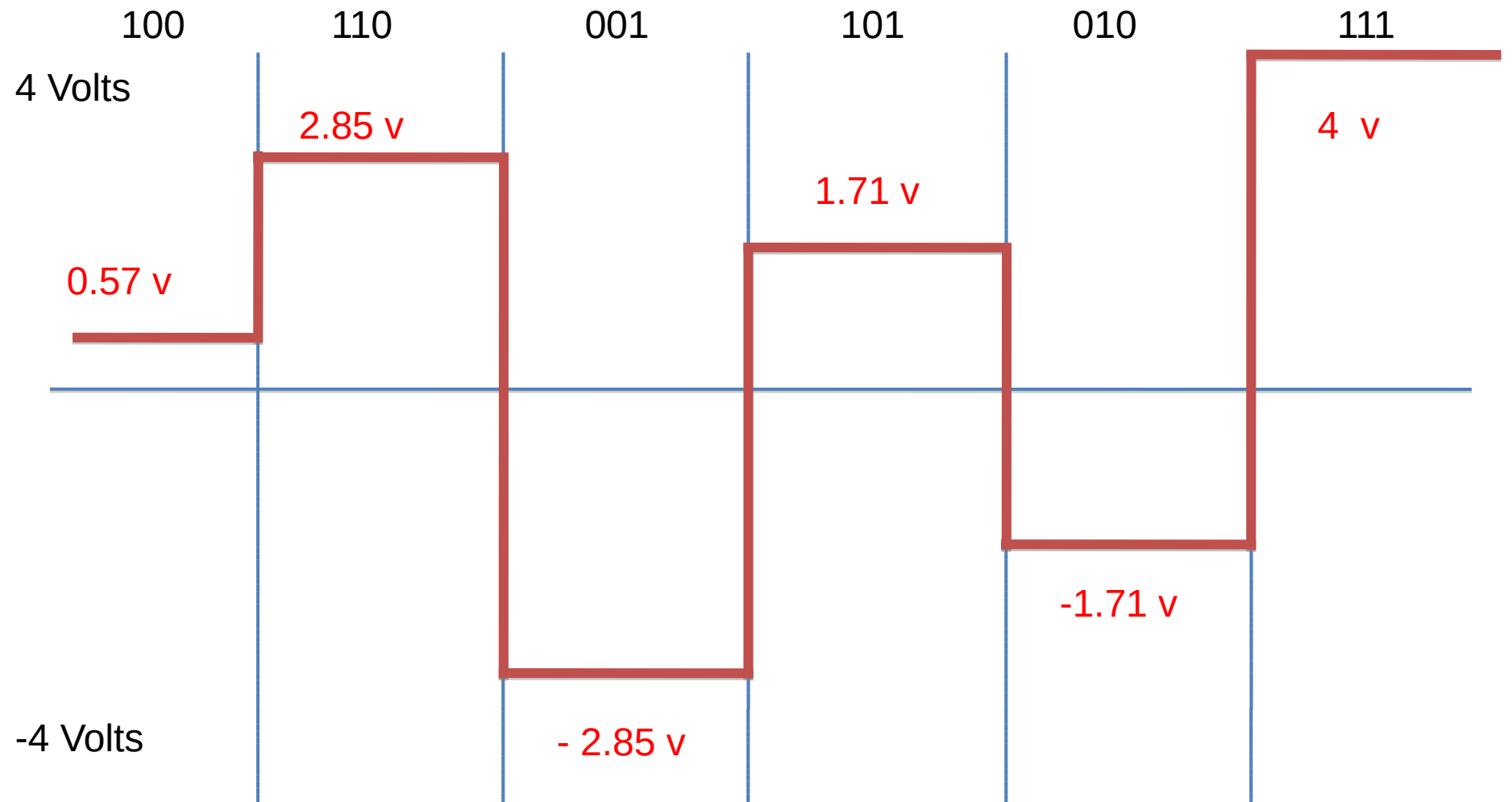
## Example – Solution (Cont.)

$$v_5 = \frac{2(4)}{8 - 1}(5) - 4 = 1.7142 \text{ volts}$$

$$v_2 = \frac{2(4)}{8 - 1}(2) - 4 = -1.7142 \text{ volts}$$

$$v_7 = \frac{2(4)}{8 - 1}(7) - 4 = 4 \text{ volts}$$

# Example – Solution (Cont.)



## Example – Solution (Cont.)

- Difference between each symbol and another can be calculated as follows:

$$\delta = \frac{2A}{M - 1} = \frac{2(4)}{8 - 1} = 1.1428 \text{ volts}$$