COMM

REEM MUIN

POWERUNIT

Data Communications and Networking Fourth Edition

Forouzan

Chapter 1

Introduction

Impact of Data Communications and Networking

- Immediate access to information
- Quicker decision making
- Accuracy
- Efficiency

Data Communication

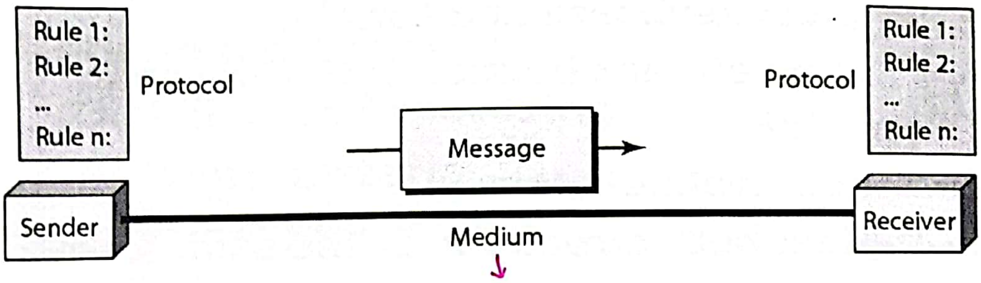
- What is Data?
- What is Communication?
- Communications vs. Telecommunications
- Characteristics of effectiveness in data communication:
  - Delivery: to intended destination
  - Accuracy: immunity to noise and alteration
  - Timeliness: reduced delays (application dependent)
  - Jitter: variation in the delivery time

Concepts of Data Communication

- Components
- Data Representation
- Direction of Data Flow

Figure 1.1 *Five components of data communication*

*مرات ممكن نضحي بال loss Packets عحساب ال delay تكون منيحـــة.



ناخذ فيه ال Bandwidth

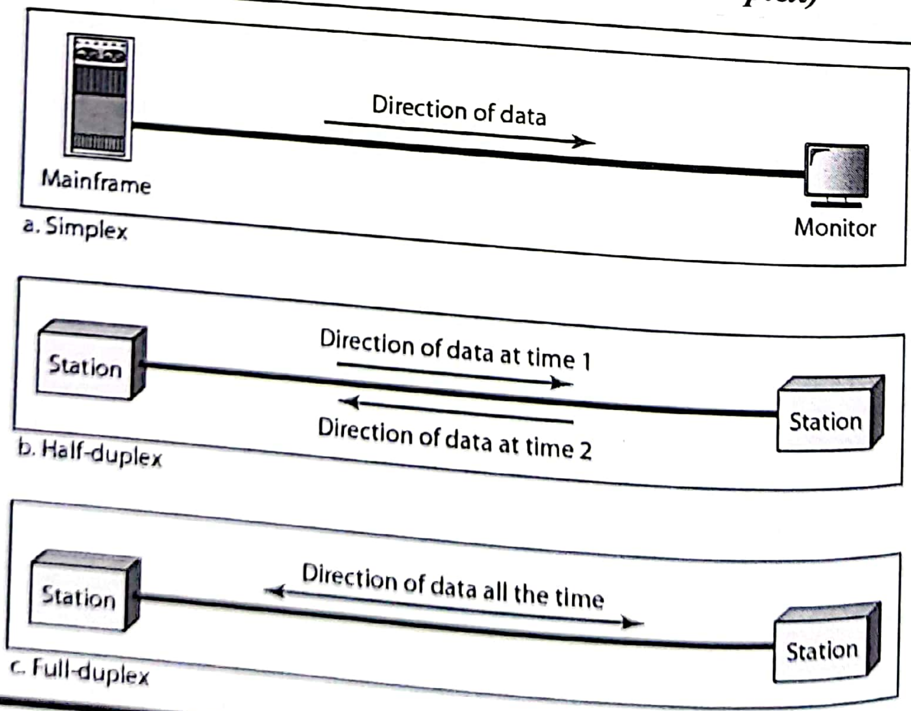
Data Representation

- Text: represented using bit patterns
  - ASCII: 7-bit (128 symbols)
  - Extended ASCII: 8-bit (256 symbols)
  - Unicode: 16-bit (65,536 symbols)
  - ISO: 32-bit (4B symbols)
- Numbers: represented in binary → -5 volt, 5 volt, -12 volt, 12 volt
- Images: bit patterns for pixels and colors
- Audio: analog or digitized form
- Video: analog or digitized form

Types of Data Flow

- **Simplex: one direction**
  - **Single link**
  - **Examples: keyboard and terminal**
- **Half-Duplex: either direction with time sharing**
  - **Single link**
  - **Example: Walkie-Talkie**
- **Full-Duplex: both directions at the same time**
  - **Single link with shared bandwidth or dual lin**
  - **Example: telephone network**

Figure 1.2 *Data flow (simplex, half-duplex, and full-duplex)*



Networks

- What is a network?
  - Set of devices or nodes connected via communication links
- Networking Concepts:
  - Distributed Processing
  - Network Criteria
  - Physical Structures
  - Network Models
  - Categories of Networks
  - Interconnection of Networks: Internetwork

Distributed Processing

- Decentralized processing
- Tasks are processed by several nodes
- Example: Web pages are centrally and remotely stored but locally processed
- Overhead of distributed processing

أصغر Network جغرافية ال PAN وأكبر وحدة WAN

↳ Personal Area Network ده

Wide Area Network ده

Network Criteria

- **Performance**
  - Bit rate vs. packet rate vs. throughput
  - Transit time (or propagation delay)
  - Response time (round trip delay)
- **Reliability**
  - Bit error rate vs. packet error rate
  - Error detection and correction
  - Link failure recovery time
  - Robustness to disasters
- **Security**
  - Protect against unauthorized access
  - Protect against alteration
  - Validity of sender

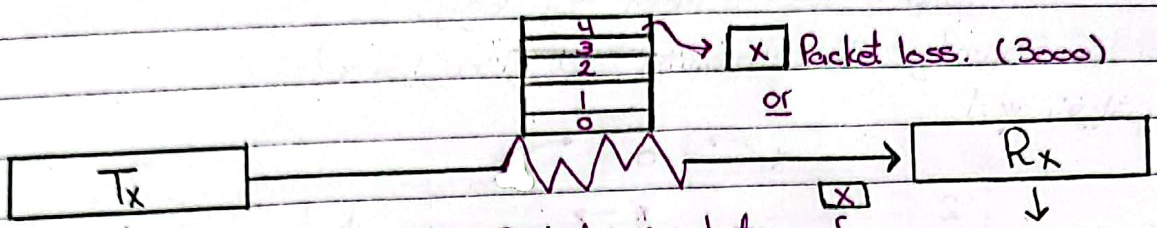
Types of Network Connections

- **Point-to-point link**
  - Dedicated link between two nodes
  - Full link capacity
- **Multi-point link**
  - Shared link among many nodes
  - Spatially shared: simultaneous access
  - Temporally shared: timely shared

كل ما زاد عدد الـ Users بزيد الشلعش بيزحم عال link
وبالتالي ممكن تصير مشاكل.

11/10/2022

* الفرق بين ال Rate وال Throughput *



Packet error data ← data ما وصلت بشكل صح

Sending Rate = 10,000 Packets Per Second

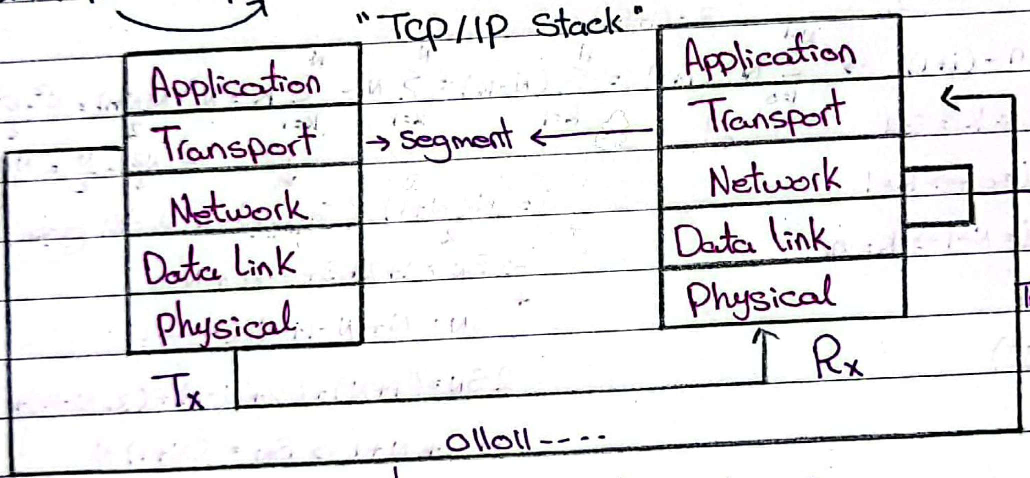
Throughput = 10,000 ← هو اللي يهمني وهو المقياس. Packets per second

كمية اللي حيوصلك، ممكن يقل عن 10,000 بسبب ال loss.

حيوصل 7,000 عشان راحوا 3,000.

* from Mpbs to Packet Per Second → بنضرب بعدد ال Packets

"Tcp/IP Stack"



هون بنكتشف ال loss فما بوصل ال Transport layer.

هون بنقاس ال Throughput

* 1983 → End-to-End Argument → good research paper. (يحتمل رأيين)

اكتشاف ال loss بال App لأنه بضمن يكون كل ال Packets وصلوا 100% لكن بخسر الوقت (لذلك الرأي الأول أفضل)

اكتشاف ال loss بين ال Physical وال data link layer لأنه أسرع لاكتشاف الخطأ.

* HDLC → like Tcp Model → High-level Digital Link Control 1979. (Data link layer)

* ال Hub انقرض صار يستخدم ال Swich حاليا (لكن ميزة ال Hub إنه أقل تكلفة).

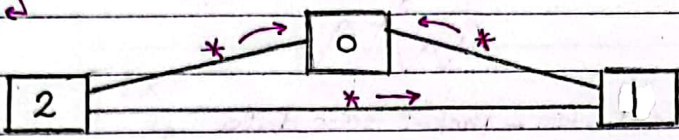
* بال Tcp/IP stack احنا حنشتغل بالمادة بجزء ال physical and data link layer.

* Quiz Idea:

A fully connected mesh topology.

خيار ممتاز للـ reliability والـ Security لو الـ Cost ما بتهمني.

لإنه ما في اشي Shared



$i = 0, 1, 2$ / L_i = Number of links on the i node / L = total Number of links / N = total Number of Nodes

i	N	L_i
0	3-1	2
1	3-(1+1)	1
2	3-(1+1+1)	0

→ الـ Links المميزين اللي مش مرتبطين بـ Nodes ثانية.

$$L = \sum_{i=0}^{N-1} L_i = \sum_{i=0}^{N-1} n-(i+1) \rightarrow \sum_{i=0}^{N-1} N-(i+1) = \sum_{k=1}^{N} (N-k) = \sum_{k=1}^{N} N - \sum_{k=1}^{N} K = N^2 - \frac{N(N+1)}{2} = N^2 - \frac{N^2}{2} - \frac{N}{2} = \frac{N^2}{2} - \frac{N}{2} = \frac{N(N-1)}{2} \#$$

$\therefore K = i+1$

$i = 0 \rightarrow k = 1$

$i = N-1 \rightarrow k = N$

$= \frac{N(N+1)}{2}$ (مجموع الأعداد الطبيعية)

$\rightarrow S_N = 1+2+3+ \cdots +N$

$+ \quad S_N = N + N-1 + N-2 + \cdots 1$

$2S_N = (1+N) + (2+N-1) + (3+N-2) + \cdots + N+1 \rightarrow S_N = \frac{(N+1)N}{2}$

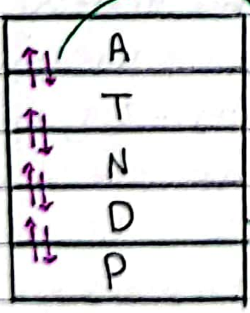
* if N is large:

$\approx \frac{N^2}{2} \rightarrow O(N^2)$

13/10/2022

https://sites.google.com/view/datacomm2022/ → موقع سلايدات المادة.

* 5 layer model (TCP/IP) Protocol Stack:



" Software of hardware تفاعل"

Ethernet

→ Protocols example: IEEE 802.3 / IEEE 802.11 → wifi / IEEE 802.5 → bluetooth

→ Protocol example: line coding

* Standards protocols: 1- IEEE 2- ITU-T

* Request for Comment (RFC): بحطوا الناس فيها تعليقات عشان من بعده يتحول الاشي لـ Standard أو يتبدل باشي سابق (بيستخدم لتطوير الـ Standard)

Five Apple

Figure 1.3 *Types of connections: point-to-point and multipoint*

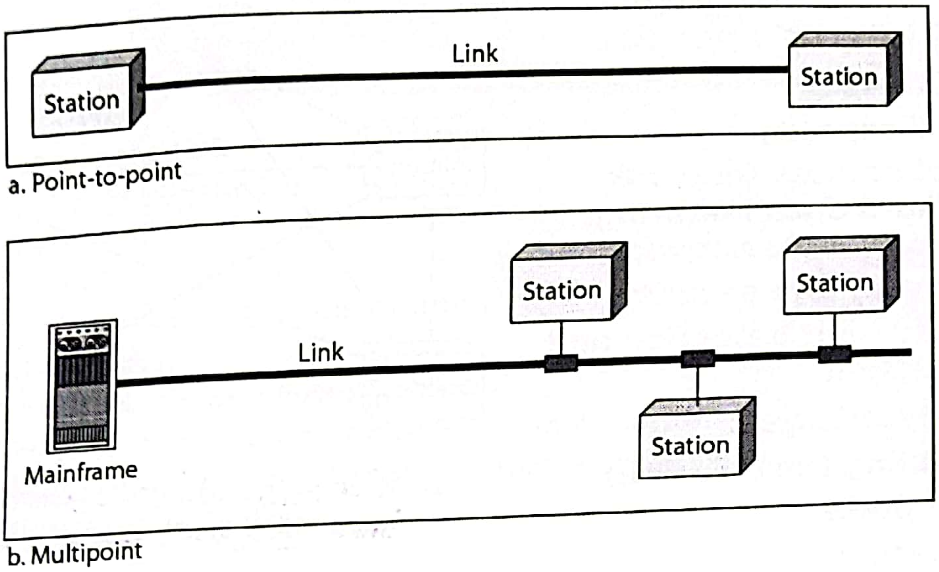


Figure 1.4 *Categories of topology*

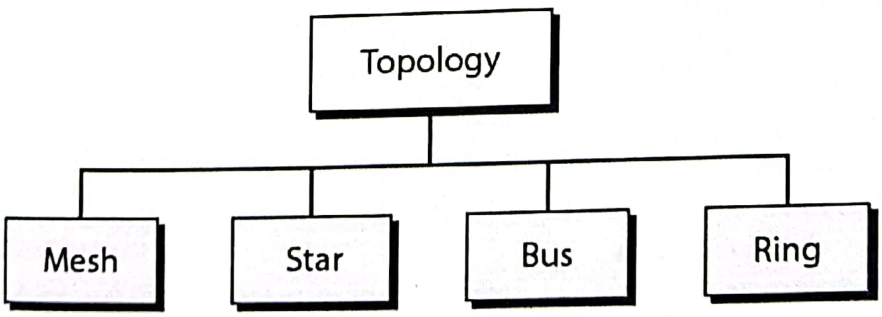
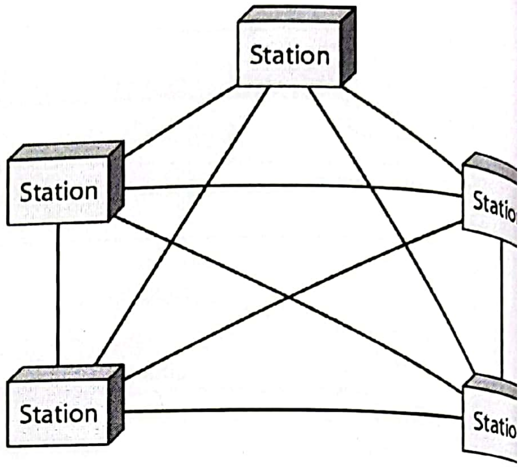


Figure 1.5 *A fully connected mesh topology (five devices)*

- **Point-to-point**
- **n(n-1)/2 dedicated links**
- **(n-1) I/O ports per device**
- **Advantages:**
  - **Full capacity**
  - **Robustness (one link failure does not bring down whole network)**
  - **Privacy and security**
  - **Easy fault detection and isolation**
- **Disadvantages:**
  - **Cabling (installation)**
  - **I/O ports**
  - **Space**
  - **Cost**



* كلو مشبوك مع كل ال stations
* مش دايما منيح لإنه في كتير links

Figure 1.6 *A star topology connecting four stations*

- **Point-to-point to the hub**
- **Indirect connections among the nodes**
- **n dedicated links to a hub**
- **One I/O port per device**
- **Advantages:**
  - **Robustness**
  - **Easy fault detection and isolation**
- **Disadvantages:**
  - **Single point of failure at the hub**
  - **Bottleneck at the hub**
    - **store-and-forward (in case of a switch)**

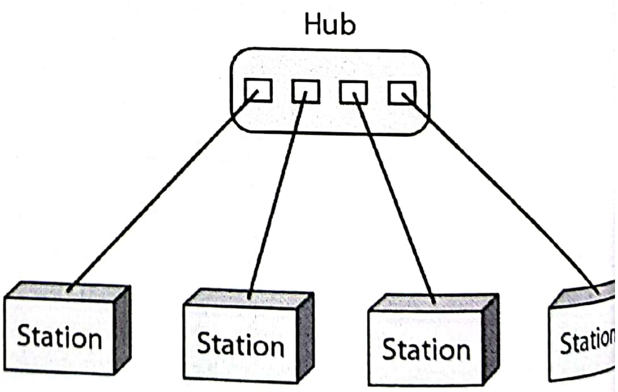
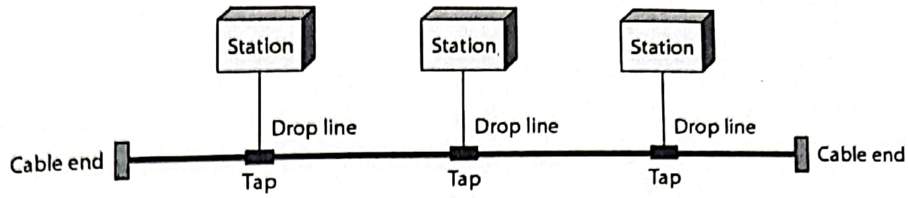


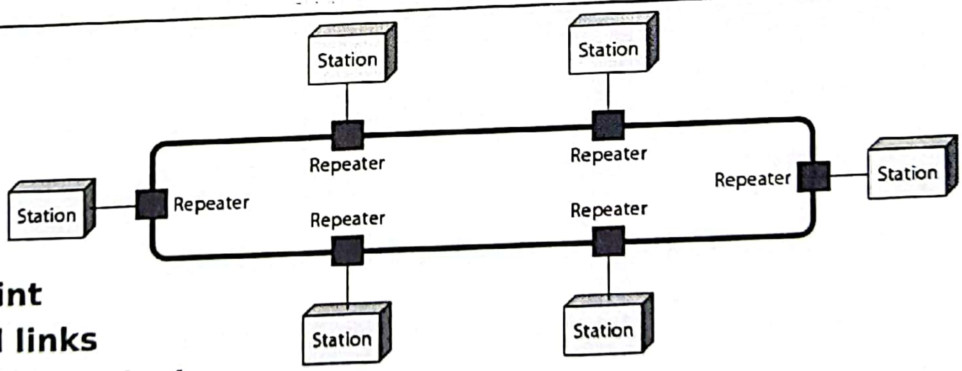
Figure 1.7 *A bus topology connecting three stations*



- Multi-point
- One backbone link
- One I/O port per device
- Advantages:
  - Ease of installation
  - Minimum cabling at installation
- Disadvantages:
  - Limited taps and distance to node
  - Limited node addition after initial installation
  - Limited quality due to signal reflection at taps
  - Difficult fault isolation

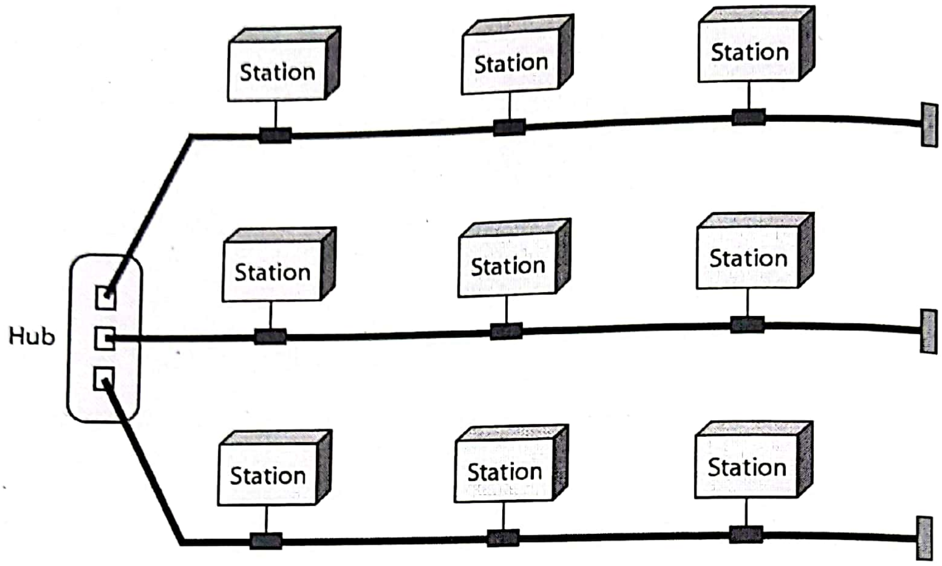
سيئ من ناحية ال reliability
والPerformance
وجيد من ناحية ال Cost.
هذا لل Star topology
(Hub)
(ال Slide الي قبل)

Figure 1.8 *A ring topology connecting six stations*



- Point-to-point
- n dedicated links
- Two I/O ports per device
- Advantages:
  - Easy installation
  - Easy fault detection and isolation
- Disadvantages:
  - Speed: store-and-forward at each node
  - Ring length and number of nodes

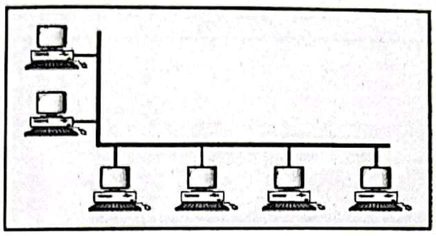
Figure 1.9 *A hybrid topology: a star backbone with three bus networks*



Categories of Networks

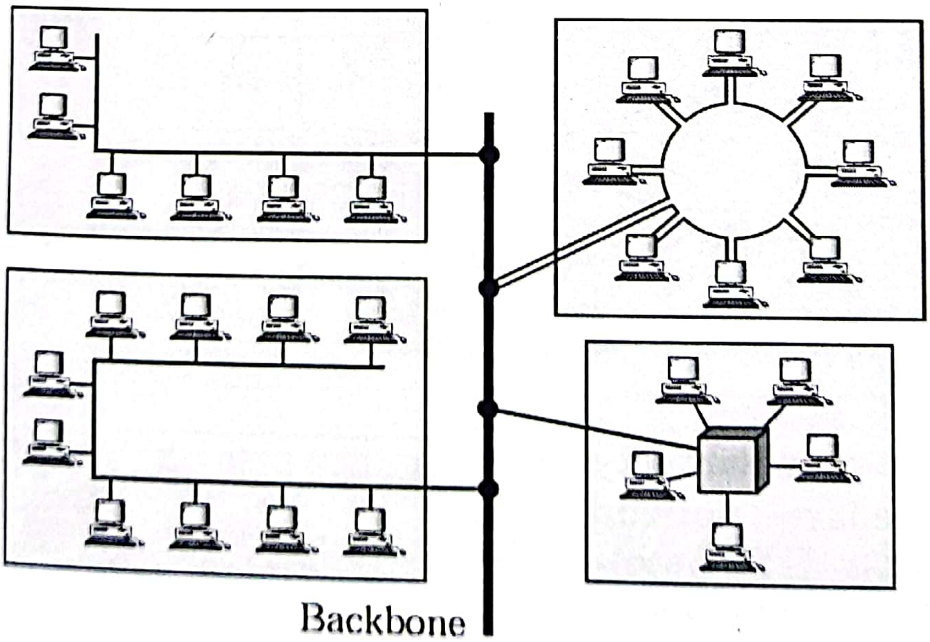
- Personal Area Network (PAN)
- Local Area Network (LAN)
- Metropolitan Area Network (MAN)
- Wide Area Network (WAN)

كمستوى المدينة



a. Single-building LAN

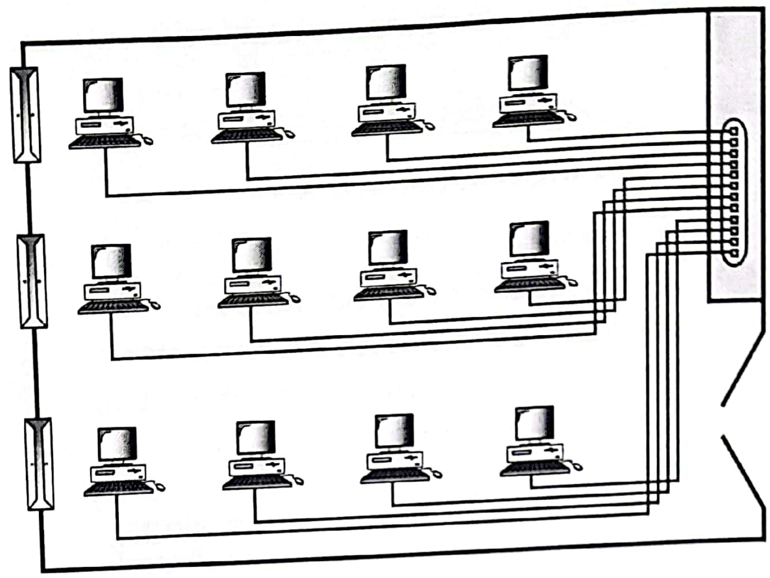
- Privately owned
- Single office, building, or campus
- Used for shared resources
  - Hardware (e.g. printer, scanner, etc)
  - Software (e.g. engineering application)
  - Data (e.g. specifications, drawings, etc.)
- Data rates up to gigabits per second
- Size is limited to a few kilometers



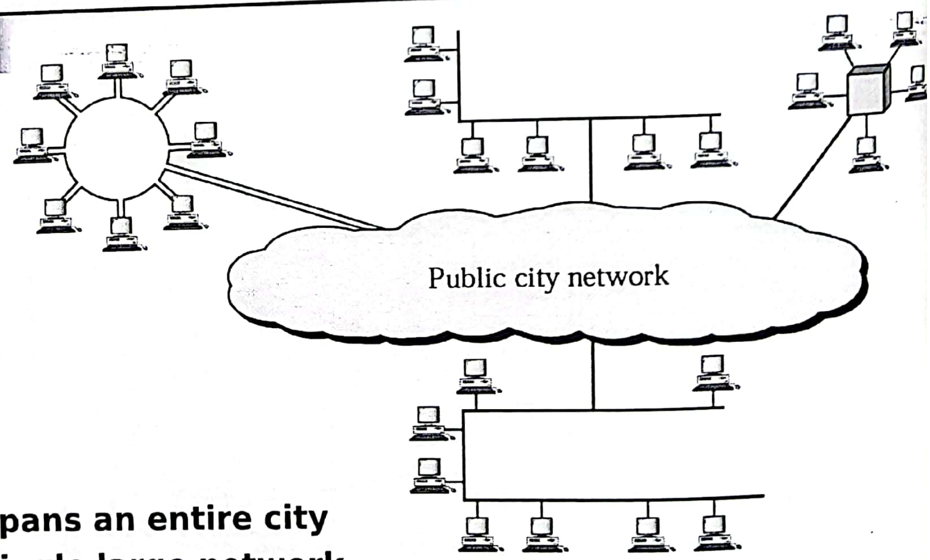
b. Multiple-building LAN

McGraw-Hill
©The McGraw-Hill Companies, Inc

Figure 1.10 *An isolated LAN connecting 12 computers to a hub in a closet*

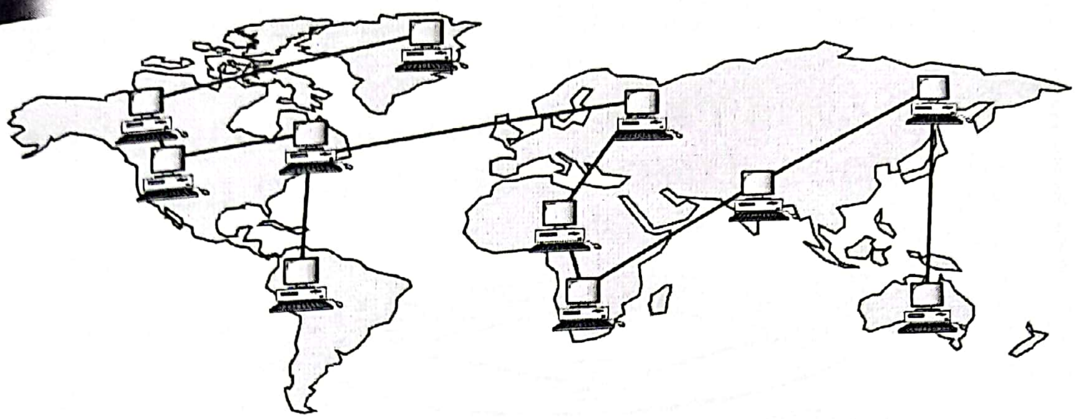


MAN



- **Spans an entire city**
- **Single large network**
  - **Cable TV (private)**
  - **Telephone (public)**
- **LAN-to-LAN connection**

Figure 1.15 *WAN*



- May spans the entire globe
- Long distance transmission
- May use public, leased, or private networks and equipment
- The concept of an enterprise network

Figure 1.11 *WANs: a switched WAN and a point-to-point WAN*

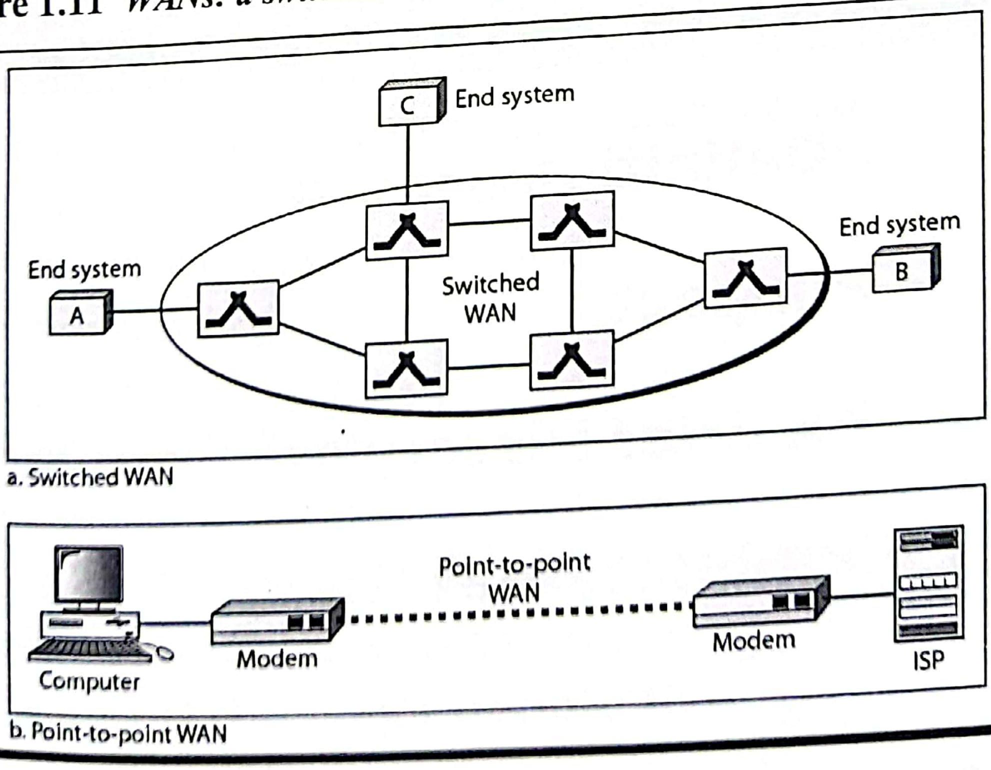
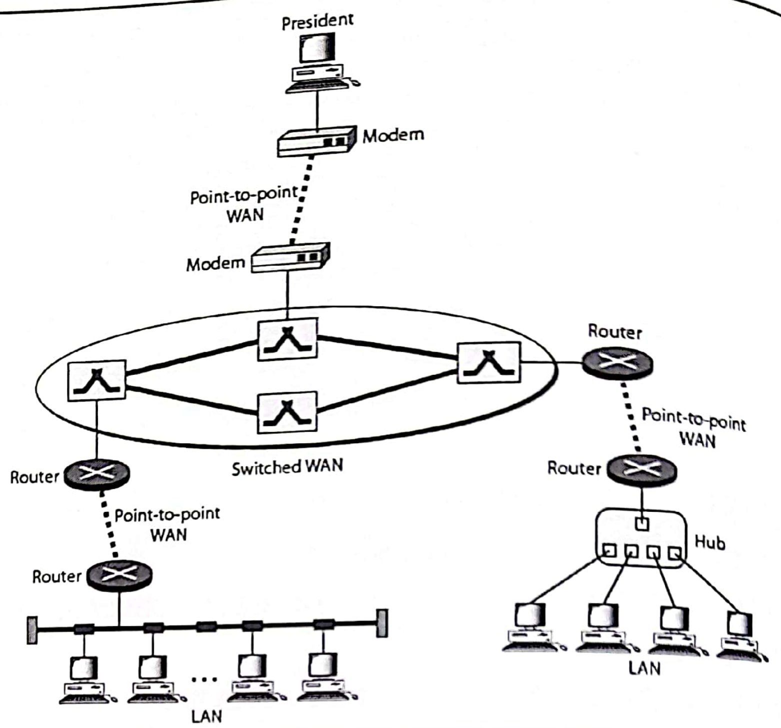


Figure 1.12 *A heterogeneous network made of four WANs and two LANs*



The Internet

- ➢Definition
- ➢Impact
- ➢A Brief History
- ➢The Internet Today

Definition

* Important

- An internet: two or more networks that can communicate with each other
- The Internet: the network of all networks

Impact of the Internet

- Revolution of communications and information exchange
  - eCommerce
  - eLearning
  - eMail
  - eGovernment
  - etc.
- Created a very well-connected world
- Near real-time access to resources
- Changed the ways people live

History of the Internet

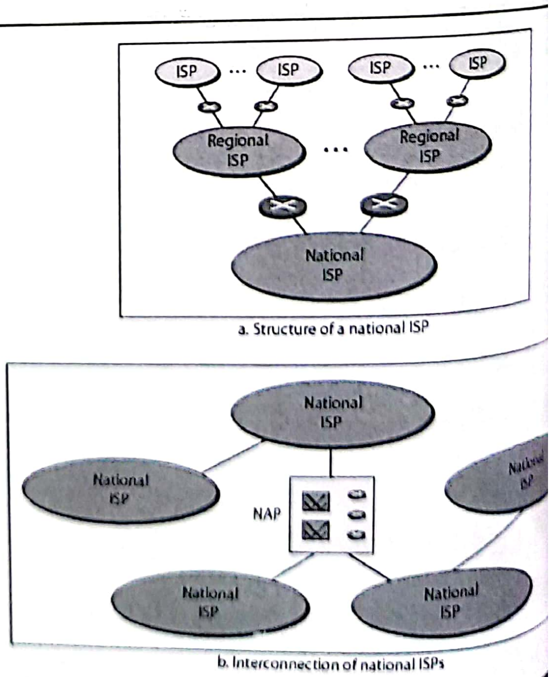
- Mid 1960's: ARPA (Advanced Research Projects Agency) of DOD (Department of Defense)
  - ARPANET was initiated in 1967 and done in 1969
  - Interface Message Processors
  - Network Control Protocol
  - 4-node internetwork
- 1972: Internetting Project
- 1973: TCP (Transmission Control Protocol)
  - End-to-end packet delivery
  - Encapsulation, datagram, segmentation, reassembly, error detection, and gateway functionality
- 1973: IP (Internetworking Protocol)
  - Datagram routing

ومع وجود الـ IoT بطل في إقبال كثير عالـ TCP لأنه بياخذ كثير وقت

* الـ Megers Performance ثابتة اللي بتغير هو الـ Technology.

Figure 1.13 *Hierarchical organization of the Internet*

- All types of networks
- Connected by switching devices
- Continuously changing
- Run by private companies
- Local Internet Service Provider (ISP): direct service to end user
- Regional ISP: provide service to ISP's
- National Service Provider (NSP): connecting private backbone networks
- Network Access Point (NAP): connects NSP's



Protocols and Standards

- Protocols
- Standards → الجميع لازم يلتزم فيها
- Standards Organizations
- Internet Standards

Definitions

- A Protocol: a set of rules that governs data communications
  - Syntax: the structure or format of the data (e.g. how many bits for each section)
  - Semantics: the interpretation of the data bit patterns ↳ كيف حتقسم الdata
  - Timing: when and how fast the data should be sent
- A Standard: a set of agreed-upon rules that govern data communications
  - Interoperability
  - Open market for competition
  - Guidelines to manufactures, vendors, government agencies, and service providers

Standard Organizations

- Standards Creating Committees:
  - International Organization for Standardization (ISO)
  - International Telecommunication Union – Telecommunications Standards (ITU-T)
  - American National Standards Institute (ANSI)
  - Institute of Electrical and Electronic Engineers (IEEE)
  - Electronic Industry Association (EIA)
- Forums (Special Interest Groups)
- Regulatory Agencies (e.g. FCC)
- Internet Standards:
  - Internet Draft: working document for 6 months
  - Request for Comment (RFC): recommended and published draft

Chapter 3

Data and Signals

" Physical Layer "

Note

To be transmitted, data must be transformed to electromagnetic signals.

3-1 ANALOG AND DIGITAL

Data can be analog or digital. The term analog data refers to information that is continuous; digital data refers to information that has discrete states. Analog data take on continuous values. Digital data take on discrete values.

منفصلة.

Topics discussed in this section:

Analog and Digital Data
Analog and Digital Signals
Periodic and Nonperiodic Signals

*الصوت بتحول ل electromagnetic Signal عن طريق إنه صوتك بتحول ل Vibration بال Voltage وبتمثل ك Signal .

Note

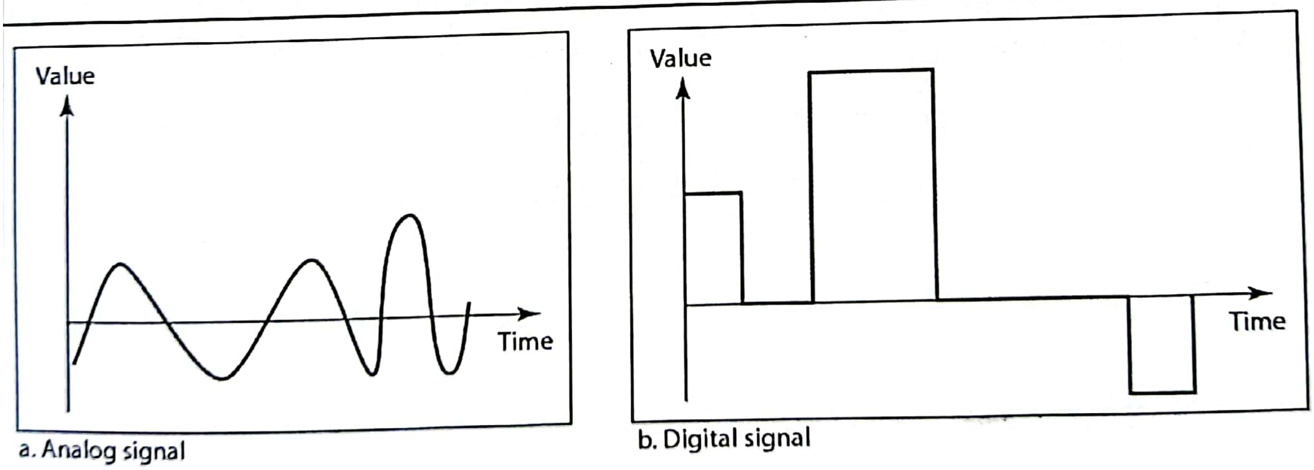
Data can be analog or digital.
Analog data are continuous and take continuous values.
Digital data have discrete states and take discrete values.

Note

Signals can be analog or digital. Analog signals can have an infinite number of values in a range; digital signals can have only a limited number of values.

3.5

Figure 3.1 *Comparison of analog and digital signals*



- Continuous
- Infinite number of possible values

- Discrete
- Finite number of possible values

3.6

Periodic and Aperiodic Signals

لازم أقدر أقيسه ما يكون ∞ وإلا بكون كل ال Signals من نوع واحد.

- Periodic signal → cycle معينة بتعيد حالها أكثر من مرة
  measurable
  - Completes a pattern within a certain time frame called a period
  - Repeats the pattern over subsequent identical periods called cycles
- Aperiodic signal → ما بتعيد حالها
  - Changes with no patterns or cycles over time
  - Both analog and digital signals can be either periodic or aperiodic

Note

In data communications, we commonly use periodic analog signals (for less bandwidth use) and nonperiodic digital signals (for representing the variation in data).

لإنه ال Bandwidth تاعها محدود.

لنمثل levels أكثر من ال Data

3.7

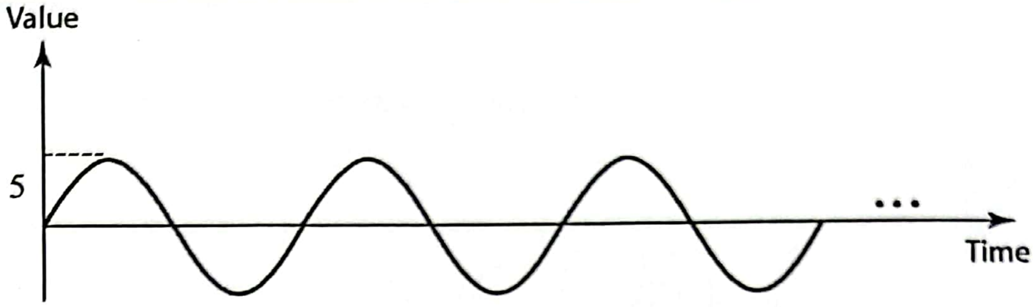
3-2 PERIODIC ANALOG SIGNALS

Periodic analog signals can be classified as simple or composite. A simple periodic analog signal, a sine wave, cannot be decomposed into simpler signals. A composite periodic analog signal is composed of multiple sine waves.

Topics discussed in this section:

Sine Wave
Wavelength
Time and Frequency Domain
Composite Signals
Bandwidth

Figure 3.2 *A sine wave*



$$s(t) = A \sin(2\pi f t + \varphi)$$

Instantaneous Amplitude

Peak Amplitude

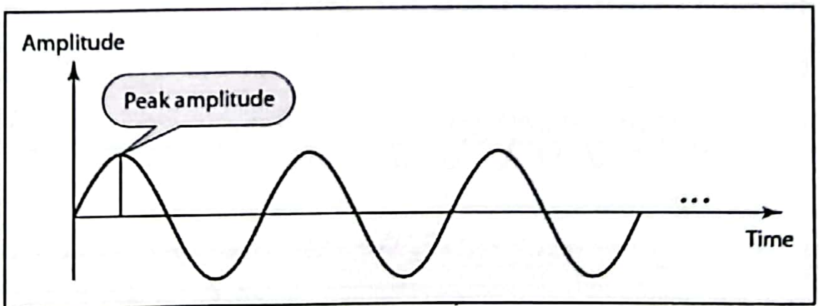
Frequency علاقة عكسية مع ال Period

Phase

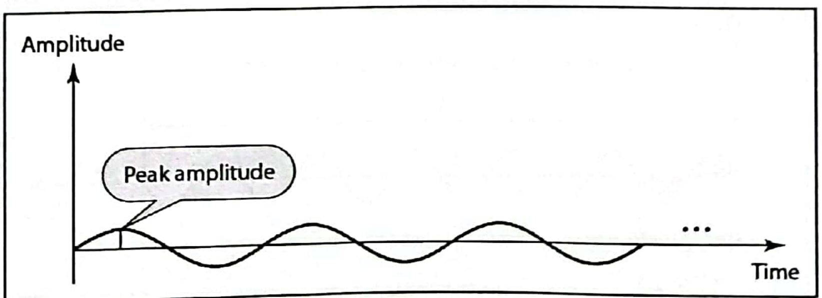
Characteristics of the sine wave

.9

Figure 3.3 *Two signals with the same phase and frequency, but different amplitudes*



a. A signal with high peak amplitude



b. A signal with low peak amplitude

3.10

Note

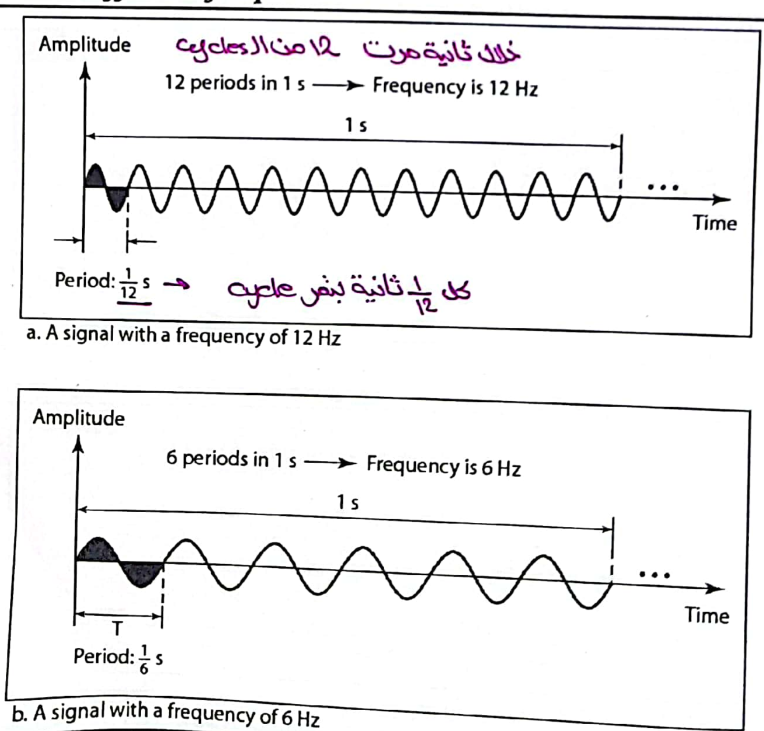
Frequency and period are the inverse (or the reciprocals) of each other.

$$\boxed{f = \frac{1}{T} \qquad \text{and} \qquad T = \frac{1}{f}}$$

Hz Second

3.11

Figure 3.4 *Two signals with the same amplitude and phase, but different frequencies*



3.12

Table 3.1 *Units of period and frequency*

Unit	*Equivalent*	*Unit*	*Equivalent*
Seconds (s)	1 s	Hertz (Hz)	1 Hz
Milliseconds (ms)	10^{-3} s	Kilohertz (kHz)	10^3 Hz
Microseconds (μs)	10^{-6} s	Megahertz (MHz)	10^6 Hz
Nanoseconds (ns)	10^{-9} s	Gigahertz (GHz)	10^9 Hz
Picoseconds (ps)	10^{-12} s	Terahertz (THz)	10^{12} Hz

3.13

Example 3.3

← بالأردن = 50 Hz (نظام أوروبي)

The power we use at home (in the USA) has a frequency of 60 Hz. The period of this sine wave can be determined as follows:

$$\boxed{T = \frac{1}{f} = \frac{1}{60} = 0.0166 \text{ s} = 0.0166 \times 10^3 \text{ ms} = 16.6 \text{ ms}}$$

3.14

Example 3.4

Express a period of 100 ms in microseconds.

Solution

From Table 3.1 we find the equivalents of 1 ms (1 ms is 10^{-3} s) and 1 s (1 s is 10^6 µs). We make the following substitutions:.

$$100 \text{ ms} = 100 \times 10^{-3} \text{ s} = 100 \times 10^{-3} \times 10^{6} \text{ µs} = 10^{2} \times 10^{-3} \times 10^{6} \text{ µs} = 10^{5} \text{ µs}$$

3.15

Example 3.5

The period of a signal is 100 ms. What is its frequency in kilohertz?

Solution

First we change 100 ms to seconds, and then we calculate the frequency from the period (1 Hz = 10^{-3} kHz).

$$100 \text{ ms} = 100 \times 10^{-3} \text{ s} = 10^{-1} \text{ s}$$

$$f = \frac{1}{T} = \frac{1}{10^{-1}} \text{ Hz} = 10 \text{ Hz} = 10 \times 10^{-3} \text{ kHz} = 10^{-2} \text{ kHz}$$

3.16

Note

Frequency is the rate of change with respect to time.

كم cycle بمر عندك every Time.

Change in a short span of time means high frequency.

Change over a long span of time means low frequency.

Note

If a signal does not change at all, its frequency is zero (DC Signal).
If a signal changes instantaneously, its frequency is infinite (only in theory). نظريا موجودة

* Chapter 2 → Self Reading.

* Chapter 3 → Data and Signals.

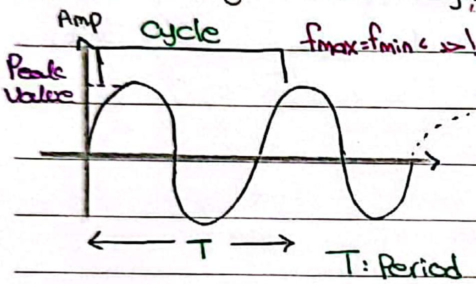
* 3.3

Data → تحمل معلومات
- Analog → EX: Voice (continuous data, Infinite number of levels)
- Digital → EX: Numbers (discrete data, limited number of levels)

Signals → أشارة
- Analog
- Digital

* Aperiodic (باللغة اللاتينية) = Non periodic.

* Periodic Signals (analog):



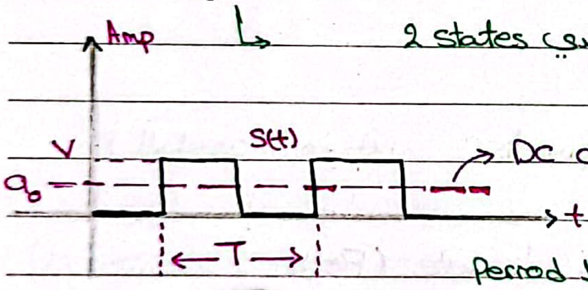
T: Period

$f_{max}=f_{min}$ ، لأنه فيها بس تردد واحد

Bandwidth = 0 → cycle every T يمشي

* كل الـ Simple periodic Signals بكون الـ bandwidth تاعهم صفر لأنه فيها تردد واحد بس. ($f_{max} = f_{min}$)

* Periodic digital Signal: ↳ لأنه عنده 2 states



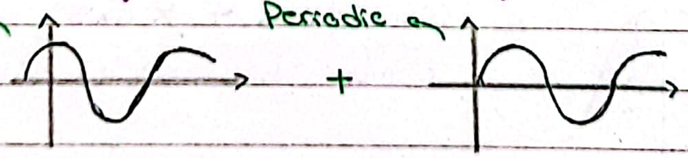
→ DC component (ممكن يكون = 0) (هون هو صفر.)

* Bandwidth = $\infty - 0 = \infty$

نفس الـ Period

$$S(t) = a_0 + \underbrace{\sum_{n=1}^{\infty} a_n \cos\left(nt \frac{\pi}{T/2}\right)}_{} + \underbrace{\sum_{n=1}^{\infty} b_n \sin\left(nt \frac{\pi}{T/2}\right)}_{}$$ (للفهم والمعرفة)

Periodic → Periodic →

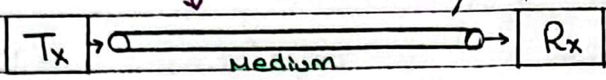


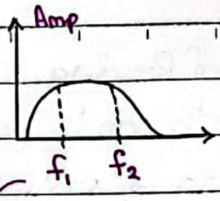
] → harmonics → أكثر من Frequency

* Frequency = $\frac{1}{T}$.

مش ضروري أمرر كل الـ harmonics

* Signal (إشارة بيبعتها بال Medium)





$B = f_{max} - f_{min}$ $\qquad$ $B = f_2 - f_1$

↳ Bandwidth = بنمرر عدد معين من الـ frequencies $\qquad$ مش كل الـ frequency بتوصل لأنه

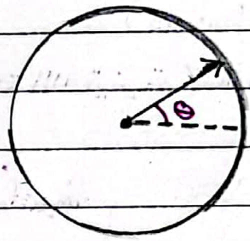
مش عدد لامحدود. في الـ Medium وفي الـ Signal $\qquad$ فيه Filters.

* ① Angular frequency : (ω) (radian frequency)

$S(t) = A \sin(\underbrace{2\pi f t}_{} + \underbrace{\psi}_{}) = A \sin(\omega t + \psi) = A \sin[\omega(t - \overbrace{\tfrac{\psi}{\omega}}^{\text{time (s)}})] = A \sin[\omega(t - t_p)]$

جواهم لازم يكون إما degrees أو Rad $\qquad$ degrees or Rad $\qquad$ ↳ degrees or rad

$t_p = \frac{\psi}{\omega}$ time Shift



$\omega = 2\pi f = \frac{2\pi}{T}$ $\frac{rad}{sec}$ $\qquad$ * $\underbrace{2\pi f}_{\omega} t = \underbrace{\omega t}_{\text{Rad}}$

بلف دورة كاملة every T.

$\therefore \frac{deg. / rad.}{Sec}$ = frequency (time (seconds))

② $f = \frac{1}{T}$, frequency $(\frac{cycles}{sec.})$ (Hz)

16/10/2022

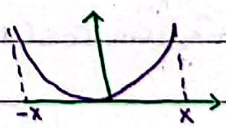
* from radians to degrees : * $\frac{\pi}{180}$ = * $\frac{2\pi}{360}$

$\therefore$ if $\psi = \omega t_p = \frac{2\pi}{T}(\frac{T}{4}) = \frac{\pi}{2}$ (للمراجعة الذهاب إلى Appendix C)

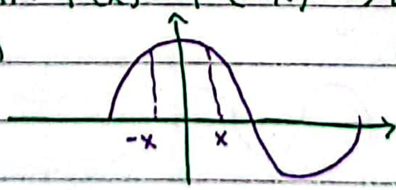
* if Signal is Periodic → fourier Series → discrete (Frequency component)

* if Signal is Aperiodic → fourier transform → Continuous (Frequency Component)

* even function : $f(x) = f(-x)$ → Ex : $f(x) = x^2$



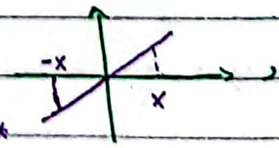
, $f(x) = \cos(x)$



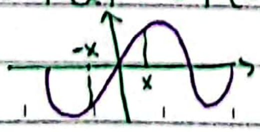
(Symmetrical of y axis)

* تكامل الـ even بيعطيني Numbers

* odd function : $f(x) = -f(-x)$ → Ex : $f(x) = x$



$f(x) = \sin(x)$



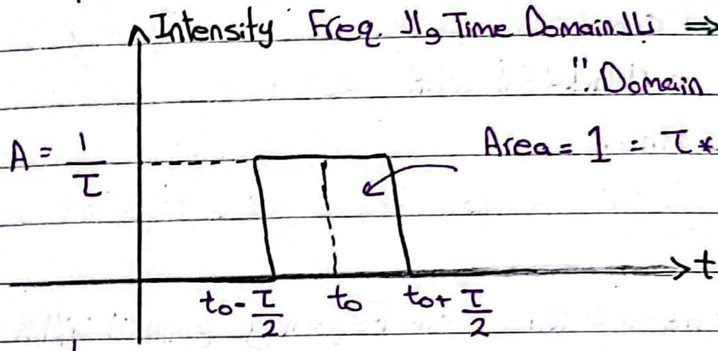
* تكامل الـ odd بيعطيني صفر.

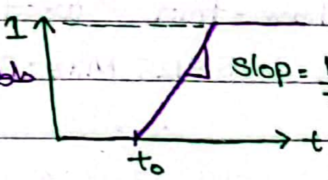
* Example: → شو بصير لو T راحت إلى صفر

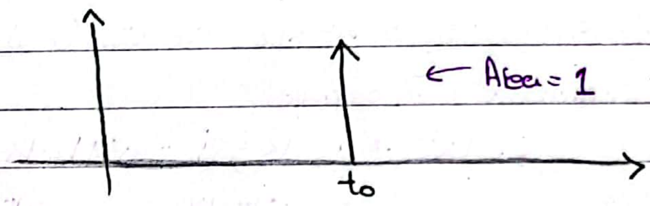
بال Time Domain والـ Freq. Domain!! $\Rightarrow \int \frac{1}{\tau}\, dt = \frac{t}{\tau} \Rightarrow$

$A = \frac{1}{\tau}$

Area $= 1 = \tau * \frac{1}{\tau} =$ طول × عرض

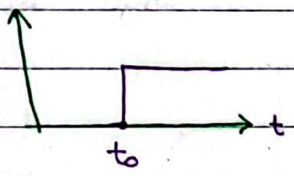






← Area = 1

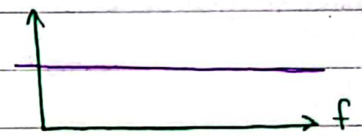
$\Rightarrow$



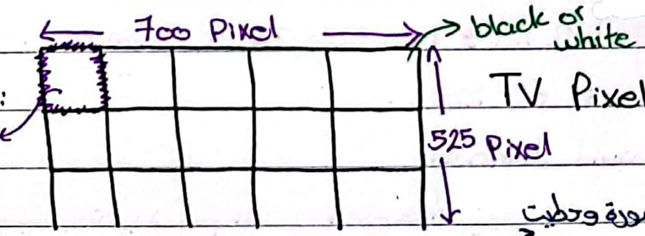
$u(t-t_0) \rightarrow$ unit step function.

$\delta(t-t_0) \rightarrow$ Impulse function.

* $\lim_{\tau \to 0} \frac{\sin \pi \tau f}{\pi \tau f} = 1$



* Example 3.15 (3.43):



TV Pixels (Screen)

$\frac{1}{12}$ من الثانية لو شلت صورة وحطيت بدالها بيكون ما نلاحظ هذا التغير

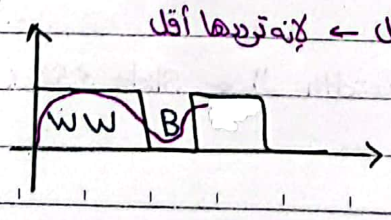
Sol. $525 \times 700 = 367500$ Pixels

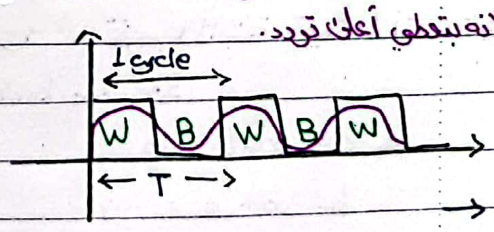
$\rightarrow \frac{367500 \text{ Pixels}}{\text{Screen}} * \frac{30 \text{ Screen}}{s}$ because: $\frac{1}{12}$ sec $\rightarrow 12$ Hz $\rightarrow 2*12$ Hz $= 24 \approx 30$

$= 11{,}025{,}000 \ \frac{\text{Pixels}}{s}$

* أسوأ احتمال للـ frequency هو إنه تكون الـ Signal عبارة عن Black white Black ...
لإنه بتعطي أعلى تردد.

* هيك مثلاً أفضل ← لإنه ترددها أقل





18/10/2022

$11{,}025{,}000 \; \frac{\cancel{Pixels}}{s} * \frac{1 \text{ cycles}}{2 \; \cancel{Pixels}} = 5{,}512{,}500 \; \frac{\text{cycles}}{s} \; (Hz) = 5.5125 \; MHz$

Multiple Not Convolution → DC

Frequency

(frequency).

$B = f_{max} - f_{min}$ = worst case − 0 (black ده أو white ده) =

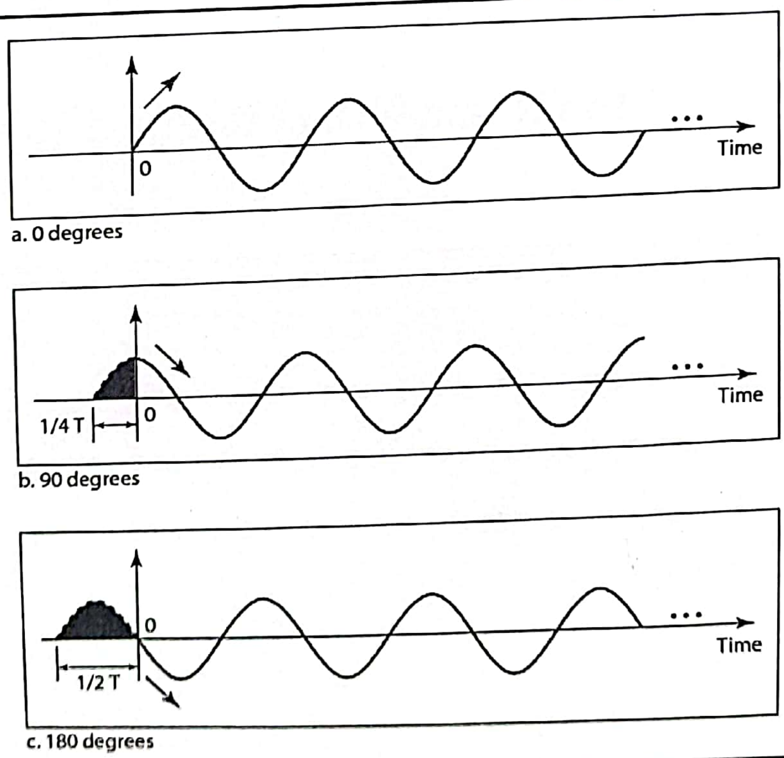
5.5125 MHz − 0 = 5.5125 MHz.

Note

The phase describes the position of the waveform relative to time 0.

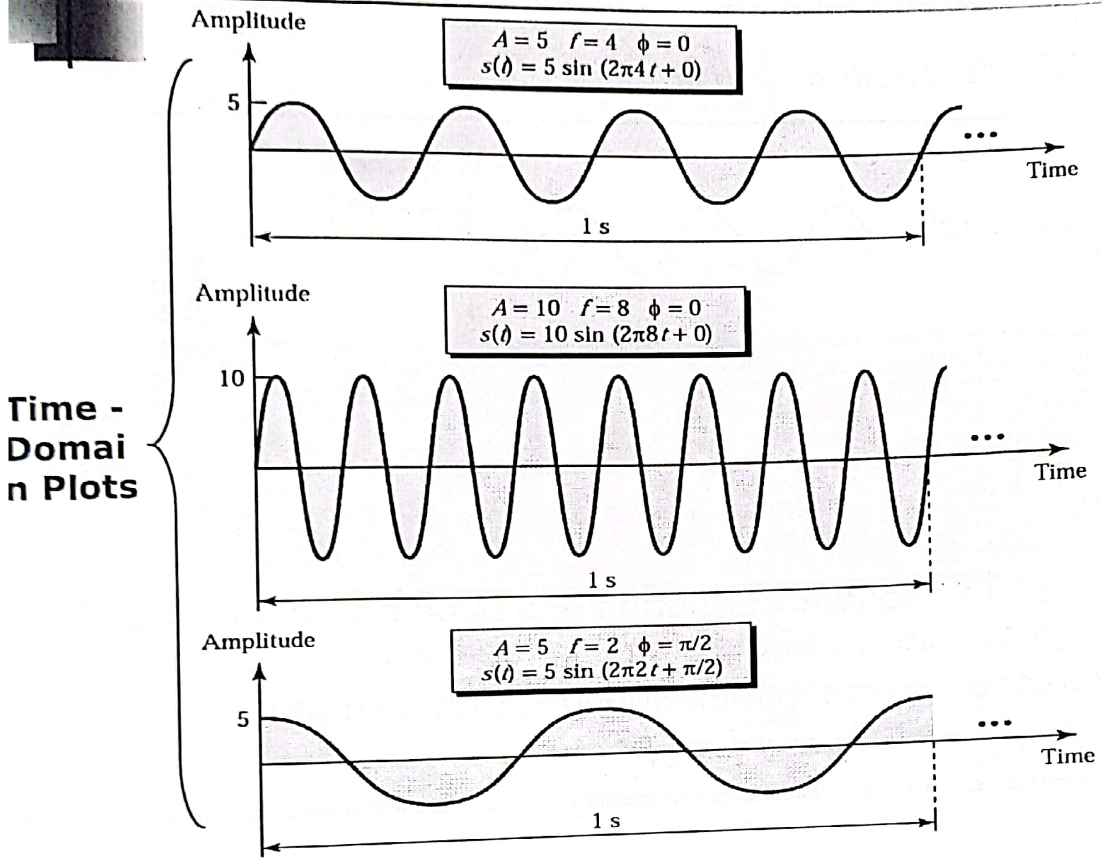
3.19

Figure 3.5 *Three sine waves with the same amplitude and frequency, but different phases* " Shift in time 0"



3.20

Sine wave examples



Example 3.6

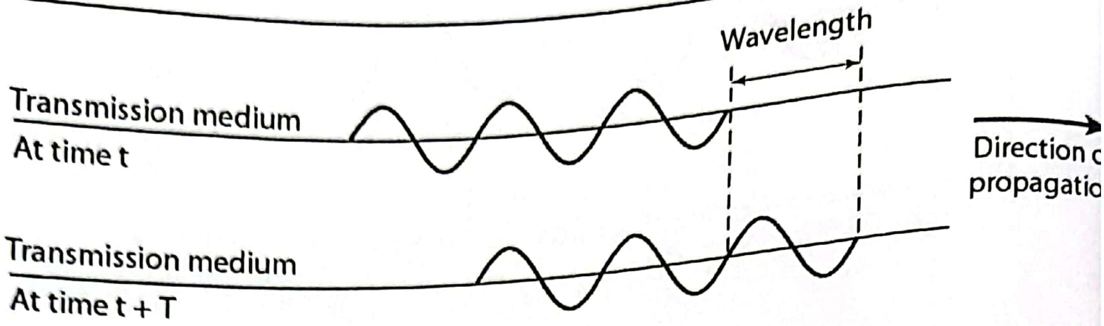
A sine wave is offset 1/6 cycle with respect to time 0. What is its phase in degrees and radians?

Solution

We know that 1 complete cycle is 360°. Therefore, 1/6 cycle is

$$\boxed{\frac{1}{6} \times 360 = 60^\circ = 60 \times \frac{2\pi}{360} \text{ rad} = \frac{\pi}{3} \text{ rad} = 1.046 \text{ rad}}$$

Figure 3.6 *Wavelength and period*

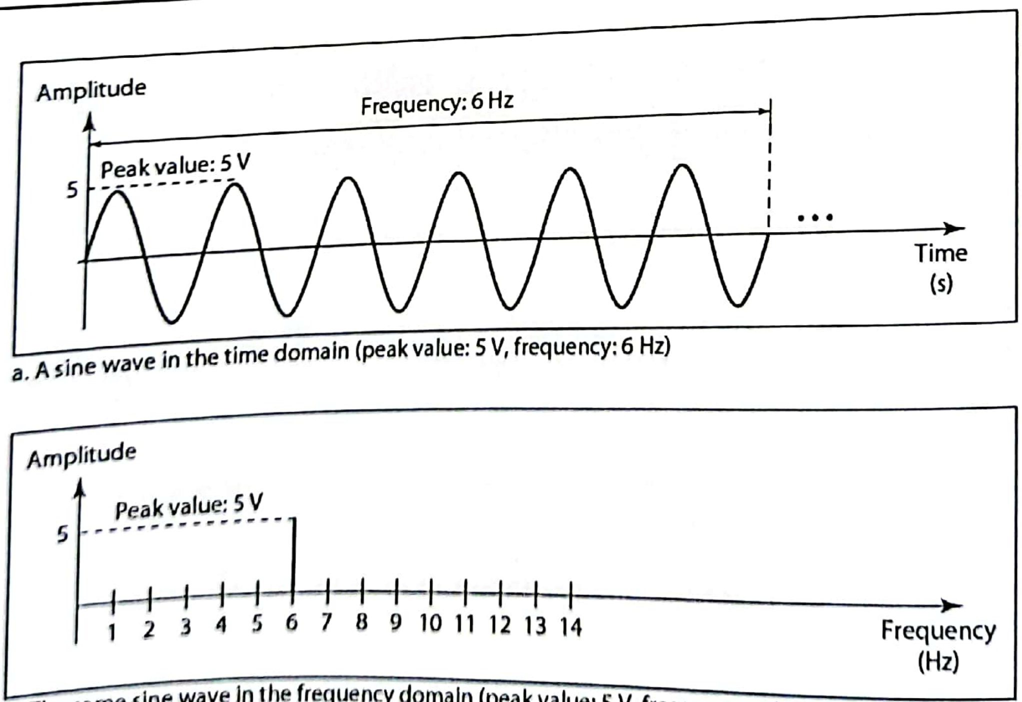


- Wavelength is the distance traveled by a simple signal in one period
- Relates the frequency or period of a simple signal to the propagation speed of the signal in medium
- Wavelength = Propagation Speed x Signal Period

$$\lambda = c \cdot T = c / f$$

3.23

Figure 3.7 *The time-domain and frequency-domain plots of a sine wave*



a. A sine wave in the time domain (peak value: 5 V, frequency: 6 Hz)

b. The same sine wave in the frequency domain (peak value: 5 V, frequency: 6 Hz)

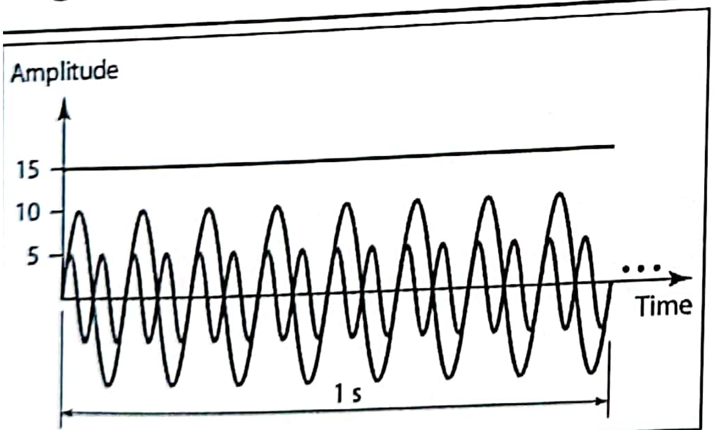
3.24

Note

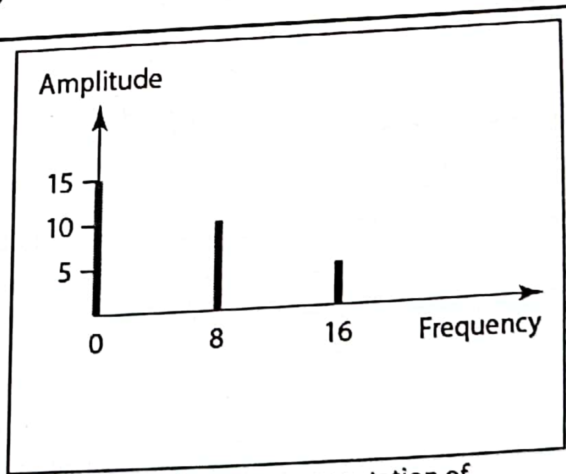
A complete sine wave in the time domain can be represented by one single spike in the frequency domain.

25

Figure 3.8 *The time domain and frequency domain of three sine waves*



a. Time-domain representation of three sine waves with frequencies 0, 8, and 16



b. Frequency-domain representation of the same three signals

<u>Example 3.7</u>

The frequency domain is more compact and useful when we are dealing with more than one sine wave. For example, Figure 3.8 shows three sine waves, each with different amplitude and frequency. All can be represented by three spikes in the frequency domain.

26

Note

A single-frequency sine wave is not useful in data communications;
we need to send a composite signal, a signal made of many simple sine waves

Note

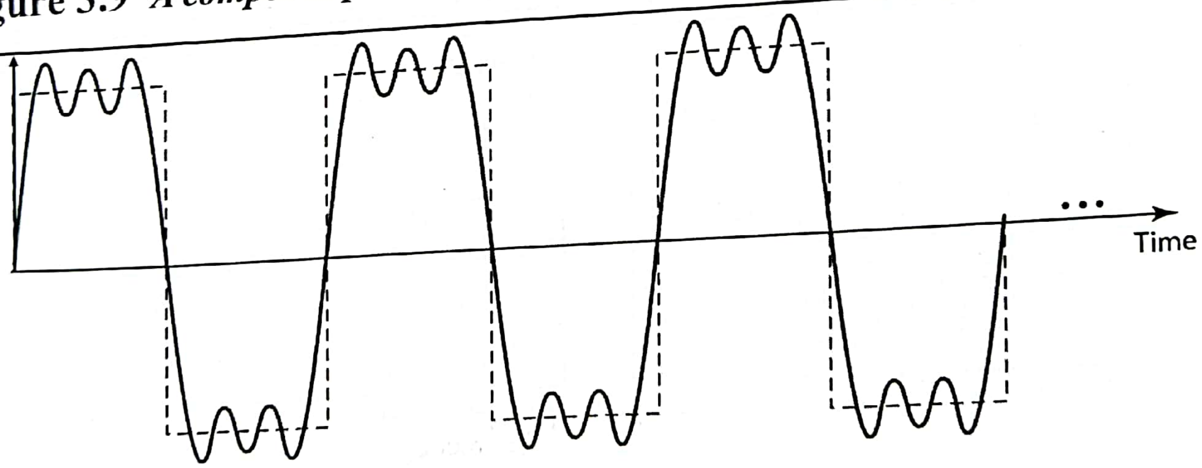
According to Fourier analysis, any composite signal is a linear combination of simple sine waves with different frequencies, amplitudes, and/or phases
Fourier analysis is discussed in Appendix C.

Note

If the composite signal is periodic, the decomposition gives a series of signals with discrete frequencies.

If the composite signal is nonperiodic, the decomposition gives a combination of sine waves with continuous frequencies.

Figure 3.9 *A composite periodic signal*



Example 3.8

Figure 3.9 shows a periodic composite signal with frequency f. This type of signal is not typical of those found in data communications. We can consider it to be three alarm systems, each with a different frequency. The analysis of this signal can give us a good understanding of how to decompose signals.

Figure 3.10 *Decomposition of a composite periodic signal in the time and frequency domains*

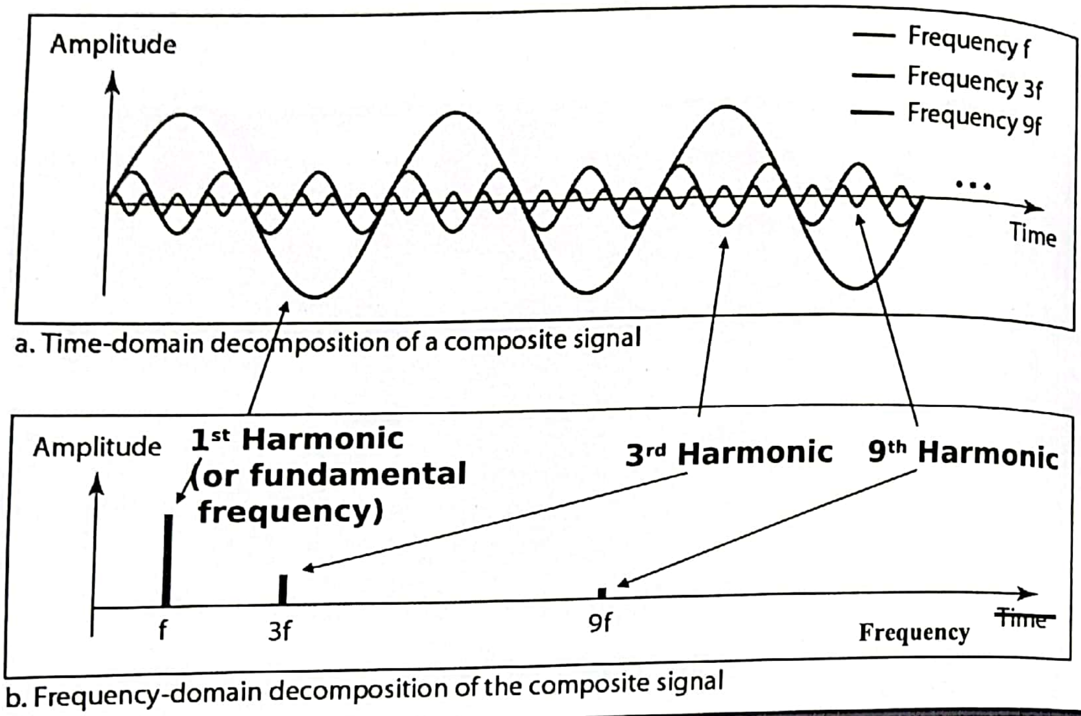
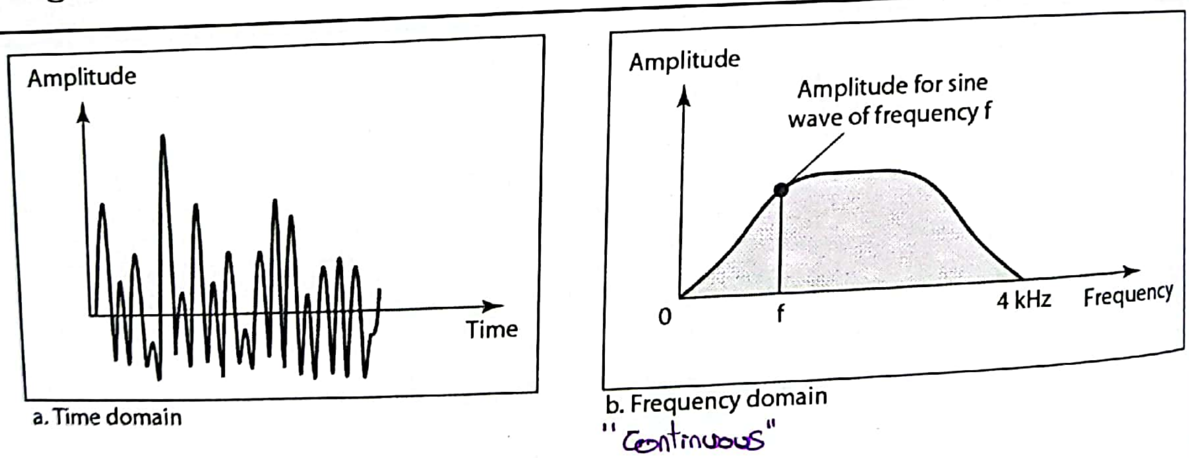


Figure 3.11 *The time and frequency domains of a nonperiodic signal*



<u>*Example 3.9*</u>

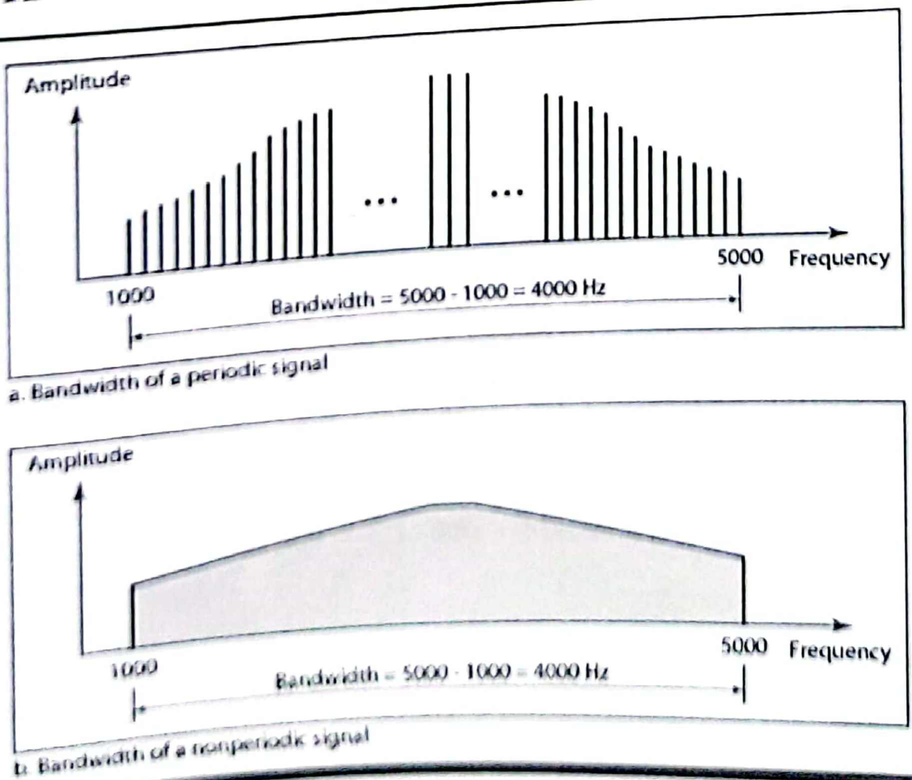
Figure 3.11 shows a nonperiodic composite signal. It can be the signal created by a microphone or a telephone set when a word or two is pronounced. In this case, the composite signal cannot be periodic, because that implies that we are repeating the same word or words with exactly the same tone.

Note

The bandwidth of a composite signal is the difference between the highest and the lowest frequencies contained in that signal.

* bandwidth signal يختلف عن ال bandwidth channel

Figure 3.12 The bandwidth of periodic and nonperiodic composite signals



Example 3.10

If a periodic signal is decomposed into five sine waves with frequencies of 100, 300, 500, 700, and 900 Hz, what is its bandwidth? Draw the spectrum, assuming all components have a maximum amplitude of 10 V.

Solution

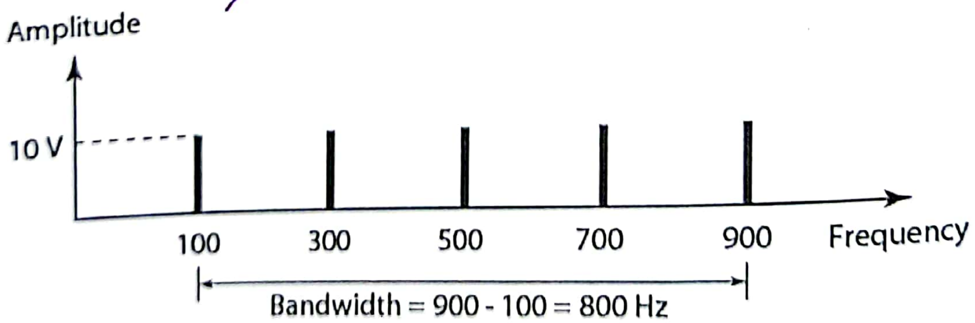
Let f_h be the highest frequency, f_l the lowest frequency, and B the bandwidth. Then

$$\boxed{B = f_h - f_l = 900 - 100 = 800 \text{ Hz}}$$

The spectrum has only five spikes, at 100, 300, 500, 700, and 900 Hz (see Figure 3.13).

Figure 3.13 *The bandwidth for Example 3.10*

بنعرف إنها Periodic لأنه طلع
معاك discrete values.



Example 3.11

A periodic signal has a bandwidth of 20 Hz. The highest frequency is 60 Hz. What is the lowest frequency? Draw the spectrum if the signal contains all frequencies of the same amplitude.

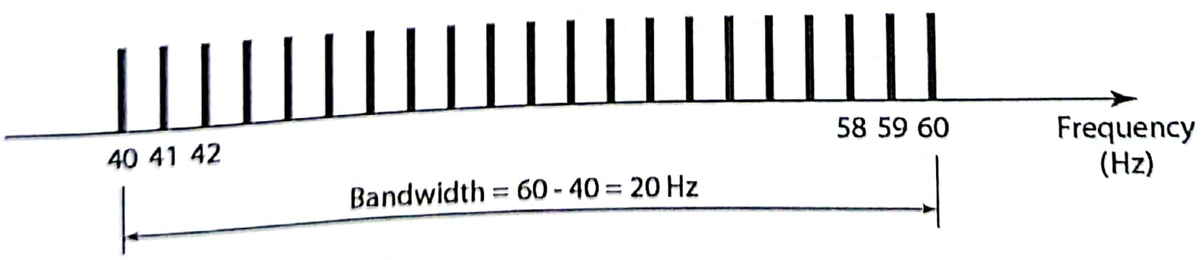
Solution

Let f_h be the highest frequency, f_l the lowest frequency, and B the bandwidth. Then

$$\boxed{B = f_h - f_l \quad \Rightarrow \quad 20 = 60 - f_l \quad \Rightarrow \quad f_l = 60 - 20 = 40 \text{ Hz}}$$

The spectrum contains all integer frequencies. We show this by a series of spikes (see Figure 3.14).

Figure 3.14 ***The bandwidth for Example 3.11***



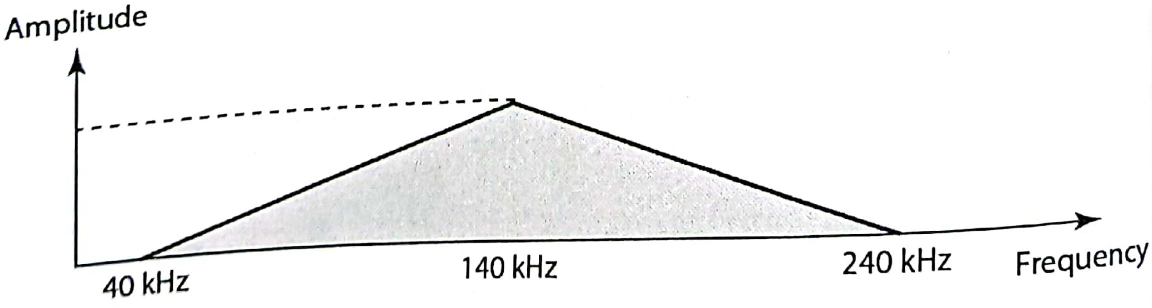
Example 3.12

A nonperiodic composite signal has a bandwidth of 200 kHz, with a middle frequency of 140 kHz and peak amplitude of 20 V. The two extreme frequencies have an amplitude of 0. Draw the frequency domain of the signal. → باعتبار انه ال Figure تكون symmetrical.

Solution

The lowest frequency must be at 40 kHz and the highest at 240 kHz. Figure 3.15 shows the frequency domain and the bandwidth.

Figure 3.15 ***The bandwidth for Example 3.12***



Example 3.13

An example of a nonperiodic composite signal is the signal propagated by an AM radio station. In the United States, each AM radio station is assigned a 10-kHz bandwidth. The total bandwidth dedicated to AM radio ranges from 530 to 1700 kHz. We will show the rationale behind this 10-kHz bandwidth in Chapter 5.

* AM, FM → Continuous أمثلة على ال
(Voices لأنها) . Frequency

Example 3.14

Another example of a nonperiodic composite signal is the signal propagated by an FM radio station. In the United States, each FM radio station is assigned a 200-kHz bandwidth. The total bandwidth dedicated to FM radio ranges from 88 to 108 MHz. We will show the rationale behind this 200-kHz bandwidth in Chapter 5.

Example 3.15

Another example of a nonperiodic composite signal is the signal received by an old-fashioned analog black-and-white TV. A TV screen is made up of pixels. If we assume a resolution of 525 × 700, we have 367,500 pixels per screen. If we scan the screen 30 times per second, this is 367,500 × 30 = 11,025,000 pixels per second. The worst-case scenario is alternating black and white pixels. We can send 2 pixels per cycle. Therefore, we need 11,025,000 / 2 = 5,512,500 cycles per second, or Hz. The bandwidth needed is 5.5125 MHz.

بعت 30 كل Second

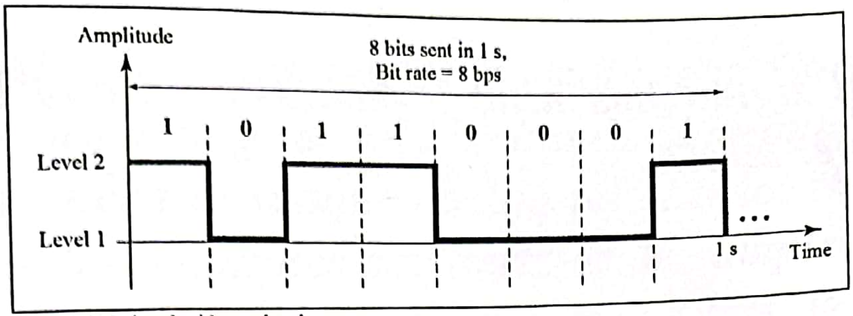
3-3 DIGITAL SIGNALS

In addition to being represented by an analog signal, information can also be represented by a ***digital signal****. For example, a 1 can be encoded as a positive voltage and a 0 as zero voltage. A digital signal can have more than two levels. In this case, we can send more than 1 bit for each level.*

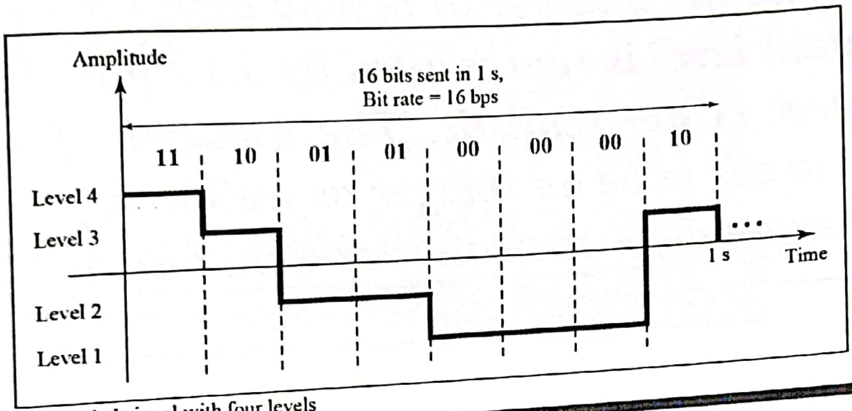
Topics discussed in this section:

Bit Rate
Bit Length
Digital Signal as a Composite Analog Signal
Application Layer

Figure 3.16 *Two digital signals: one with two signal levels and the other with four signal levels*



a. A digital signal with two levels



b. A digital signal with four levels

* لو بدي أرفع ال data Rate برفع ال Num of levels بس عليا حيكون صعب شوي لوجود ال Noise

Example 3.16

A digital signal has eight levels. How many bits are needed per level?

Solution

We calculate the number of bits using the formula

$$\boxed{\text{Number of bits per level} = \log_2 8 = 3}$$

Each signal level is represented by 3 bits.

18/10/2022

$11,025,000 \frac{\text{Pixels}}{s} * \frac{1 \text{ cycles}}{2 \text{ Pixels}} = 5,512,500 \frac{\text{cycles}}{s} (Hz) = 5.5125 \text{ MHz}$ (frequency).

↳ Multiple Not Convolution → DC ... Frequency

$B = f_{max} - f_{min}$ = worst case = 0 (black كله أو white كله) =

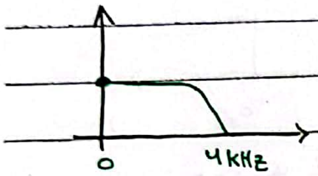
$5.5125 \text{ MHz} - 0 = 5.5125 \text{ MHz}$.

* Nyquist ← لازم الـ Sampling freq ضعف أعلى تردد موجود عندك.

Example Slide 3.49 :

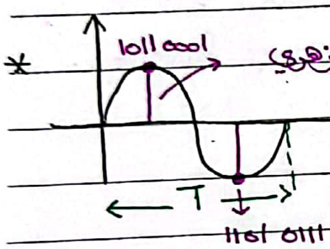
* الإنسان بيسمع Frequency من $20 Hz \rightarrow 20 KHz$ وبحكي من range تقريبا $0 \rightarrow 8 KHz$.

* لو حكيت من $0 \rightarrow 4 KHz$ بكون مازال كلامك واضح ومفهوم.



→ Continuous (non periodic)

كلما أخذت أكثر كل ما كان أفضل،
أقل عدد مسموح إنك تاخذه هو 2.



حنمثل هدول فقط اللي بالزهري

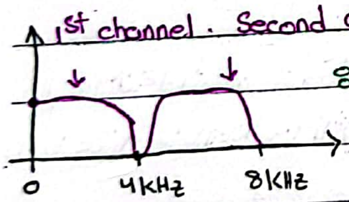
$f = \frac{1}{T} \Rightarrow f_s \geq 2f \Rightarrow f_s = 2f$

Sampling frequency ↲

$\Rightarrow f_s = \frac{2 \text{ Samples}}{T} = 2f$

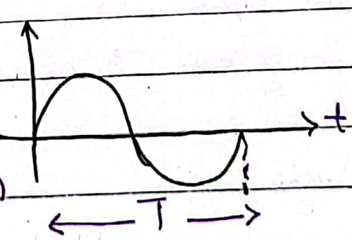
$\rightarrow f_s = 2 * (4 * 10^3) = 8 * 10^3 \frac{\text{Samples}}{s} \Rightarrow 8 * 10^3 \frac{\text{Samples}}{s} * \frac{8 \text{ bits}}{1 \text{ Samples}}$

$= 64$ kbps.



* لو كان أكثر من شخص بده يحكي بنقسم الـ frequency بينهم :

* Wavelength (طول الموجة) :



$f = \frac{1}{T} \rightarrow \frac{\lambda}{T} \cong C$ (wave length (meters))

↳ $\lambda f = C \rightarrow \lambda = C \times T$

سرعة الضوء ← تقريبا ثابتة قيمتها أقل بقليل من 300,000 km/s

* Slide 3.52 ← الـ bandwidth الهم =

a- $\infty - f = \infty$

b- $\infty - 0 = \infty$

Five Apple

Example 3.17

A digital signal has nine levels. How many bits are needed per level?

Solution

We calculate the number of bits by using the formula. Each signal level is represented by 3.17 bits. However, this answer is not realistic. The number of bits sent per level needs to be an integer as well as a power of 2. For this example, 4 bits can represent one level.

Example 3.18

Assume that we need to download text documents at the rate of 100 pages per minute. What is the required bit rate of the channel?

Solution

منمعطيات
السؤال

A page has an average of 24 lines with 80 characters in each line. If we assume that one character requires 8 bits, the bit rate is

$$100 \times 24 \times 80 \times 8 = 1.536 \text{ M bits per minute} = 25.6 \text{ kbps}$$

$$\boxed{100 \times 24 \times 80 \times 8 = 1{,}636{,}000 \text{ bps} = 1.636 \text{ Mbps}}$$ ← wrong

Example 3.19

A digitized voice channel, as we will see in Chapter 4, is made by digitizing a 4-kHz bandwidth analog voice signal. We need to sample the signal at twice the highest frequency (two samples per hertz or 8k samples/second). We assume that each sample requires 8 bits. What is the required bit rate?

Solution

The bit rate can be calculated as

$$\boxed{2 \times 4000 \times 8 = 64{,}000 \text{ bps} = 64 \text{ kbps}}$$

Example 3.20

What is the bit rate for high-definition TV (HDTV)?

Solution

HDTV uses digital signals to broadcast high quality video signals. The HDTV screen is normally a ratio of 16:9. There are 1920 by 1080 pixels per screen, and the screen is renewed 30 times per second. Twenty-four bits represent one color pixel.

$$\boxed{1920 \times 1080 \times 30 \times 24 = 1{,}492{,}992{,}000 \text{ or } 1.5 \text{ Gbps}}$$

The TV stations reduce this rate to 20 to 40 Mbps through compression.

Bit Length

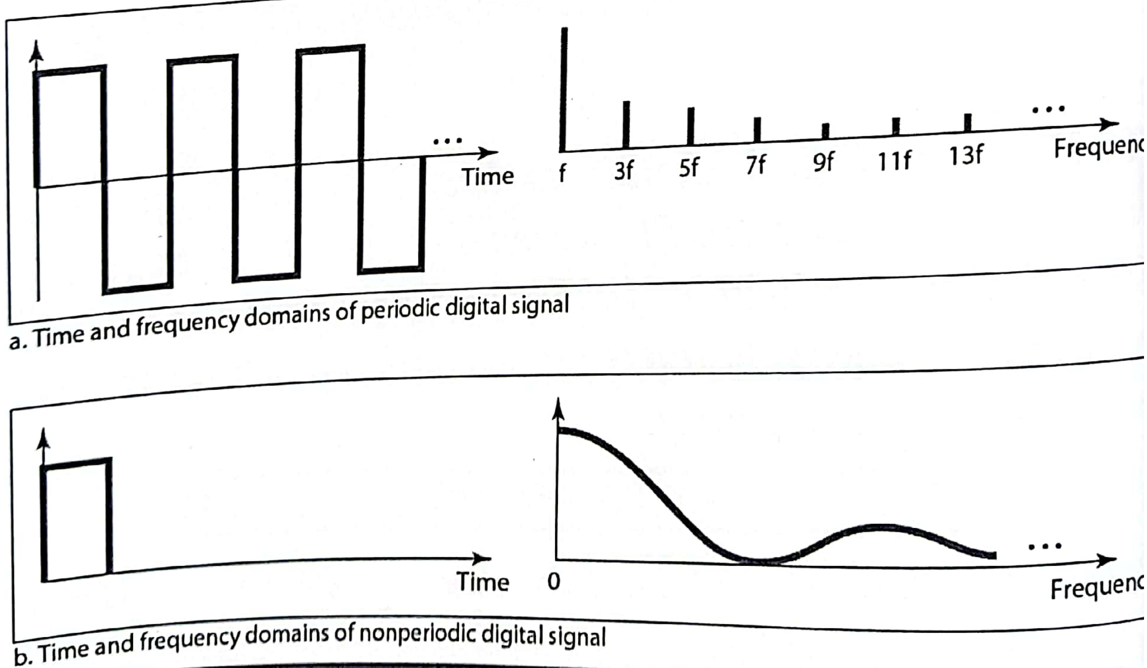
- The wavelength is the distance that one cycle occupies on the transmission medium, which is a characteristic of the analog signal.

Wavelength = Propagation Speed x Period

- The bit length is the distance that one bit occupies on the transmission medium, which is a characteristic of the digital signal.

Bit Length = Propagation Speed x Bit Duration
= Propagation Speed / Bit Rate

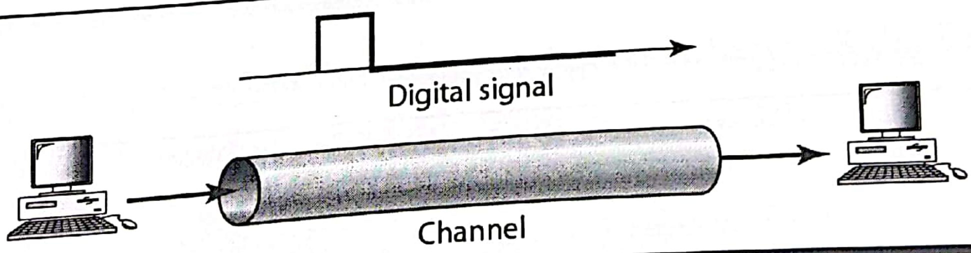
Figure 3.17 *The time and frequency domains of periodic and nonperiodic digital signals*



Note

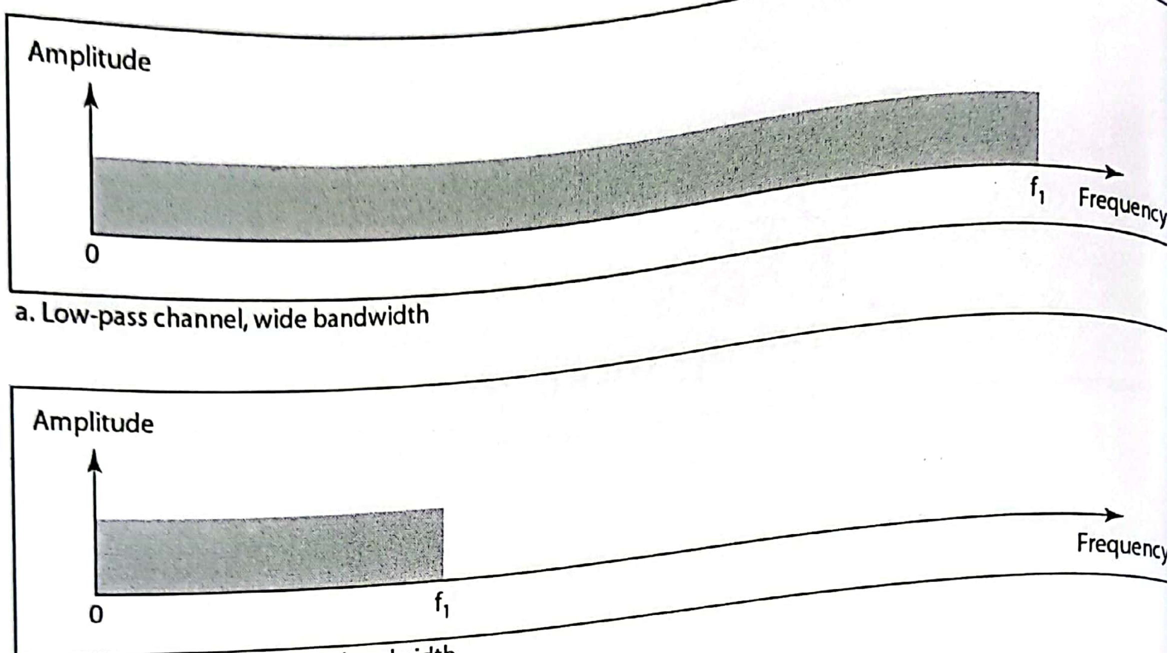
A digital signal is a composite analog signal with an infinite bandwidth.

Figure 3.18 *Baseband transmission*



- Baseband transmission:
  - Sending a digital signal over a channel without changing it to analog (i.e.; without modulation)
  - Requires low-pass channel
- The low-pass channel has a bandwidth that starts from zero
  - E.g. a cable connecting two or more computers
  - Narrow-band versus wide-band low-pass channels

Figure 3.19 *Bandwidths of two low-pass channels*

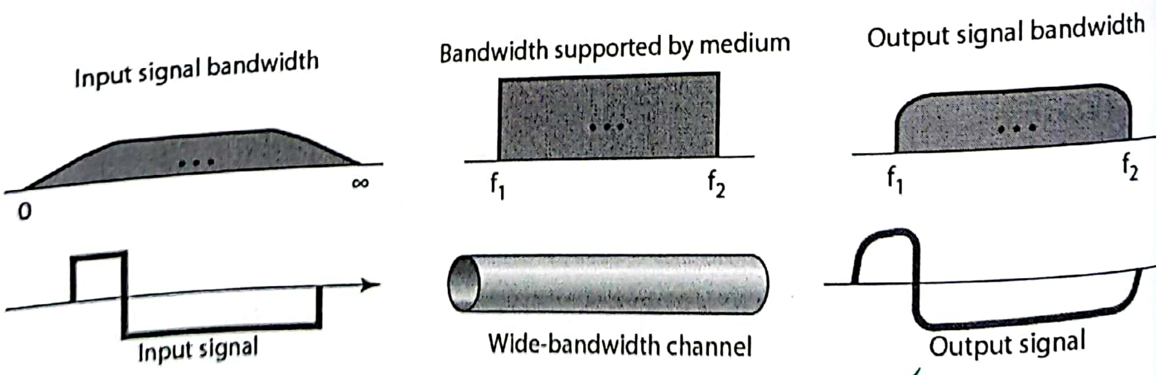


a. Low-pass channel, wide bandwidth

b. Low-pass channel, narrow bandwidth

Case 1: Low-Pass Channel with Wide Bandwidth

Figure 3.20 *Baseband transmission using a dedicated medium*



شبیه کپی بشکل
Input Signal را

Note

Baseband transmission of a digital signal that preserves the shape of the digital signal is possible only if we have a low-pass channel with an infinite or very wide bandwidth.

Example 3.21

An example of a dedicated channel where the entire bandwidth of the medium is used as one single channel is a LAN.

Almost every wired LAN today uses a dedicated channel for two stations communicating with each other.

- *In a bus topology LAN with multipoint connections, only two stations can communicate with each other at each moment in time (timesharing); the other stations need to refrain from sending data.*
- *In a star topology LAN, the entire channel between each station and the hub is used for communication between these two entities.*

- *LANs will be covered in Chapter 14.*

Case 2: Low-Pass Channel with Limited Bandwidth

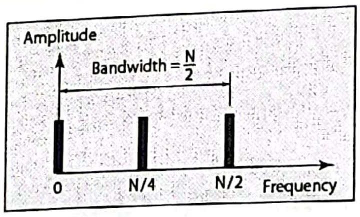
← بالحياة العملية عادة هيك بتكون.

- Approximate the digital signal with an analog signal depending on the available bandwidth
- Consider a digital signal of bit rate N:
  - Rough approximation
    - Use first harmonic only (least bandwidth usage)
    - The worst case is a sequence of alternating bits of 0's & 1's
    - To simulate the worst case, we need an analog signal of frequency with f=N/2
    - More signal components are needed to represent the different patterns. For example, the signal phase
  - Better approximation
    - Use more harmonics (more bandwidth usage)
    - Results in a better representation of the digital signal

* First harmonic → frequency = 1

Figure 3.21 *Rough approximation of a digital signal using the first harmonic for worst case*

A digital signal with 3-bit pattern



* الـ Square wave بنقربها لـ Sin wave
(خلي هالموضوع بعين الاعتبار)

phase

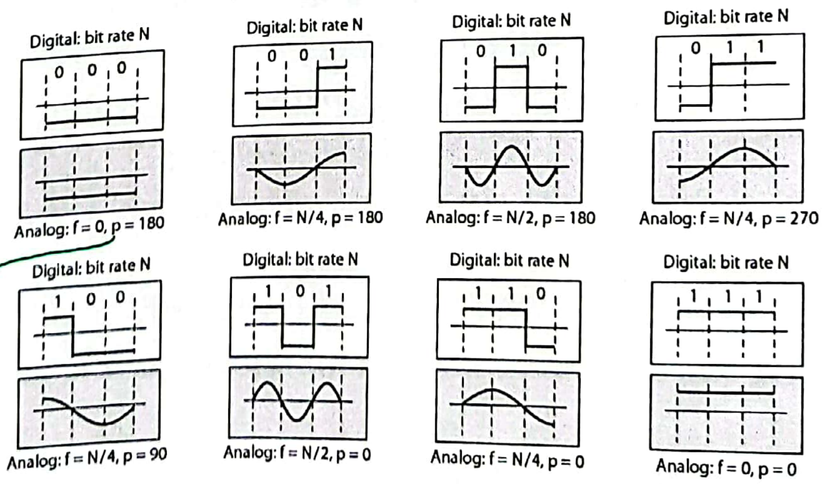
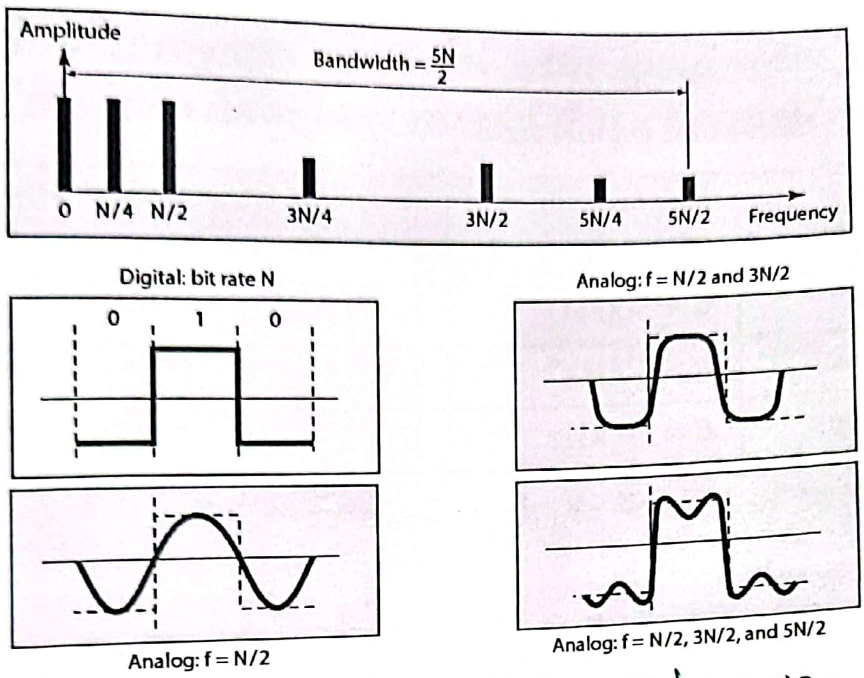


Figure 3.22 *Simulating a digital signal with first three harmonics*



* اللي بالأحمر worst case Alternative Zeros and ones.

* الـ fourier series للـ Square wave بتكون لما تكون even cos وبتطلع بس الأشياء الفردية ، أما الـ Sin بالتكامل بتروح كلها صفر.

* min bandwidth → take one harmonic.

* لو ضليت أزيد بالـ harmonic لأزيد الدقة حدفع ضريبة انه الـ bandwidth بزيد والـ bandwidth مشكلة بالاتصالات صار ومسافة خطرة.

3.61

Note

In baseband transmission, the required bandwidth is proportional to the bit rate; if we need to send bits faster, we need more bandwidth.

3.62

Table 3.2 *Bandwidth requirements*

Bit Rate	*Harmonic 1*	*Harmonics 1, 3*	*Harmonics 1, 3, 5*
$n = 1$ kbps	$B = 500$ Hz	$B = 1.5$ kHz	$B = 2.5$ kHz
$n = 10$ kbps	$B = 5$ kHz	$B = 15$ kHz	$B = 25$ kHz
$n = 100$ kbps	$B = 50$ kHz	$B = 150$ kHz	$B = 250$ kHz

« Propotronal of 2 العلاقة تقريباً »

3.63

Example 3.22

What is the required bandwidth of a low-pass channel if we need t send 1 Mbps by using baseband transmission?

Solution

The answer depends on the accuracy desired.

من الجدول فوق

a. The minimum bandwidth, is B = bit rate /2, or 500 kHz.

b. A better solution is to use the first and the third harmonics with B = 3 × 500 kHz = 1.5 MHz.

c. Still a better solution is to use the first, third, and fifth harmonics with B = 5 × 500 kHz = 2.5 MHz.

3.64

* Bandwidth الـ Signal بيختلف عن Bandwidth الـ Channel.

* Low pass → بتبدأ من الصفر
وبتنتهي حسب ما انت بدك.

Slide 3.59

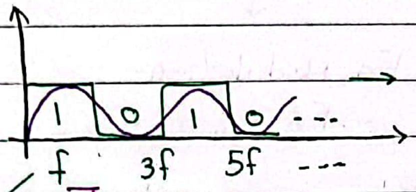
هون أخذنا 3bits كمثال.

لو نبعت ...101 ← worst case

* خنشغل الـ worst case انه
نبعث Alternative من 0,1

* 2 worst cases:
A- 10101...
B- 01010....

* N: data rate
* T: Period



* $N = \frac{2}{T}\ \frac{bits}{s} \rightarrow N = 2f \rightarrow f = \frac{1}{T} \rightarrow \frac{N}{2} = f \rightarrow f \propto N$

* $Bandwidth = \frac{N}{2}$ (Bandwidth = Max - Min = f - 0 = f)

* كل ما بدك data rate عالي لازم تزيد الـ Bandwidth.

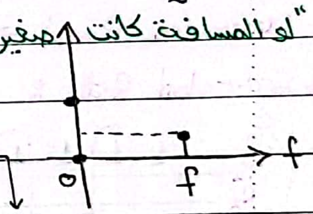
20/10/2022

~~Example~~ 3.22 Page 3.65

* Rough approximation is referred to as using the first harmonic, $f = \frac{N}{2}$

* Low-Pass (Base Band) (we should start from 0) → "لو المسافة كانت صغيرة"

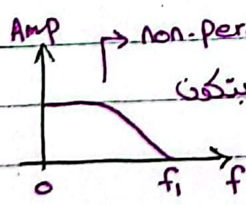
Minimum Bandwidth: $B = f - 0 = f$



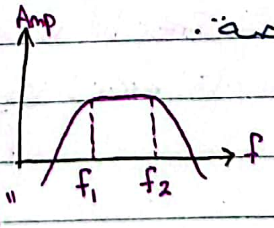
* $B = \frac{N}{2} \rightarrow N = 2B$ → Maximum Data Rate (first harmonic) → Best case

* other Cases: $B = \frac{3N}{2} \rightarrow N = \frac{2}{3}B$ / $B = \frac{5N}{2} \rightarrow N = \frac{2}{5}B$
Second ↲ third ↲

* Band pass ← ما بتبدأ من الصفر مثل الـ Low-Pass (هو حالة عامة أما الـ Low-Pass حالة خاصة.



→ non-periodic ←
لإنه الـ Values بتكون
Continuous
↓
"لو كانت Signal"



Low-pass: $B = f_1 - 0$ Band Pass: $B = f_2 - f_1$

* الـ Band-pass أفضل لإنه أعم وبياخد أكثر من حالة مو زي الـ Low-Pass
بس حالة وحدة.

* الـ Band ممكن يتعرف لـ Signal أو لـ Channel.

* الـ Band-pass channel دائمًا Continuous.

Five Apple

Example 3.22

We have a low-pass channel with bandwidth 100 kHz. What is the maximum bit rate of this channel?

Solution

The maximum bit rate can be achieved if we use the first harmonic. The bit rate is 2 times the available bandwidth, or 200 kbps.

مثلاً 8 بدك ترمي ورقة لصديقك وتضمن يوصله فبتلفها بحجرة (carrier) وبترميها هيك بتضمن وصولها.

لما نعمل transmission للـ Signal بكون فيها Power محدودة ما بتقدر توصل لأماكن بعيدة فهيك بنضطر أحملها ع Signal ثانية فيها Power أقوى لتوصل مسافة أبعد ، الـ Modulation بتزود من الـ Power للـ Signal.

وحدة من أهدافها نعمل shift للـ bandwidth.

Broadband Transmission (Using Modulation)

- Changing the digital signal to an analog signal for transmission (or carrying the digital signal on top of an analog signal called carrier signal)
- Allows for bandpass channel usage, which is the channel that does not start from zero
- Bandpass is more available than low-pass. Why?
- Bandpass channels can be divided into smaller channels for bandwidth sharing (e.g. FDMA) → Part of the bandwidth all the time → جزء محجوز الك طول الوقت

  ↳ Frequency division Multiplexing

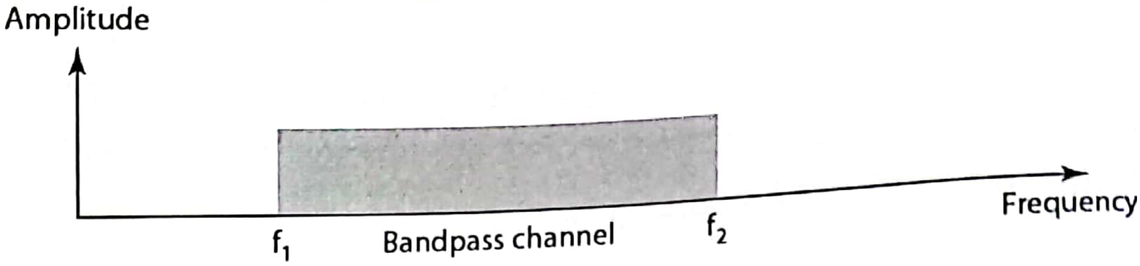
  ↳ مثلاً نقسم الطيف الكهرومغناطيسي للصوت لـ 4K ← 8K ← 12K وهكذا.

Figure 3.23 *Bandwidth of a bandpass channel*

low-pass channel

* / bandpass channel لازم نمثلها إنها Continuous لأنها تمثل جميع النقاط

اللي ممكن تستوعبها أما الـ Signal تصنف كـ Periodic أو non-Periodic



* ليه ما في channel بتمرق كل الـ frequencies ؟ لأنه في frequencies

بتمر وفي لأ يعني زي كأنه بفلتر الـ frequencies فلترة .

3.67

Note

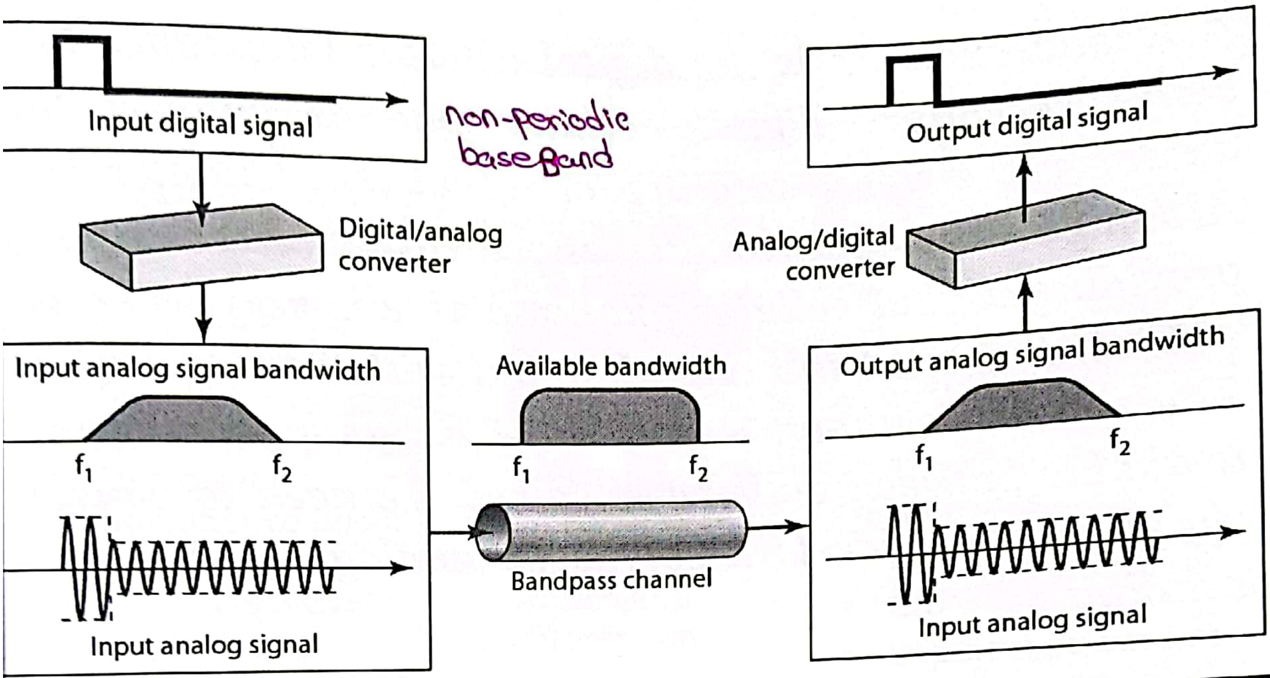
If the available channel is a bandpass channel, we cannot send the digital signal directly to the channel; we need to convert the digital signal to an analog signal before transmission.

لأنه الـ Squarewave ← اذا كانت Periodic الها الـ DC component إلى ∞ - اذا عندك channel معينة حتوصل الـ freq. اللي تحت وفوق وحيمشي جزء من الـ channel بالوسط .

إذا وصلنا الـ freq اللي فوق (العالي) هو مشكلة لإنه أحس) لأنه الـ Power قليل والـ Amp قليل) أما الـ freq اللي تحت موكويس لإنه Power و Amp عالي. في مشكلة إنه ما بنقدر نبعث square wave على bandpass channel لازم تحولها زي ما مكتوب .

3.68

Figure 3.24 *Modulation of a digital signal for transmission on a bandpass channel*



Example 3.24

An example of broadband transmission using modulation is the sending of computer data through a telephone subscriber line, the line connecting a resident to the central telephone office. These lines are designed to carry voice with a limited 0→4 bandwidth. The channel is considered a bandpass channel. We convert the digital signal from the computer to an analog signal, and send the analog signal. We can install two converters to change the digital signal to analog and vice versa at the receiving end. The converter, in this case, is called a modem which we discuss in detail in Chapter 5.

Example 3.25

A second example is the digital cellular telephone. For better reception, digital cellular phones convert the analog voice signal to a digital signal (see Chapter 16). Although the bandwidth allocated to a company providing digital cellular phone service is very wide, we still cannot send the digital signal without conversion. The reason is that we only have a bandpass channel available between caller and callee. We need to convert the digitized voice to a composite analog signal before sending.

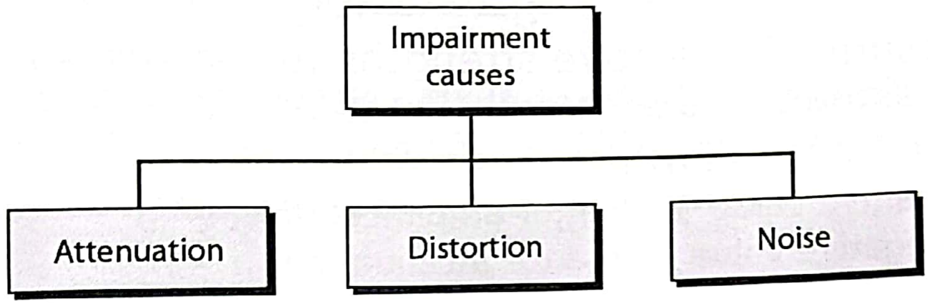
3-4 TRANSMISSION IMPAIRMENT

*Signals travel through transmission media, which are not perfect. The imperfection causes signal impairment. This means that the signal at the beginning of the medium is not the same as the signal at the end of the medium. What is sent is not what is received. Three causes of impairment are **attenuation, distortion,** and **noise.***

Topics discussed in this section:

Attenuation
Distortion
Noise

Figure 3.25 *Causes of impairment*



Attenuation

- Loss of signal energy due to channel resistance
  و Inductors Capacitance في كمان
- Use power amplifier to boost the signal
  « لحل المشكلة »

Distortion

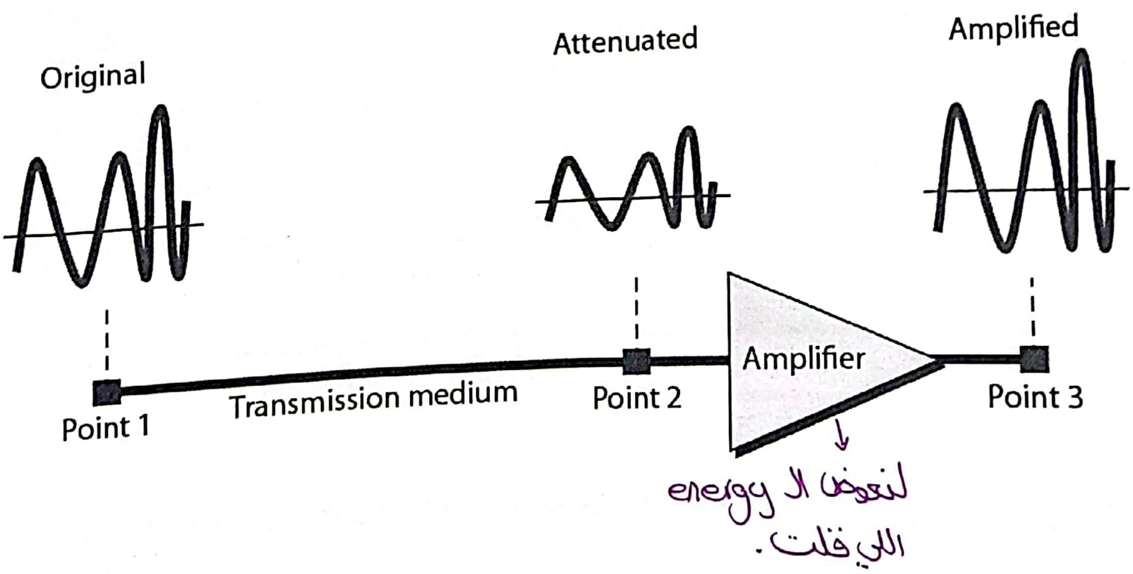
- Change in composite signals shape due to channel limitations
- Each component propagates differently and arrives at a different time

Noise

- Unwanted random energy in the channel that may corrupt the signal
- Thermal: electrons in the wires
- Induced: by motors
- Crosstalk: wire on wire
- Impulse: power line

73

Figure 3.26 *Attenuation*

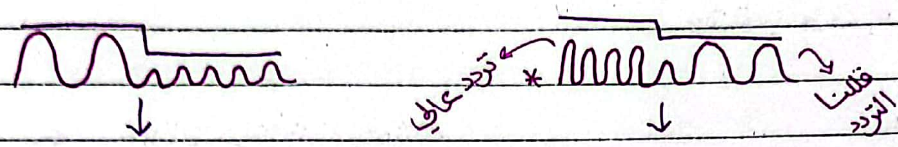


.74

والـ freq. عالي المقاومة قليلة

* $Z_C = \frac{1}{j\omega C}$ → $|Z_C| = \frac{1}{\omega C}$ → Angular frequency

* الـ BaseBand صار اسمه broadband لإنه تطورت الأمور وكبرت.



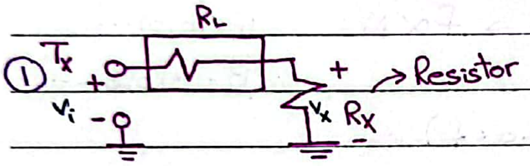
* Amp Modulation

* Fm Modulation ↳ Frequency

* phase Modulation

* Slide 3.72 : Impairment causes:

3 Problems: 1- Attenuation 2- Distortion 3- Noise



* $V_x = \left(\frac{R_x}{R_L + R_x}\right) V_i$

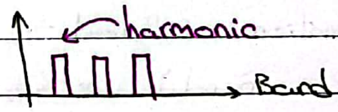
* $R_L = 0 \Rightarrow V_x = V_i$ → $R_L \uparrow$ $V_x \downarrow$

* Periodic → discrete / Non-periodic → continuous

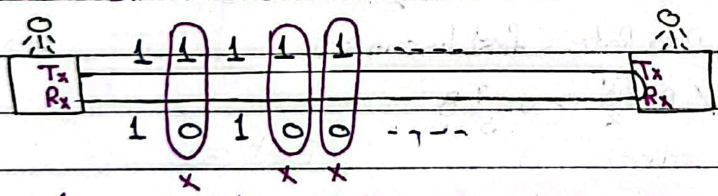
* Worst case: 010101... / Best case: 0000... or 1111...

* Data Rate ↑ → Bandwidth ↑

* Minimum Band → 1 harmonic →



* Error bit Rate : (bit error rate)



* 10^6 → All bits (total ones اللي بيتقم)
* 10^2 → Worng bits → Error bit Rate بطلعوم الـ
* $P = \frac{10^2}{10^6} = 10^{-4}$ → على الصفحة لازم يكون الـ error أقل

Decibel (dB)

→ نستخدمها لنمثل ال Power أو ال Voltage لل Signal.

- Measures the relative strengths (or power levels) of two signals or a signal at two different points
- dB = 10 $\log_{10}(P_2/P_1)$
  - Positive value → gain (or amplification) تضخيم
  - Negative value → loss (or attenuation)

بدل الضرب والقسمة.

- dB values can be added or subtracted easily for several points (cascading)

- 10 $\log_{10}(P_2/P_1)$ = 3 dB ➔ P_2 is as twice as P_1
- 10 $\log_{10}(P_2/P_1)$ = -3 dB ➔ P_2 is as half as P_1
- 10 $\log_{10}(P_2/P_1)$ = 10 dB ➔ P_2 is ten times P_1
- 10 $\log_{10}(P_2/P_1)$ = -10 dB ➔ P_2 is 10% of P_1

Example 3.26

Suppose a signal travels through a transmission medium and its power is reduced to one-half. This means that P_2 is (1/2)P_1. In this case, the attenuation (loss of power) can be calculated as

$$\boxed{10\log_{10}\frac{P_2}{P_1} = 10\log_{10}\frac{0.5P_1}{P_1} = 10\log_{10}0.5 = 10(-0.3) = -3\text{ dB}}$$

نعرف إنه نص ال Power راحت.

A loss of 3 dB (–3 dB) is equivalent to losing one-half the power.

Example 3.27

A signal travels through an amplifier, and its power is increased 10 times. This means that $P_2 = 10P_1$. In this case, the amplification (gain of power) can be calculated as

$$10 \log_{10} \frac{P_2}{P_1} = 10 \log_{10} \frac{10P_1}{P_1}$$

$$= 10 \log_{10} 10 = 10(1) = 10 \text{ dB}$$

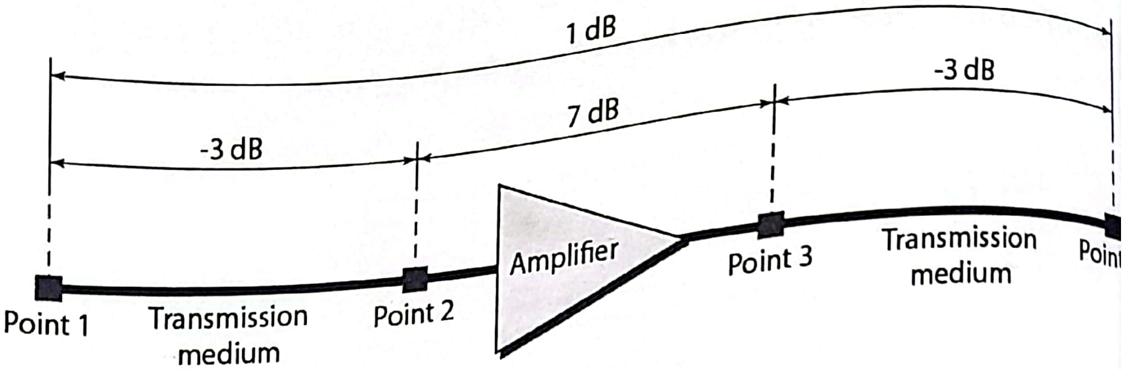
Example 3.28

One reason that engineers use the decibel to measure the changes in the strength of a signal is that decibel numbers can be added (or subtracted) when we are measuring several points (cascading) instead of just two.
In Figure 3.27 a signal travels from point 1 to point 4. In this case, the decibel value can be calculated as

$$\text{dB} = -3 + 7 - 3 = +1$$

صرت اجمع
واطرح مباشرة
بدل ما اضرب واقسم

Figure 3.27 *Decibels for Example 3.28*



بدل ما أضرب وأقسم بصير أجمع وأطرح .

Example 3.29

Sometimes the decibel is used to measure signal power in milliwatts. In this case, it is referred to as dB_m and is calculated as $dB_m = 10 \log_{10} P_m$, where P_m is the power in milliwatts. Calculate the power of a signal with $dB_m = -30$.

$* \; dB_m = 10 \log \frac{P}{1mw}$ ثابتة

Solution

We can calculate the power in the signal as

$$\boxed{\begin{array}{c} dB_m = 10 \log_{10} P_m = -30 \\ \log_{10} P_m = -3 \qquad P_m = 10^{-3}\ \text{mW} \end{array}}$$

$\cancel{10}^{\log_{10} P_m} = 10^{-3} \rightarrow P_m = 10^{-3}$ ← رفعت للقوى 10

Example 3.30

The loss in a cable is usually defined in decibels per kilometer (dB/km).

If the signal at the beginning of a cable with −0.3 dB/km has a power of 2 mW, what is the power of the signal at 5 km?

← كل كيلو بنزل 0.3 الـ Signal ← طاقة

Solution

The loss in the cable in decibels is 5 × (−0.3) = −1.5 dB.

We can calculate the power as

$$\text{dB} = 10\log_{10}\frac{P_2}{P_1} = -1.5$$

total loss من البداية للنهاية

$$\frac{P_2}{P_1} = 10^{-0.15} = 0.71$$

$$P_2 = 0.71P_1 = 0.7 \times 2 = 1.4\ \text{mW}$$

لو طلع أكثر من 2mW بكون غلط لأنه أنا بنزل فأكيد مش حفقد أكثر من 2mW وأنا أساساً ماعندي إلا 2mW

3.81

* Formula of $dB = 10\log_{10}\frac{P_2}{P_1}$

← كل ما زادت الـ freq نقصت طول الموجة .

→ Phase Distortion بالأخص

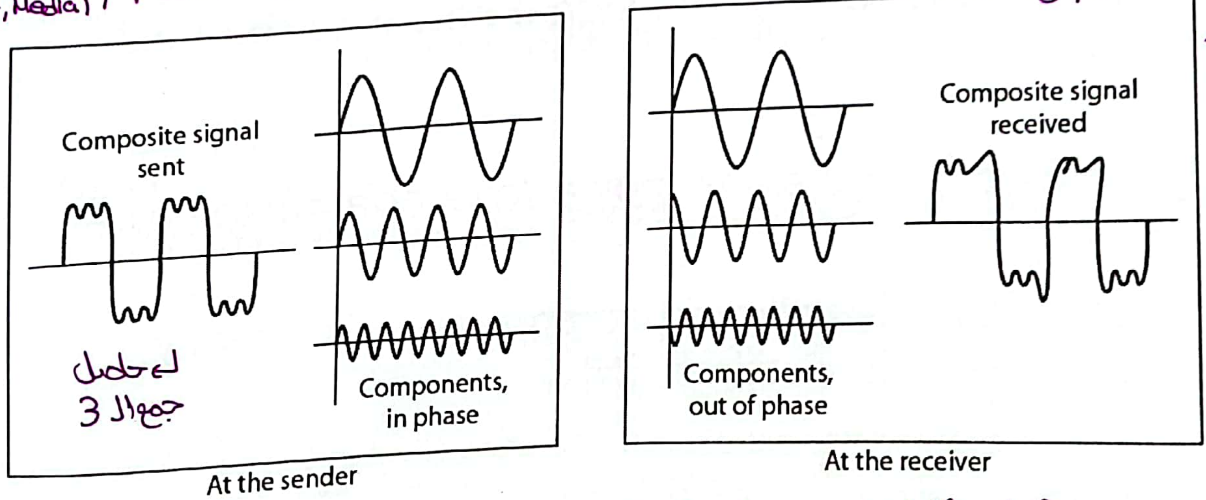
Figure 3.28 ***Distortion***

$\lambda f = c$ * بس كل ليست دقيقة الأدق : $\lambda f = u$ phase ← velocity

function تعتمد u الـ ← $= \frac{1}{\sqrt{\mu\varepsilon}}$

(f, Media) من الـ f والـ Media

حيمشي بسرعة مختلفة



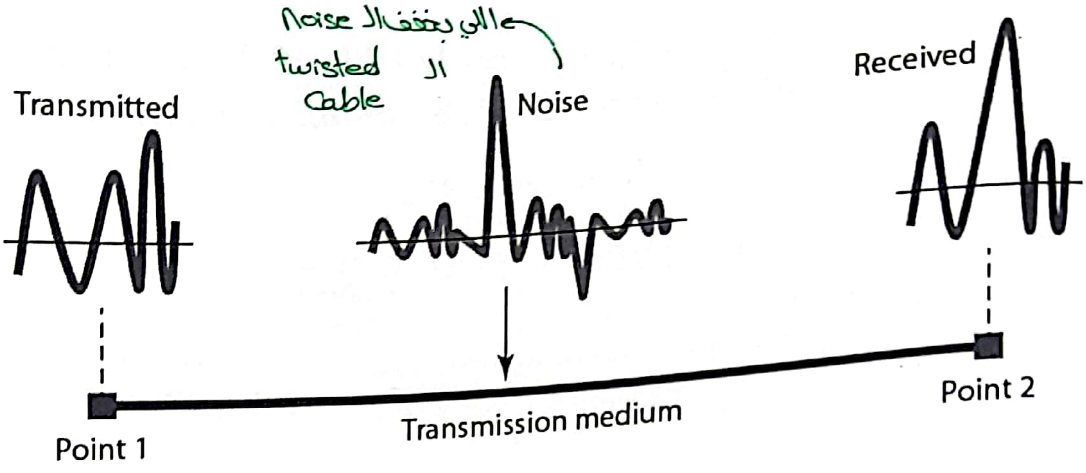
* different freq في الـ Media تمشي بسرعات مختلفة وهذا بيسمى distortion

← مدام مشيت بسرعات مختلفة بمسافة ثابتة فالزمن اللي بتوصل فيه مختلف

لذلك صار فيه اختلاف زي ما هو مبين بالصورة .

3.82

Figure 3.29 *Noise*



3.83

* كل ما كانت أعلى كل ما كانت أفضل.
* نسبة ال Power تاعت ال Signals إلى نسبة ال Power تبعت ال Noise.

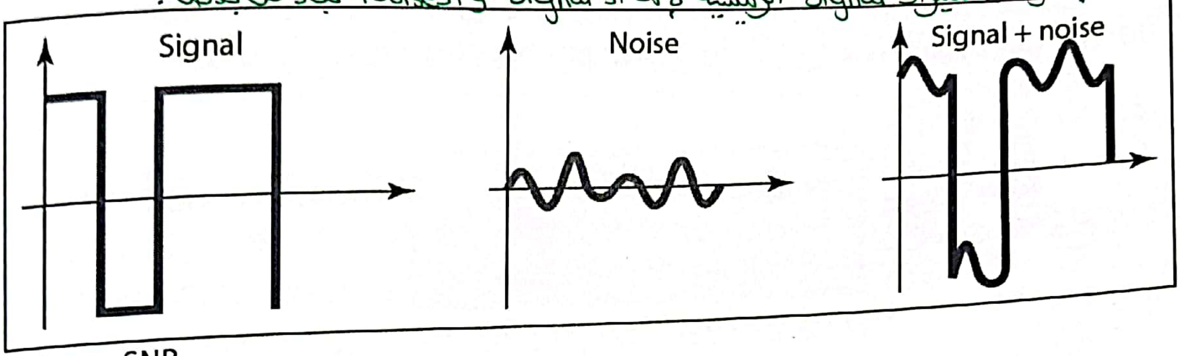
Signal-to-Noise Ratio (SNR)

- SNR is the ratio of the signal power to the noise power
- SNR=average signal power/average noise power
- Since SNR is ratio of two powers, it is usually expressed in decibel, SNR_{dB}
- $SNR_{dB} = 10 \log_{10} SNR = 10 \log_{10} (P_s / P_n)$

3.84

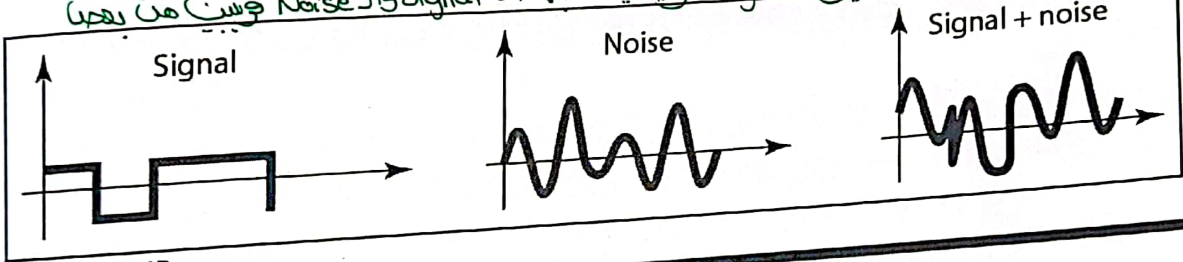
Figure 3.30 *Two cases of SNR: a high SNR and a low SNR*

بنقدر هون نميز الـSignal الرئيسية لأنه الـ Signal والـNoise بعاد عن بعض .



a. Large SNR

مابنقدر هون نميز الـ Signal الرئيسية لأنه الـ Signal والـNoise قريبين من بعض



b. Small SNR

.85

Example 3.31

The power of a signal is 10 mW and the power of the noise is 1 μW; what are the values of SNR and SNR_{dB}?

__Solution__

The values of SNR and SNR_{dB} can be calculated as follows:

$$SNR = \frac{10{,}000\ \mu W}{1\ mW} = 10{,}000$$

$$SNR_{dB} = 10 \log_{10} 10{,}000 = 10 \log_{10} 10^4 = 40$$

3.86

Example 3.32

The values of SNR and SNR_{dB} for a noiseless channel are

ideal case
مش موجودة بالواقع .

$$\boxed{\begin{aligned} SNR &= \frac{\text{signal power}}{0} = \infty \\ SNR_{dB} &= 10 \log_{10} \infty = \infty \end{aligned}}$$

→ noiseless channel

We can never achieve this ratio in real life; it is an ideal case.

3-5 DATA RATE LIMITS

*A very important consideration in data communications how fast we can send data, **in bits per second**, over a chann*
Data rate depends on three factors:
1. *The bandwidth available*
2. *The number of signal levels used*
3. *The quality of the channel (the level of noise)*

Shannon اكتشفوا

Topics discussed in this section:

* Noiseless Channel: Nyquist Bit Rate
* Noisy Channel: Shannon Capacity
* Using Both Limits

Noiseless Channel: Nyquist Bit Rate

- Defines the theoretical maximum bit rate possible for noiseless channels

- Nyquist Bit Rate = 2 x Bandwidth x $\log_2(L)$ Signal levels
  - Bit rate is the maximum bit rate in bps
  - Bandwidth is the channel bandwidth in Hz
  - L is the number of signal levels used

How many cycles per second.

مثلاً لو بدي ابعث 0,1 الـ L بتكون بتساوي 2 و $\log_2 2 = 1$.

- Increasing the number of levels is not practical:
  - It imposes a burden on the receiver
  - It complicates the receiver design
  - It reduces the reliability of the system (i.e.; increase the probability of reception errors)

*حيتأثر استقبال الـ reciever للـ signal. (كبير بكون)

*أيضًا بكون فيه noise.

Note

Increasing the levels of a signal may reduce the reliability of the system.

Example 3.33

Does the Nyquist theorem bit rate agree with the intuitive bit rate described in baseband transmission?

Solution

They match when we have only two levels. We said, in baseband transmission, the bit rate is 2 times the bandwidth if we use only the first harmonic in the worst case. However, the Nyquist formula is more general than what we derived intuitively; it can be applied to baseband transmission and modulation. Also, it can be applied when we have two or more levels of signals.

Example 3.34

Consider a noiseless channel with a bandwidth of 3000 Hz transmitting a signal with two signal levels. The maximum bit rate can be calculated as

$$\boxed{\text{BitRate} = 2 \times 3000 \times \log_2 2 = 6000 \text{ bps}}$$

Example 3.35

Consider the same noiseless channel transmitting a signal with four signal levels (for each level, we send 2 bits). The maximum bit rate can be calculated as

$$\text{BitRate} = 2 \times 3000 \times \log_2 4 = 12{,}000 \text{ bps}$$

Example 3.36

We need to send 265 kbps over a noiseless channel with a bandwidth of 20 kHz. How many signal levels do we need?

__*Solution*__

We can use the Nyquist formula as shown:

$$265{,}000 = 2 \times 20{,}000 \times \log_2 L$$

$$\log_2 L = 6.625 \qquad L = 2^{6.625} = 98.7 \text{ levels}$$

2^7 ← قريبه لأقرب عدد

Since this result is not a power of 2, we need to either increase the number of levels or reduce the bit rate. If we have 128 levels, the bit rate is 280 kbps. If we have 64 levels, the bit rate is 240 kbps.

Noisy Channel: Shannon Capacity

- Defines the theoretical maximum bit rate possible for noisy channels *كلما زاد ال Bandwidth
- Channel Capacity = Bandwidth x $\log_2(1+\text{SNR})$ كل مازاد ال data Rate
  وكلما زاد ال SNR كل مازاد ال bit Rate
  - * Capacity is the maximum possible bit rate in bps
  - * Bandwidth is the channel bandwidth in Hz
  - * SNR is the signal-to-noise power ratio ← مش بال dB
- Defines a channel characteristic not the method of transmission

Example 3.37

Consider an extremely noisy channel in which the value of the signal-to-noise ratio is almost zero. In other words, the noise is so strong that the signal is faint. For this channel the capacity C is calculated as

باهتة أو غير موجودة.

$$\boxed{C = B \log_2 (1 + \text{SNR}) = B \log_2 (1 + 0) = B \log_2 1 = B \times 0 = 0}$$

لـ مش قادر ابعت اشي رغم ال Bandwidth العالية والسبب انه SNR=0 .

This means that the capacity of this channel is zero regardless of the bandwidth. In other words, we cannot receive any data through this channel.

Example 3.38

We can calculate the theoretical highest bit rate of a regular telephone line. A telephone line normally has a bandwidth of 3000 Hz. The signal-to-noise ratio is usually 3162. For this channel the capacity is calculated as

$$C = B \log_2 (1 + \text{SNR}) = 3000 \log_2 (1 + 3162) = 3000 \log_2 3163$$
$$= 3000 \times 11.62 = 34{,}860 \text{ bps}$$

* ممكن تبعت أقل من هيك بس مش ممكن تبعت أكثر من هيك.

This means that the highest bit rate for a telephone line is 34.860 kbps. If we want to send data faster than this, we can either increase the bandwidth of the line or improve the signal-to-noise ratio.

97

Example 3.39

The signal-to-noise ratio is often given in decibels. Assume that $SNR_{dB} = 36$ and the channel bandwidth is 2 MHz. The theoretical channel capacity can be calculated as

$$\text{SNR}_{\text{dB}} = 10 \log_{10} \text{SNR} \rightarrow \text{SNR} = 10^{\text{SNR}_{\text{dB}}/10} \rightarrow \text{SNR} = 10^{3.6} = 3981$$
$$C = B \log_2 (1 + \text{SNR}) = 2 \times 10^6 \times \log_2 3982 = 24 \text{ Mbps}$$

98

Example 3.40

لو ال SNR كانت كبيرة بنقدر نقرب باستخدام ال formula الي تحت

For practical purposes, when the SNR is very high, we can assume that SNR + 1 is almost the same as SNR. In these cases, the theoretical channel capacity can be simplified to

* بالامتحان بحددلك تقرب أو لأ *

$$\boxed{C = B \times \frac{SNR_{dB}}{3}}$$

For example, we can calculate the theoretical capacity of the previous example as

$$\boxed{C = 2\ \text{MHz} \times \frac{36}{3} = 24\ \text{Mbps}}$$

Example 3.41

We have a channel with a 1-MHz bandwidth. The SNR for this channel is 63. What are the appropriate bit rate and signal level?

Solution

First, we use the Shannon formula to find the upper limit.

$$\boxed{C = B \log_2 (1 + SNR) = 10^6 \log_2 (1 + 63) = 10^6 \log_2 64 = 6\ \text{Mbps}}$$

Example 3.41 (continued)

The Shannon formula gives us 6 Mbps, the upper limit. For better performance we choose something lower, 4 Mbps, for example. Then we use the Nyquist formula to find the number of signal levels.

*استخدام ال 2 formulas مع بعض.

$$\boxed{4\text{ Mbps} = 2 \times 1\text{ MHz} \times \log_2 L \longrightarrow L = 4}$$

اخترنا شوي أقل من ال Max (6Mbps) عشان ما نضغط عالحفة.

*الحالة الأولى كان noise channel والثانية كان noiseless

Note

The Shannon capacity gives us the upper limit; the Nyquist formula tells us how many signal levels we need.

3-6 PERFORMANCE

*trade off → تضحي بشيء مقابل شيء / ← لازم تكون أعلى ما يمكن.

*One important issue in networking is the **performance** of the network—how good is it? We discuss quality of service, an overall measurement of network performance, in greater detail in Chapter 24. In this section, we introduce terms that we need for future chapters.*

Topics discussed in this section:

Bandwidth → unit: cycles/s (Hz) → بعدها بتصير → bits per second
Throughput
Latency (Delay)
Bandwidth-Delay Product

3.103

Note

In networking, we use the term bandwidth in two contexts.

- ← بسياق الكهرباء والاتصالات
  The first, bandwidth in hertz, refers to the range of frequencies in a composite signal or the range of frequencies that a channel can pass.
  حزمة الـ freq اللي ممكن الـ channel تمرقها أو إنها تكون موجودة بالـ Signal.

- ← سياق مختلف بمجال الاتصالات
  The second, bandwidth in bits per second, refers to the speed of bit transmission in a channel or link.

3.104

Example 3.42

he bandwidth of a subscriber line is 4 kHz for voice or ata. The bandwidth of this line for data transmission an be up to 56,000 bps using a sophisticated modem to hange the digital signal to analog.

شرحنا بمثال الصوت
انه من 4 kHz → 8 kHz →
→ 8 * 7 = 56

لأنهم 7 bits مو 8 bits → بالنظام الأمريكي
في النظام الأوروبي، آخر bit استخدموها
للـSynchronise

Example 3.43

If the telephone company improves the quality of the line and increases the bandwidth to 8 kHz, we can send 112,000 bps by using the same technology as mentioned in Example 3.42.

بوقف عند ال Sender وبعد كم Packet عم ببعت Per second

بوقف عند ال reciever وبعد ال (recieved Packets per Second) Completed Jobs

Throughput versus Bandwidth

لو في أحسن الأحوال ال Throughput = Bandwidth
لو في أسوأ الأحوال بكون الدي كبير جداً.

reference لـ الطالب.

- The Bandwidth is a potential measurement of how much we can send data over the link

data link layer على مستوى ال — أهم من ال Bandwidth

- The Throughput is the actual measurement of how fast we can send data over the link

- The throughput can be limited by the capabilities of the communicating devices or the availability of the network resources

* لو طلعت على مستوى ال App layer بصير بدل ال throughput ال "Goodput":
قليل الاستخدام لكنه موجود (بنقاس من ال App layer لـ App layer)

Example 3.44

A network with bandwidth of 10 Mbps can pass only an average of 12,000 frames per minute with each frame carrying an average of 10,000 bits. What is the throughput of this network?

Solution

We can calculate the throughput as

$$\boxed{\text{Throughput} = \frac{12{,}000 \times 10{,}000}{60} = 2 \text{ Mbps}}$$

The throughput is almost one-fifth of the bandwidth in this case.

Latency (or Delay)

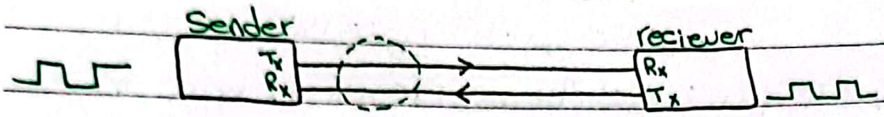
- The latency (or delay) defines the total time needed to transfer an entire message from source to destination

- The total delay consists of four components: → موجود بالسلايدات خارجية
  - Propagation delay = Distance / Propagation Speed
  - Transmission delay = Message Size / Bandwidth
  - Queuing delay: depends on the current network status
  - Processing delay: depends on the speed and capabilities of the nodes

- The Jitter defines the variations in the delay

23/10/2022

Slide 3.84 : مثال على نوع من ال noise



فبصير فيه noise . ← "بصير في مجال مغناطيسي" وا

Slide 3.99 : Proof :

$$C = B\left(\frac{SNR_{dB}}{3}\right) \rightarrow C = B\log_2(1+SNR) \rightarrow {}^*SNR_{dB} = 10\log_{10}SNR$$

ratio not dB ← الأصلية

$$\rightarrow SNR = 10^{\frac{SNR_{dB}}{10}} \rightarrow C = B\log_2\left(10^{\frac{SNR_{dB}}{10}}\right) = B\,\frac{SNR_{dB}}{10}\log_2 10 \quad \frac{\ln 10}{\ln 2}$$

$$= \frac{B\,SNR_{dB}}{\left[\frac{10}{\log_2 10}\right]} \rightarrow \approx 3$$

25/10/2022

* السلبيات الخارجية * ← مكونات ال Delay

* nodal → total delay

* Prop → يعتمد على ال media اللي انت موجود فيها (السرعة اللي بنتقل فيها ال Packet / bit من طرف إلى الآخر). ← "تعتمد عالمسافة والسرعة"

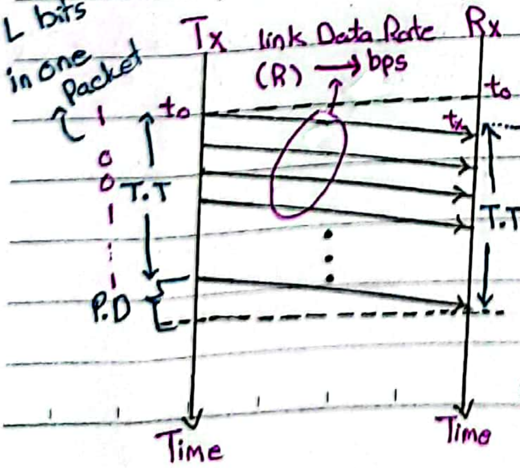
* Proc → بعدين بحددوين error check بعمل ع لاستقبال ال Packet في الأول وبعدين بحدد وين بدها تطلع

* trans → الزمن اللازم حتى نرسل كل ال bits الموجودة بال packet

* queueing → Random مش محدد ما بنقدر نتنبأ فيه وهو اللي عامل مشاكل (ال Varience المشكلة مش ال average).

زمن الانتقال من ال transmitter إلى ال reciever ، يعتمد عالمسافة والسرعة

* Space Time Diagram *



* Propagation Delay (P.D.) = $t_x - t_o$ = length of wire / ≈ speed of light

* Transmit Time (T.T.) = $L\left(\frac{1}{R}\right)$ → time to transmit one bit

لـ الزمن من tx لحد استلام اخر bit = $\frac{L}{R}$ Seconds ← حجم ال Packet

* Total time to transmit a packet = P.D + T.T

* هلأ إذا كانت الـ link سريعة بكون الـ Dominant Factor واللي بسيطر هو الـ P.D.

* بنمشي بالأمثلة تقريباً على سرعة الضوء.

* Caravan Analogy ← مثال لنعرف نحسب الـ Delay. (قديماً لتبعث الـ Packet ← [T.T + P.D ← هيك حتكون.
لـ 10 سيارة.

* Queueing Delay:

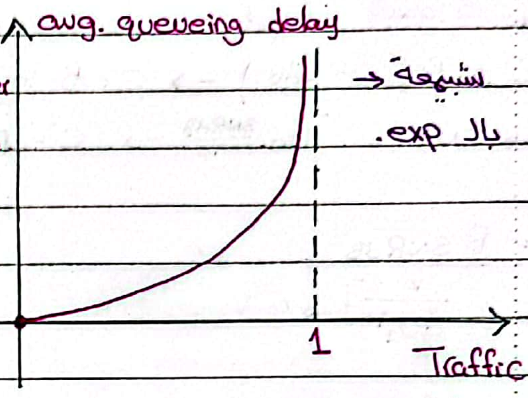
R → bandwidth

L → Packet size

a → السرعة اللي بتيجي فيها الـ Packet عالـ link

La / R → a] → Packet arrival rate. ←

Packet Service rate ← [$\frac{R}{L}$

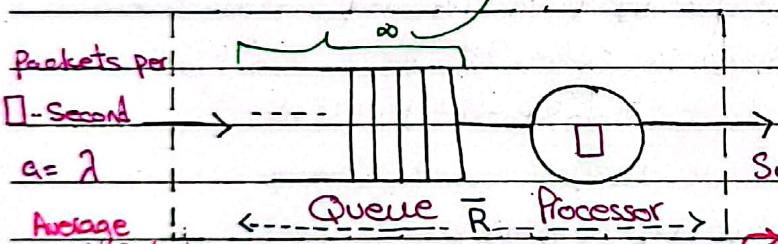


Traffic Intensity (U) ← La / R ← ما بالواحدة.

في اشي بيجي كذا (بدخل بسرعة معينة) وبطلع بسرعة معينة.

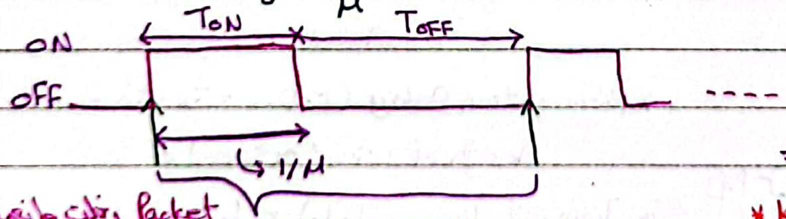
* Proof:

* Communication Link: افترضنا إنه ∞ لنسهل الـ Proof



λ = arrival rate (Packets/s) / μ: Service rate (Packets/s)

* traffic Intensity = $\frac{\lambda}{\mu}$ (Utilisation of the link (U)).



$\Rightarrow \frac{T_{ON}}{T_{ON}+T_{OFF}} = \frac{1/\mu}{1/\lambda} = \frac{\lambda}{\mu} = U$

Packet كامل بتاخد service.

$1/\lambda$ Packet Interarrival time (الزمن بين Packet و Packet).

* K: number of Packets in the link, (K-1) in the queue).

* $\mho$: fraction of time the link is busy.

* P_k : Probability of having exactly K packets in the link.

$\rightarrow P_k = \left(1-\frac{\lambda}{\mu}\right)\left(\frac{\lambda}{\mu}\right)^k$, $k=0,1,2,\dots$

$= (1-\mho)\,\mho^k \rightarrow$

* $k=0 \rightarrow P_0 = 1-\mho$ ← احتمال الـ queue يكون empty — P["link is empty"]

* $k=\infty \rightarrow P_\infty \propto (\uparrow)^\infty = 0$ وهنا منطق

← فحصت الـ boundary عشان أشوف لو الاشي منطقي.

fraction

queue without limit

← معناه احتمال يكون عندك عدد لانهائي من الـ Packets = 0 وهنا منطق.

* P["link is busy (فيو واحد أو أكثر)"] →

$= \sum_{k=1}^{\infty} P_k = \sum_{k=1}^{\infty} (1-\mho)\,\mho^k = 1-P_0$

30 / 10 / 2022

* Proof تكملة الـ :

* $\lambda \leq \mu \rightarrow$ system is stable (داخل قد اللي طالع)

* Condition for stability : $\lambda < \mu$ → random ما في مساواة لأنه احنا بناخذ

فالقيم (average) فمستحيل يكون اللي طالع قد اللي داخل. Packets

* إذا كان الـ arrival أكثر من اللي بيطلع معناه هذا الـ queue بنمو بشكل لانهائي (unstable).

لـ انفرض عنا غرفة بيدخل ويطلع منها أشخاص، لنفرض بفوت 5 أشخاص بالدقيقة

ويطلع 5 أشخاص بنفس الدقيقة فهيك مو مشكلة، بس لو كان اللي بفوتوا 6 واللي بطلعوا

5 فهاي مشكلة لأنه كل دقيقة في 5 بيتراكموا.

* Average : if $X = \{x_0, x_1, x_0, x_0, x_2, \dots\}$ → قيم عشوائية (N)

$\rightarrow \frac{x_0 + x_1 + x_0 + x_0 + x_2 + \dots}{N}$

مثلاً كم مرة تكررت x_2

$= \frac{N_0 x_0 + N_1 x_1 + N_2 x_2 + \dots}{N}$ → $N = N_0 + N_1 + N_2 + \dots$

* if N is very Large : $\lim_{N\to\infty}\left[\frac{N_0 x_0 + N_1 x_1 + N_2 x_2 + \dots}{N}\right]$

$= \boxed{\lim_{N\to\infty}\frac{N_0}{N}}\, x_0 + \lim_{N\to\infty}\frac{N_1}{N}\, x_1 + \lim_{N\to\infty}\frac{N_2}{N}\, x_2 + \dots$ — $P(x_0)$; Probability

$= P(x_0)\,x_0 + P(x_1)\,x_1 + P(x_2)\,x_2 + \dots$

* $\bar{X} = \sum_x x\,P(x)$ → الـ Average لحاله ما بساعد الـ Variance مهم أيضاً.

Mean
Average

$* \ \underbrace{P_0} + \underbrace{P_1 + P_2 + \cdots\cdots} = 1$

$\rightarrow P[\text{"link is empty"}] + P[\text{"link is busy"}] = 1$

$\rightarrow P[\text{"link is busy"}] = 1 - P_0 = 1 - (1 - \mho) = \mho$

$* \ P_\infty \rightarrow 0$

$\rightarrow P_k = \left(1 - \frac{\lambda}{\mu}\right)\left(\frac{\lambda}{\mu}\right)^k$

*أول Packet بتفوت بتشوف ال link فاضي الي بعدها حتشوفه busy وهكذا لحد K-1 ماحتدخل ال Sys إلا بعمليات .

* Let N be the number of packets in the link. We want to compute the average of packets ($\bar{N}$)

$\bar{N} = \sum_{k=0}^{\infty} k P_k = \sum_{k=0}^{\infty} k(1-\mho)\mho^k = (1-\mho)\sum_{k=0}^{\infty} k\mho^k$ → geometric Series

$= \cancel{(1-\mho)} \underbrace{\frac{\mho}{(1-\mho)^2}} = \frac{\mho}{1-\mho}$

من لحظة دخول ال Packet لل link لحد ما تطلع

$\bar{X}$: average throughput R: average response time	$\boxed{\bar{X} = \frac{\bar{N}}{R}}$

$\sum_{n=0}^{\infty} r^n = \frac{1}{1-r}, \ |r|<1 \rightarrow \frac{d}{dr}\sum_{n=0}^{\infty} r^n = \frac{d}{dr}\left[\frac{1}{1-r}\right] \Rightarrow \sum_{n=1}^{\infty} n r^{n-1} = \frac{1}{(1-r^2)}$

$\rightarrow \sum_{n=1}^{\infty} n r^n = \frac{r}{(1-r)^2}$

1/11/2022

$\boxed{* \ \bar{N} = \frac{\mho}{1-\mho}}$

$\bar{N}$: number of Packets (avg).
$\bar{R}$: Response time (avg). → موضحة برسمة الخلف
$\bar{X}$: Throughput (avg).

$\because \bar{X} = \frac{\bar{N}}{R}, \ \bar{R} = \bar{D} \rightarrow \bar{R} = \bar{D} = \frac{\bar{N}}{\bar{X}?}$

$\bar{X} = \text{Rate (Packets/s)} \ * P[\text{"link Busy"}] + \text{Rate (Packets/s)} \ P[\text{"link Idle"}]$

أسرع rate

$= \mu\mho + 0 * (1-\mho)$

$= \mu\mho = \mu\left(\frac{\lambda}{\mu}\right) = \underline{\underline{\lambda}}$

* μ is an upperbound for $\bar{X}$.

$\therefore \bar{D} = \frac{\bar{N}}{\bar{X}} = \frac{\mho}{1-\mho}\left(\frac{1}{\lambda}\right) = \frac{\frac{\lambda}{\mu}\left(\frac{1}{\lambda}\right)}{1-\mho} = \frac{1/\mu}{1-\mho} = \frac{\bar{S}}{1-\mho}$ → Service time (avg)

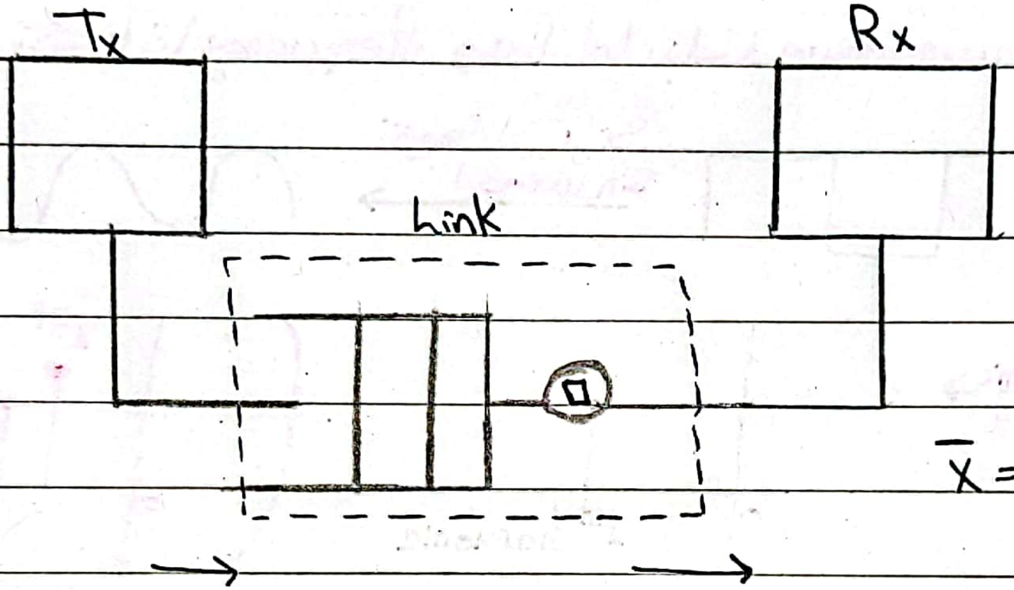
$\bar{S} = \frac{1}{\mu}$ (seconds).

$* \ \bar{D} = \frac{1}{\mu(1-\frac{\lambda}{\mu})} = \frac{1}{\mu - \lambda} \rightarrow \bar{D} = \frac{1}{\mu-\lambda}$

*لما λ تقترب من M ال Delay بيروح لـ ∞ الوضع بكون سيء .

3/11/2022

* ملاحظة مهمة :



$\bar{x} = \lambda$

← الـ Throughput هو أي شيء بطلع بعد منه الـ recieved packets .

← هون $\bar{x} = \lambda$ كيف الـ throughput اللي عم يطلع هو نفس اللي عم يدخل ؟ بالوضع الطبيعي اللي بيدخل في منه اشي بطلع loss واشي منه بكمل ، بس هون كل اللي بدخل بطلع لإنه الـ Size تبع الـ queue بيساوي ∞ ، أما لو كان N فكان حيختلف الـ throughput .

* الـ Size of the buffer بالعادة بختاروه نفس حجم الـ bandwidth delay product .

Example 3.45

What is the propagation time if the distance between the two points is 12,000 km? Assume the propagation speed to be 2.4×10^8 m/s in cable.

Solution

We can calculate the propagation time as

$$\boxed{\text{Propagation time} = \frac{12{,}000 \times 1000}{2.4 \times 10^8} = 50 \text{ ms}}$$

The example shows that a bit can go over the Atlantic Ocean in only 50 ms if there is a direct cable between the source and the destination.

Example 3.46

What are the propagation time and the transmission time for a 2.5-kbyte message (an e-mail) if the bandwidth of the network is 1 Gbps? Assume that the distance between the sender and the receiver is 12,000 km and that light travels at 2.4×10^8 m/s.

Solution

We can calculate the propagation and transmission time as shown on the next slide:

Example 3.46 (continued)

$$\text{Propagation time} = \frac{12{,}000 \times 1000}{2.4 \times 10^8} = 50 \text{ ms}$$

$$\text{Transmission time} = \frac{2500 \times 8}{10^9} = 0.020 \text{ ms}$$

Note that in this case, because the message is short and the bandwidth is high, the dominant factor is the propagation time, not the transmission time. The transmission time can be ignored.

Example 3.47

What are the propagation time and the transmission time for a 5-Mbyte message (an image) if the bandwidth of the network is 1 Mbps? Assume that the distance between the sender and the receiver is 12,000 km and that light travels at 2.4×10^8 m/s.

Solution

We can calculate the propagation and transmission times as shown on the next slide.

Example 3.47 (continued)

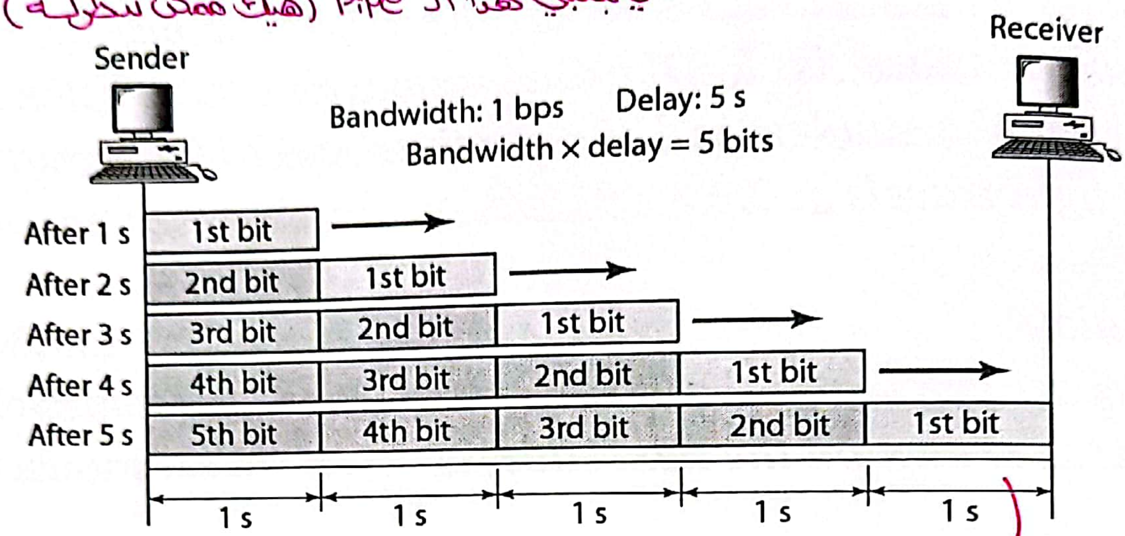
$$\text{Propagation time} = \frac{12{,}000 \times 1000}{2.4 \times 10^8} = 50 \text{ ms}$$

$$\text{Transmission time} = \frac{5{,}000{,}000 \times 8}{10^6} = 40 \text{ s}$$

Note that in this case, because the message is very long and the bandwidth is not very high, the dominant factor is the transmission time, not the propagation time. The propagation time can be ignored.

Figure 3.31 *Filling the link with bits for case 1*

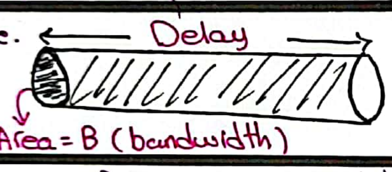
bits عدد ال ← # bits * Delay = bandwidth delay product
اللي بتعبي هذا ال Pipe (هيك ممكن نتخيله).



بعد 5 ثواني بايب وصلت ال bit 1 الى ال Receiver.
* كل ال Pipe تعبت خلال فترة 5 ثواني.

بال Transport عبارة عن round trip time.

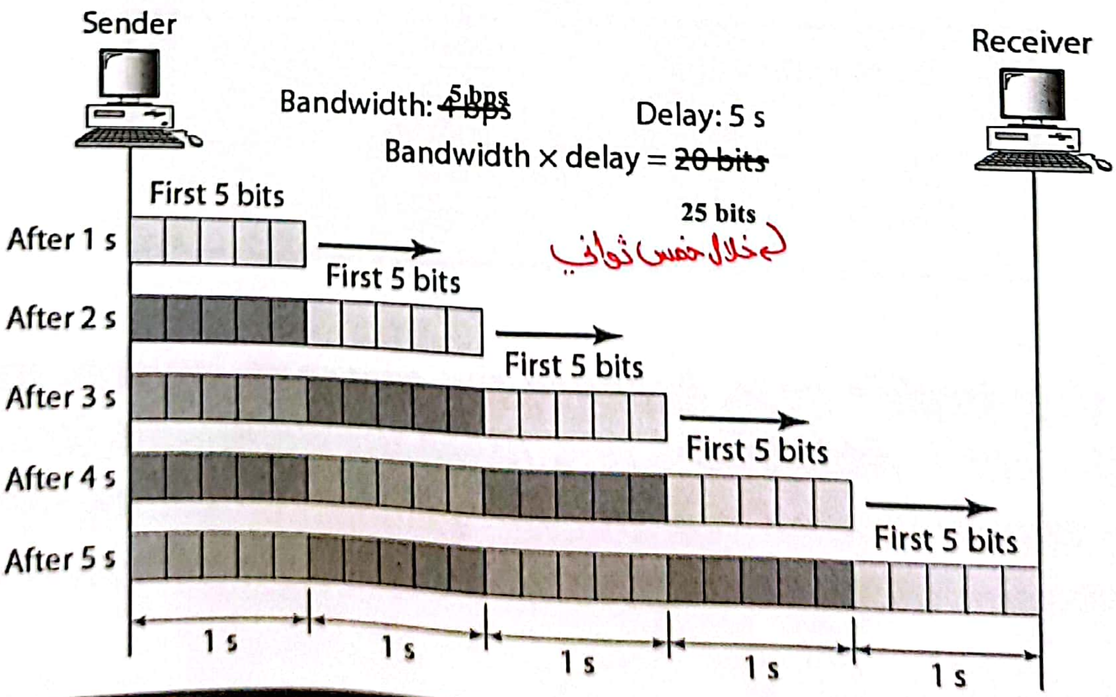
3.115



* ال Pipe capacity in bits/s :
Volume = B × D
BDP → Bandwidth delay product

* كل ما زاد ال Delay أو ال Bandwidth كل ما زاد ال Pipe volume.

Figure 3.32 *Filling the link with bits in case 2*



3.116

Note

The bandwidth-delay product defines the number of bits that can fill the link.

3.117

Chapter 4

Digital Transmission

تحويلات الـ Signal

↳ Digital to Digital

↳ Digital to Analog

وهكذا.

4-1 DIGITAL-TO-DIGITAL CONVERSION

*In this section, we see how we can represent digital data by using digital signals. The conversion involves three techniques: **line coding**, **block coding**, and **scrambling**. Line coding is always needed; block coding and scrambling may or may not be needed.*

Topics discussed in this section:

Line Coding
Line Coding Schemes
Block Coding
Scrambling

Line Coding

بسهولة إرسال ال Signal
باستخدام أمثل لل resources.

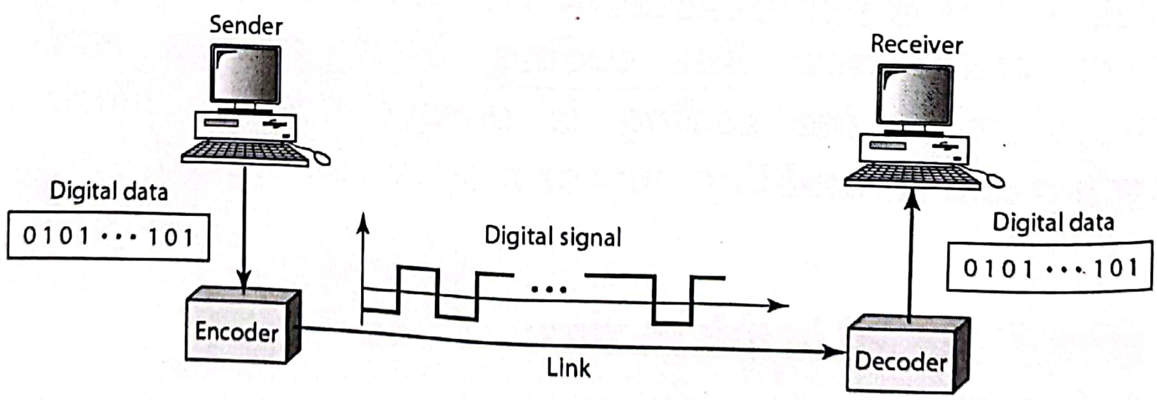
- Line coding is the process of converting digital data to digital signals (Digital to Digital Conversion)
- At the sender, data elements are encoded into signal elements
- At the receiver, signal elements are decoded into data elements

عمليات عكسية بين ال Sender وال reciever

* بنحتاجه مرات عشان ممكن تكون ال Digital الأولى مستهلكة bandwidth كثير فبحولها لـ form آخر من ال digital signal ما بستهلك bandwidth وهكذا.

* ال B بال Hertz وال Complexity وال power كله لازم ناخذه بعين الاعتبار.

Figure 4.1 *Line coding and decoding*



Characteristics of line coding

مجموعة من العوامل لازم ننتبهلها أثناء التحويل وتصميم ال line coding.

- Data Element vs. Signal Element
- Data Rate vs. Signal Rate
- Bandwidth → بمعنى ال Hertz (حزمة من الترددات) مش bps.
- Baseline Wandering
- DC Components → * غير مرغوب بال DC في ال Signal والـ line coding
  * ال frequency للـ DC component = 0.
  * لو ال digital signal فيها DC component بدي أرسلها ع channel ال B تبعها ما بمرق zero frequency حيصير عليها blocking فلازم تحولها من digital إلى digital ثاني ما فيه DC component
- Self-Synchronization
- Built-in Error Detection
- Immunity to noise and interference → مانع.
- Complexity → يكون بسيط.

* ال channels اللي ما بتمرق DC component مثلاً: ال transformers (كذا السيركت كهربائيا).

4.5

ال Tx لما يبعت بده يكون معه clock مشان يعرف وين بداية ونهاية ال bit وكذلك ال Rx (لازم يكونوا نفس ال clock وإلا بصير في أخطاء)

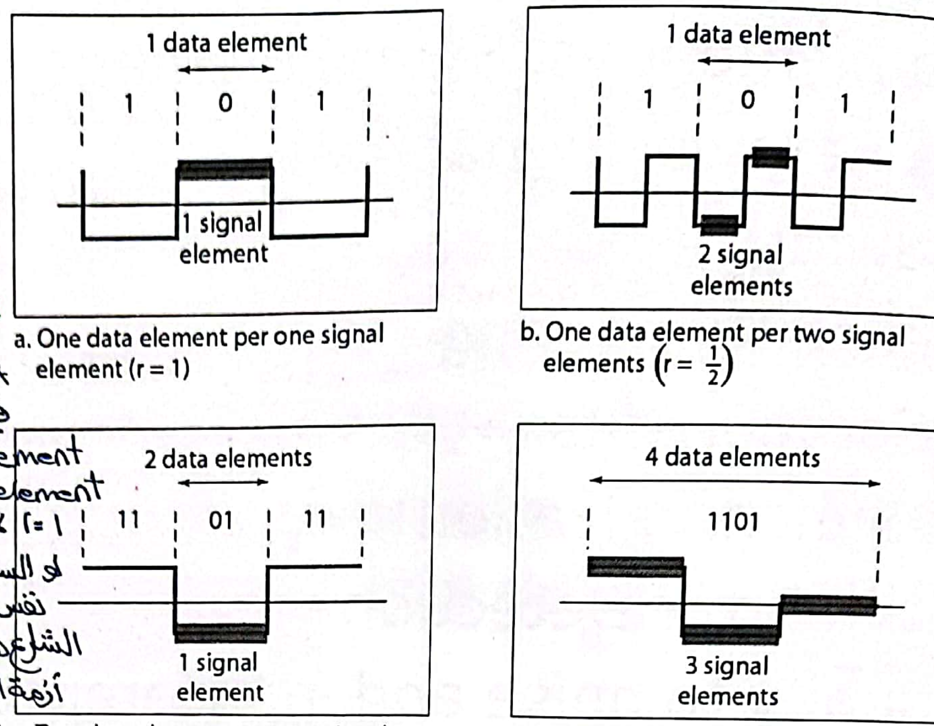
Signal Elements vs. Data Elements

- A data element is the smallest entity that can represent a piece of information (a bit) → " في حالتنا هي ال Bit, إما 0 وإما 1 "
- A signal element is the shortest unit in time of a digital signal (a baud) → أقصر قطعة بالزمن
- Data elements are what we need to send
- Signals elements are what we can send
- The ratio, r, is defined as the number of data elements carried by each signal element

* ممكن أكثر من Data element على Signal element. (بالدفتر).

4.6

Figure 4.2 *Signal element versus data element*



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

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

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

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

r = عدد ال data element نسبةً إلى ال signal element.

$r = \frac{\text{Data element}}{\text{Signal element}}$

* افرض احنا بسيارة وراكبين فيها أشخاص، الأشخاص data element والسيارة signal element، لو شخص واحد اللي راكب r = 1، ولو شخصين r = 2، لو السيارة وراها قاطرة وبيسوق نفس الشخص بتصير r = ½. الشارع هو ال channel - فاللي بيسوي أزمة السيارة مش الأشخاص فال signal rate هو اللي بأثر.

← هو اللي بأثر على ال bandwidth (يعني يكون قليل جدًا).

يعني يكون عالي.

Data Rate vs. Signal Rate

how many bps انت قاعد بتبعث.

- The data rate (or bit rate), N, is the number of data elements (bits) sent in one second
- The signal rate (or baud rate or pulse rate or modulation rate), S, is the number of signal elements (bauds) sent in one second ← or symbol

how many signal element تبعثها بالثانية.

- The goal is to increase the data rate (and hence the speed of transmission) while decreasing the signal rate (and hence the required bandwidth)

$$\boxed{S = c \times N \times \frac{1}{r}}$$

له عامل يعتمد على ال case اللي حتقوم بإرسالها.

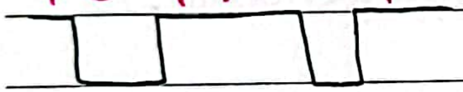
- Where c is the case factor that depends on the case

Chapter 4 :

Slide 4.8 :

11111111 → like DC component   Best case

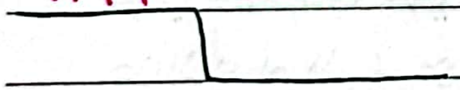
1 0 1 1 0 1



?

→ الSignal بتتغير حسب ال patterns.   Averge case (بين الحالتين)

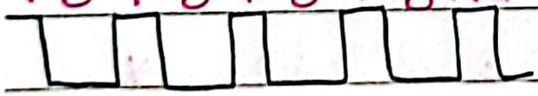
1111 0000



?

1 0 1 0 1 0 1 0 1 → أو العكس

↓ like squarewave   worst case (عشان كتير تقلبات)



Example 4.1

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

$$\boxed{S = c \times N \times \frac{1}{r} = \frac{1}{2} \times 100{,}000 \times \frac{1}{1} = 50{,}000 = 50 \text{ kbaud}}$$

* كلما r بنزيد بنزل ال Signal Rate *

4.9

The Bandwidth

- As learned previously, a digital signal that carries information is nonperiodic and its bandwidth is continuous and infinite
- Also, many of its components have very small amplitude that can be ignored

بالعادة كل ما نبعد عن ال low frequencies بنزل ال Amplitude بال fourier transform وهذا جيد لإنه انت بتحذف ال frequencies العالية فما رح يتضرر كثير شكل ال signals الأصلية فبنعمل save للbandwidth.

- Therefore, the "effective bandwidth" of the real-life digital signal is finite

لإنه بننسى الترددات العالية بس بناخذ الي ب range معين

- The bandwidth is proportional to the signal rate

الي ال power بتركز فيه بناخذهم.

- Given N: $B_{\min} = c \times N \times \frac{1}{r}$
- Given B: $N_{\max} = \frac{1}{c} \times B \times r$

4.10

* كلما بنزيد ال Data rate بنزيد ال Bmin *

Example 4.2

The maximum data rate of a channel (see Chapter 3) is $N_{max} = 2 \times B \times \log_2 L$ (defined by the Nyquist formula). Does this agree with the previous formula for N_{max}?

# of data elements
Signal element(s)

Solution

A signal with L levels actually can carry $\log_2 L$ bits per level. If each level corresponds to one signal element and we assume the average case ($c = 1/2$), then we have

$$\boxed{N_{max} = \frac{1}{c} \times B \times r = 2 \times B \times \log_2 L}$$

4.11

Note

Although the actual bandwidth of a digital signal is infinite, the effective bandwidth is finite.

* ال receiver لما يستقبل الإشارة بحدد
اذا هي 0 ولا 1 فبحسب ال avg
Power في ال Signal
(running/time avg)

Baseline Wandering

يتجول.

- While decoding the received signal, the receiver, calculates the running average of the received signal power, called the "baseline"
- The incoming signal power is evaluated against the baseline in order to determine the value of the data element
- Long strings of 0's or 1's can cause a drift of the baseline, called "baseline wandering", which makes it hard for the receiver to correctly decode the signal
- A good line coding scheme should prevent baseline wandering → لازم تكون بتعالج هاي النقطة

good line coding لازم ما يكون فيه DC component
عشان مرات ال transmittion line
بتكون طويلة فبالغالب بكون عنا low pass
effect ومرات بحتاج أعمل transformer (أفصل
أجزاء كهربائية بالسيركت عن بعضها) فهذا بعمل
block لل DC ، أما بال LAN عادي يكون في DC component.

DC Components

- Long constant voltage levels (DC) in digital signals create low-frequency components
- DC components are mostly filtered out in systems that can't pass low frequencies
- A good line coding scheme should have no DC components

متى بتبدأ ال bit ومتى بتنتهي ال bit.

عند ال Sender وعند ال reciever لازم يكونوا نفس الأشي لأنه لو اختلفوا حيصير الاستقبال كله خاطئ.

Self-synchronization

- The sender and receiver must be exactly synchronized in order for the received signals to be interpreted correctly

  عكسه asynchronized ما بعتمد ع clock s بعتمد ع Start and Stop bit.

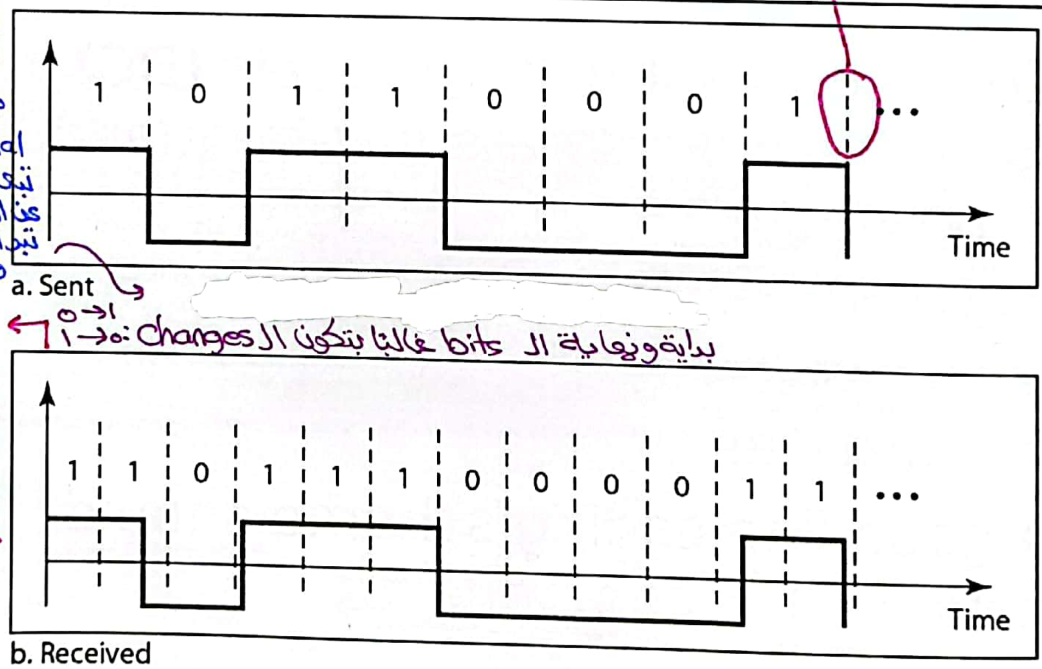
  كلما تزيد المسافة كلما بتصير معقدة أكثر

- Bit duration, the start, and the end of the bits should be exactly identified by the receiver
- If the sender and the receiver are running at different clock rates, the bit intervals will not match and the receiver may misinterpret the signals
- A good "self-synchronized" line coding scheme should keep the receiver well-synchronized

Figure 4.3 *Effect of lack of synchronization*

duration of the bit

الحل نخلي الخطوط المتقطعة matching مع بعض، نفس ال Signal الي باعتها نبعث معها معلومات عن ال clock او متى تبدأ ال bit ومتى تنتهي

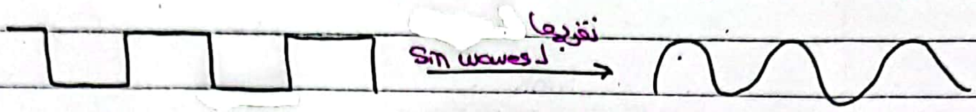


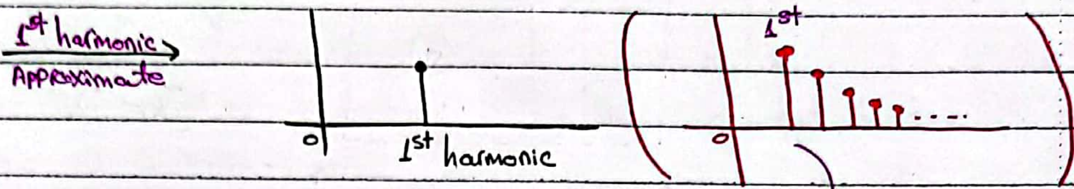
بداية ونهاية ال bits غالبا بتكون ال changes: 0→1 1→0

تغيير الفترات فتأثر استقبال ال bits، فاكد بستلم data مختلفة تماماً عن الي تم إرسالها

Slide 4.10 : $B_{min} = C \times N \times \frac{1}{r}$

* بدي الـ Signal تكون أقل frequency وهنا لما بناخذ Squarewave :



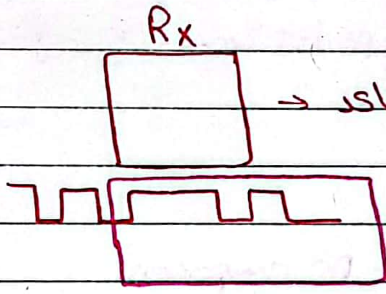


أخذنا فقط الـ 1st harmonic (Min frequency)

فلويش عرفنا ليه حط بالمعادلة Min ، وحط Bandwidth لأنه base band ← ماشي

من الـ zero للوصول للـ first harmonic) باعتبار إنها Periodic)

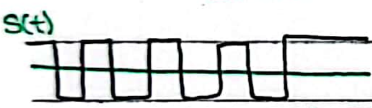
Slide 4.13 :



* الـ Reciever بحسب الـ avg عشان يعرف إنه هاي الـ Volt لو إجا اشي فوقها ..... يكون 1 ولو إجا اشي أقل منها بعتبرو 0 عشان يقدر يميز الـ bit.

بستقبل قادر ←

→ time window to compute the time avg.



لـ لنفرض دخلت لنا هاي الـ Signal ← Squarewave :

$\hookrightarrow S(t) = \frac{1}{T}\int_{t_0}^{t_0+T} s(\tau)\,d\tau$ (مجموع القيم على عددها)

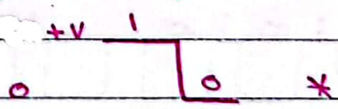
افرض ما إجاني هنا الـ pattern اجاني هيك :

اجاني sequence طويل

من الـ 0 أو الـ 1 ← حيخربط

الـ value avg اللي بحسبها الـ Reciever. (لو كانوا 0 كان الأخضر حينزل مش يطلع).

← هيك طريقة غلط لأنه حيتخربط الـ Reciever لازم أعمل طريقة للـ incoding بإنه ما أميز إنه الـ 1=+V والـ 0=0



Example 4.3

← مثال بيبينلك قديه الاختلاف ممكن يسبب أخطاء.

In a digital transmission, the receiver clock is 0.1 percent faster than the sender clock. How many extra bits per second does the receiver receive if the data rate is 1 kbps? How many if the data rate is 1 Mbps?

Solution

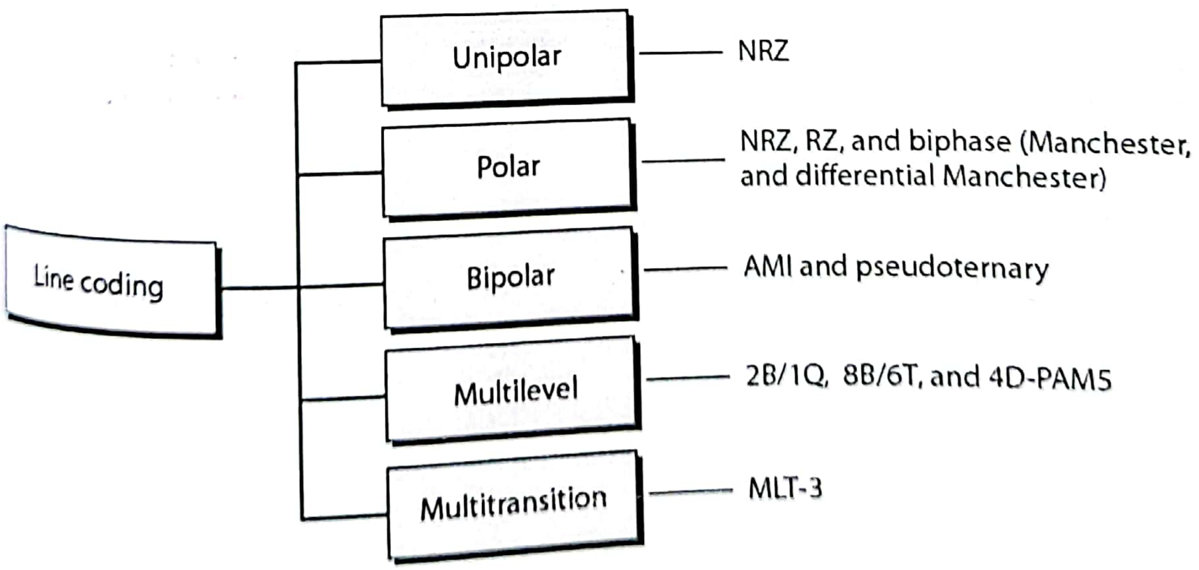
At 1 kbps, the receiver receives 1001 bps instead of 1000 bps.

1000 bits sent	1001 bits received	1 extra bps

At 1 Mbps, the receiver receives 1,001,000 bps instead of 1,000,000 bps.

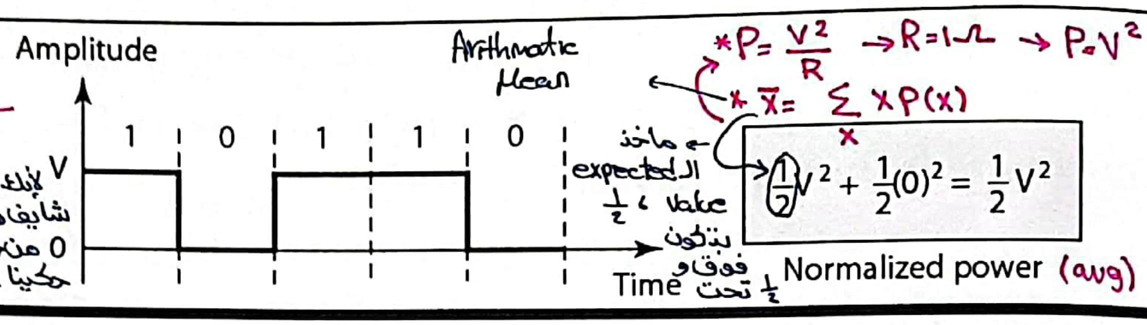
1,000,000 bits sent	1,001,000 bits received	1000 extra bps

Figure 4.4 *Line coding schemes* → " Line Coding أنواع الـ "



← لإنك خلال فترة استقبال ال bit ما بترجع لـ Zero بتضلك بنفس level الاستقبال ال Signal

Figure 4.5 *Unipolar NRZ (Non-Return to Zero) scheme*



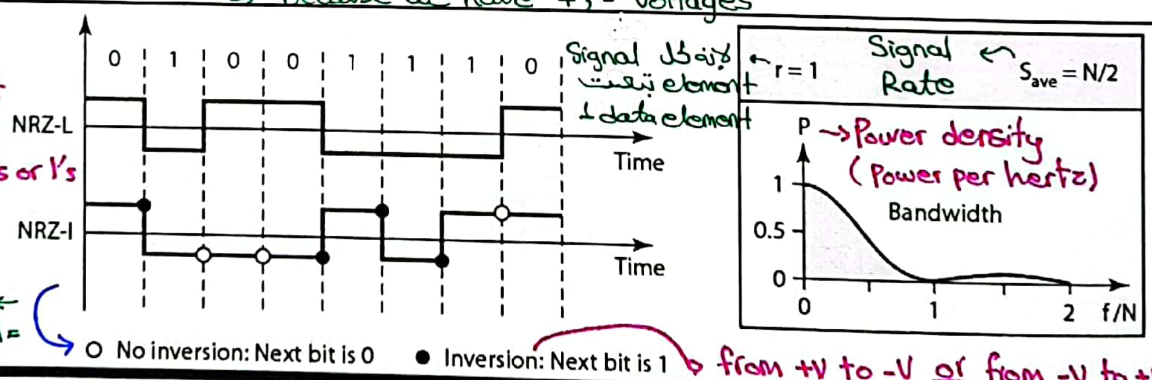
← مشاكله:
1- Baseline wandering → لإنك شايفهون Snapshot منحالة معينة بس 0 حكينا إنه في cases، فلو اجا مثلاً Sequence طويل من ال 0's حتستقبلوا كإنها DC ولو 1 كله V.
2- بتصرف Power

- Unipolar: All signal levels at one side of the time axis → مافيه negative أبداً
- NRZ since the signal does not return to zero in the middle of the bit → لإنك مابترجع لـه خلال فترة إرسال ال bit
- Relatively very costly scheme in terms of normalized power → مكلف بسبب ال Power
- Hence, not a popular scheme nowadays
  ← مش منتشر حالياً كثير

*logic 0 = V / logic 1 : -V

Figure 4.6 *Polar NRZ-L (NRZ-Level) and NRZ-I (NRZ-Invert) schemes*
↳ because we have +, - Voltages

الغرض منها ← تقليل ال DC component، بس لو اجا Sequence 0's or 1's مارح تقدر تحلها.



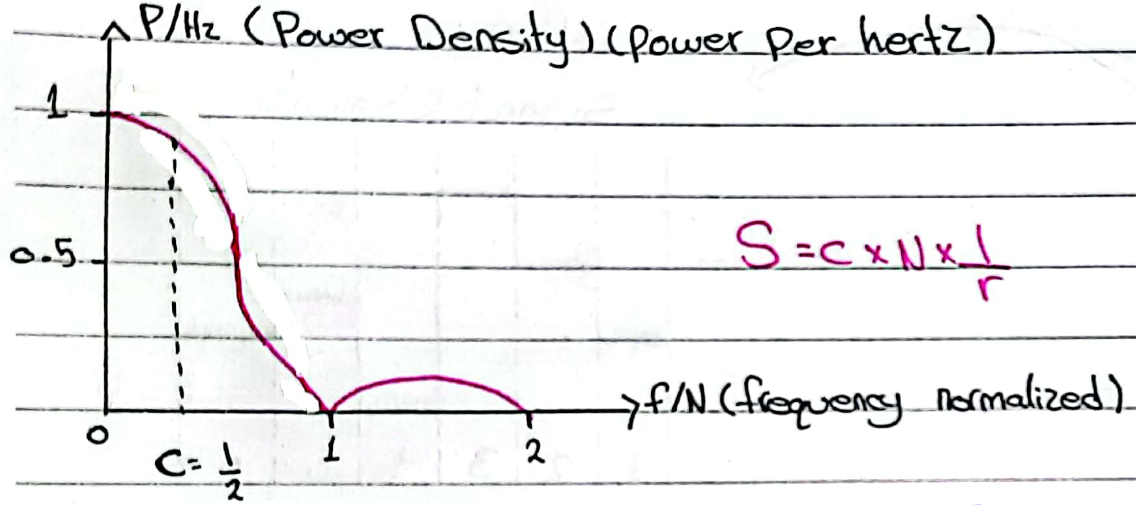
← انا ال Next bit 1 = بعمل inversion و لكن انا 0 مابعمل Inversion (يعتمد على ال Next bit)

- Polar: The signal levels are on both sides of the time axis
- NRZ-L: the voltage level determines the value of the bit
- NRZ-I: the change or lack of change determines the value
- Baseline Wandering and Synchronization problems: in both, but twice as severe in NRZ-L
  ← خففنا من مشكلة ال Baseline wandering في حالة ال Sequence of 1's
- Both have the DC Component problem

$*S = c \times N \times \frac{1}{r}$, $r = 1$, $c = ?$ (بالدفتر)

6/11/2022

Slide 4.20



$$S = c \times N \times \frac{1}{r}$$

$\underbrace{B_{min}}_{\to\ f}$ $= C \times N \times \frac{1}{r}$ ← $r=1$ → $f = C \times N$ → $\frac{f}{N} = C$ (case path)

↳ f اعتبرناها

"1st harmonic" باستخدام

* لو أخذت Alternating one's or zero's هتكون ال DC component = 0
* لو C=1 (worst case senario) هيكون ال DC = 0
* لو $C = \frac{1}{2}$ ← (نمثل ال data rate) $S_{avg} = \frac{1}{2} \times N = \frac{N}{2}$

↳ كلما نزيد ال data rate كلما بنزيد ال avg rate وهذا senario مقبول تقريبًا. (فيه ال bandwidth عالي)

Note

NRZ-L and NRZ-I both have an average signal rate of N/2 Bd.

Baud

4.21

Note

NRZ-L and NRZ-I both have a DC component problem.

4.22

Example 4.4

A system is using NRZ-I to transfer 1 Mbps data. What are the average signal rate and minimum bandwidth?

Solution

The average signal rate is S = N/2 = 500 kbaud. The minimum bandwidth for this average baud rate is B_{min} = S = 500 kHz.

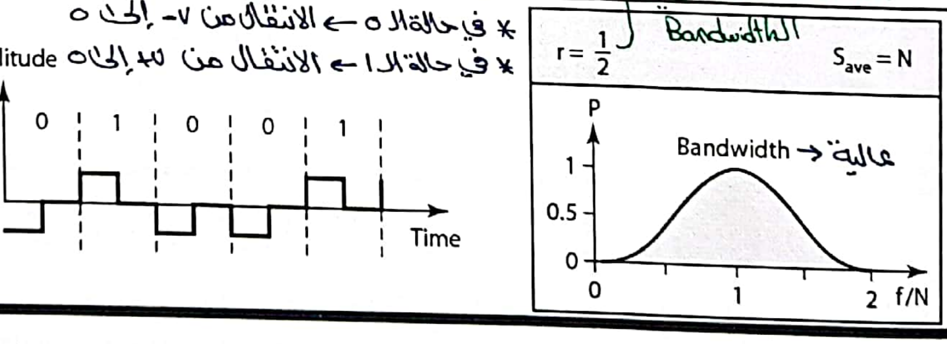
← لإنه أثناء فترة إرسال ال bit انت بنتقل من V ل V- أخرى.

كل 2 Signal element يبعث 1 data وهنا بزيد ال element Bandwidth

Figure 4.7 ***Polar RZ (Return-to-Zero) scheme***

* في حالة ال 0 ← الانتقال من V- إلى 0
* في حالة ال 1 ← الانتقال من V+ إلى 0

*هون ما في DC Component لإنك نص المدة فوق ونص تحت. بس في مشكلة ال bandwidth العالي.



- RZ: the signal changes at the beginning and in middle of bit
- Solves the synchronization problem in NRZ schemes ← لإنه صرت اعرف متى برسل وبنتقل
- No DC Component problem
- Needs two signal levels to encode a bit → more bandwidth
- Uses 3 signal levels → more complex
- Not a popular scheme

ال bit من خلال ال change تبع ال Signal لإنه موجود ما بيعرف إنه هو حالياً بنص ال bit

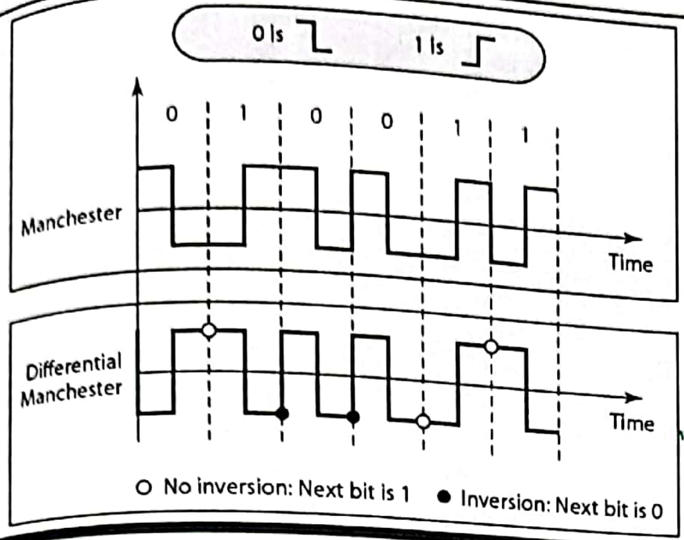
أصعب على ال reciever تصميمه.

مستخدم في ال Ethernet

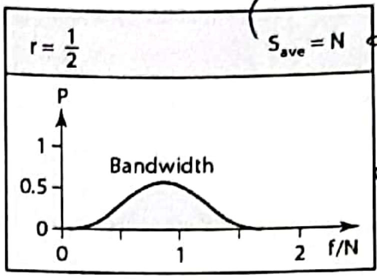
Figure 4.8 ***Polar biphase: Manchester and differential Manchester schemes***

في return-to-zero

(انتقال في منتصف ال bit)



سيئ لإنه يكون ال (Signal rate) avg rate هو نفسه ال data rate



(مناسب نستخدمه في ال LAN فقط لإنه ال Bandwidth كثير عريض فبمرق ال Data ↓. أقل Power بس لسا Bandwidth عالي.

"2 levels of signal"
لأحسن من 3

- Overcome the problems of NRZ-L & NRZ-I
- The cost is doubling the signal rate and hence the minimum bandwidth

4.25

Note

In Manchester and differential Manchester encoding, the transition at the middle of the bit is used for synchronization.

4.26

Note

The minimum bandwidth of Manchester and differential Manchester is 2 times that of NRZ.

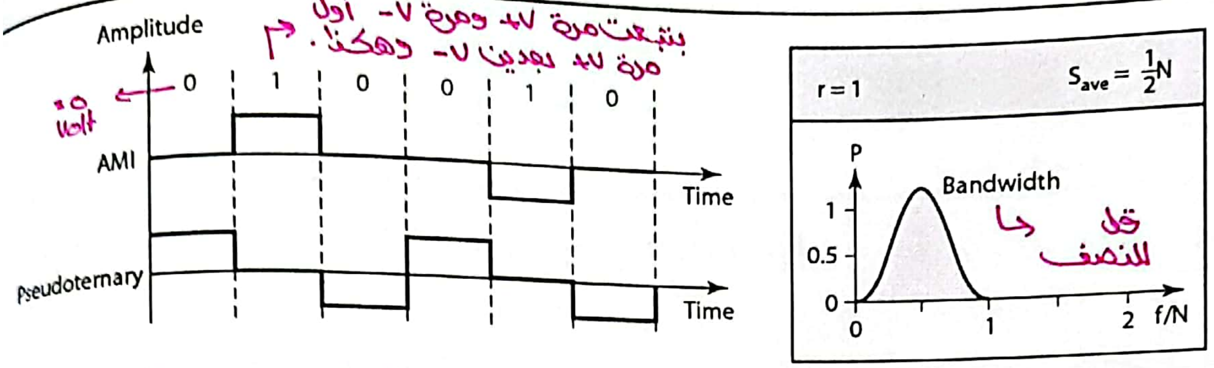
هاي تعتبر سيئة.

Note

In bipolar encoding, we use three levels: positive, zero, and negative.

لأنه في قطبيتين ← +V ← -V

Figure 4.9 Bipolar schemes: AMI and pseudoternary



- AMI (Alternate Mark Inversion) = Alternate 1 Inversion
- AMI: 0 = zero volt, 1 = alternating positive and negative
- Pseudoternary: 1 = zero volt, 0 = alter. pos. and neg. → عكس AMI
- No DC component. Why? → because avg = 0
- What about synchronization? → AMI → بحلها لو كانت 1 لو 0 لا
  → Pseudoternary → بحلها بمرحلة ال 0 بدل 1 لا

4.29

* لو احنا Sequence of 0's بيكون كله = 0 لو 1's بيكون نص الوقت فوق ونصه تحت فبلغي ال Baseline wandering .

* ال Synchronization بتكون مضرة بالعادة لما يكون كل ال Signal 0 أو 1 لأنه بضيع بداية ونهاية ال bit

Multi-Level Line Encoding Schemes (mBnL)

- The goal is to increase the number of bits per baud by encoding a pattern of m DE's into a pattern of n SE's → لكل Symbol بيبعث عدد كبير من ال bits كل ما زاد كل ما كان أحسن.
- Binary data elements ➔ 2^m data patterns
- L signal levels ➔ L^n signal patterns
- If $2^m = L^n$ ➔ each data pattern has a signal pattern
- If $2^m < L^n$ ➔ data patterns form a subset of the signal patterns
  - Extra signal patterns can be used for better synch. and error detection
- If $2^m > L^n$ ➔ not enough signal patterns (not allowed)
- mBnL ➔ Binary patterns of length m, signal patterns of length n with L signal levels → بتنكتب بهذه الطريقة .
- L can be replaced by:
  - B (Binary) ➔ L=2 (e.g.; 2B2B)
  - T (Ternary) ➔ L=3 (e.g.; 8B6T)
  - Q (Quaternary) ➔ L=4 (e.g.; 2B1Q)

4.30

8/11/2022

Slide 4.30 :

Data Elements

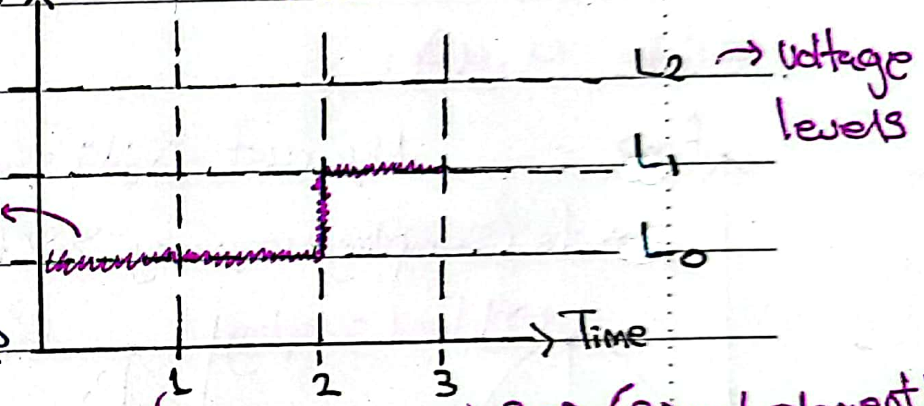
* m bits = نفرض انو
* compinations = 2^m

لـ مثلاً عندي 2 bits ← $4 = 2^2$

↳ 00, 01, 10, 11

Signal Element

Amplitude *L = # of levels الـ نفرض انو*



3=n (Signal element)

بقدر أمثله بـ Path من الـ Signal element

* if $2^m = L^n$ → كل الـ compinations من الـ Data element
* if $2^m < L^n$ → عادي بكون عندك Signal levels زيادة
* if $2^m > L^n$ → ما بنفع عندك مشكلة لأنه مش كل الـ compinations اللي بالـ Data element حقدر أمثلهم.

*Compinations = L^n ↴

كل الـ compinations اللي ممكن نرسمها. ممكن يستخدموا للـ error ↳ detection أو الـ Synchronisation أو هيك.

Five Apple

Note

In *mBnL* schemes, a pattern of *m* data elements is encoded as a pattern of *n* signal elements in which $2^m \leq L^n$.

حالتين مزبوطة وحالة الأكبر مش مزبوطة.

4.31

Signal element

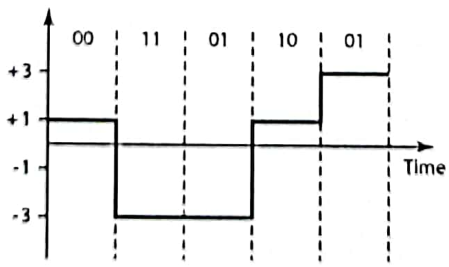
2 data element encodded in 1 SE

L=4 levels

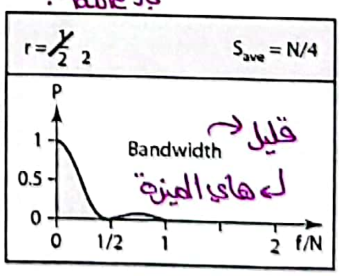
Figure 4.10 *Multilevel: 2B1Q (2 Binary 1 Quaternary) scheme*

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

Transition table → لازم نلتزم بال table.



Assuming positive original level



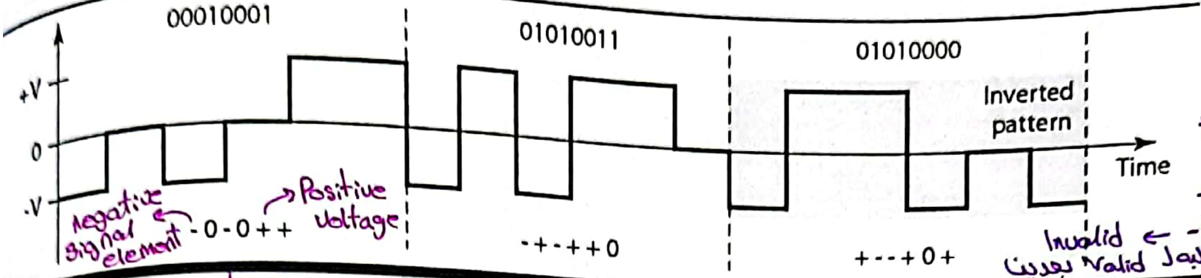
مشكلة

↑L → ↑complex receiver

- Faster data rate. Why? because r=2
- L = 4 → complex receiver & $2^2 = 4^1$ → no extra signal patterns
- What about the DC Comp.? Not balanced (scrambling used to help) ← لسا موجودة
- What about synchronization and baseline wandering? (same)

4.32

Figure 4.11 *Multilevel: 8B6T (Eight Binary 6 Ternary) scheme* ← Ethernet بيستخدموها



+--+0+ ↓ -++-0-

لو = -1 ← Invalid
تقلبه بعدين Valid

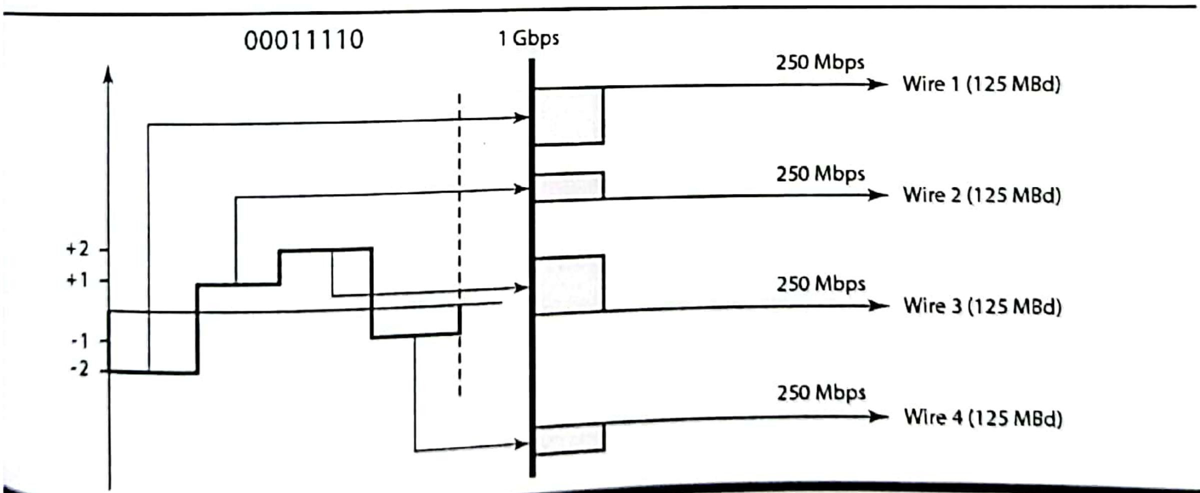
ولو بعض +

- $3^6 - 2^8 = 473$ extra signal patterns ➔ good synchronization and error detection → Valid data (Send or receive) if it is negative (Invalid)
- Each pattern has a weight of either 0 or +1 DC values
- If two consecutive patterns with +1 weight, the sender inverts the second pattern. Why? to reduce baseline wandering
- How does the receiver recognize the inverted pattern?
- $S_{avg} = ½ \times N \times 6/8 = 3N/8$

*بيقلل الـ Baseline wandering.

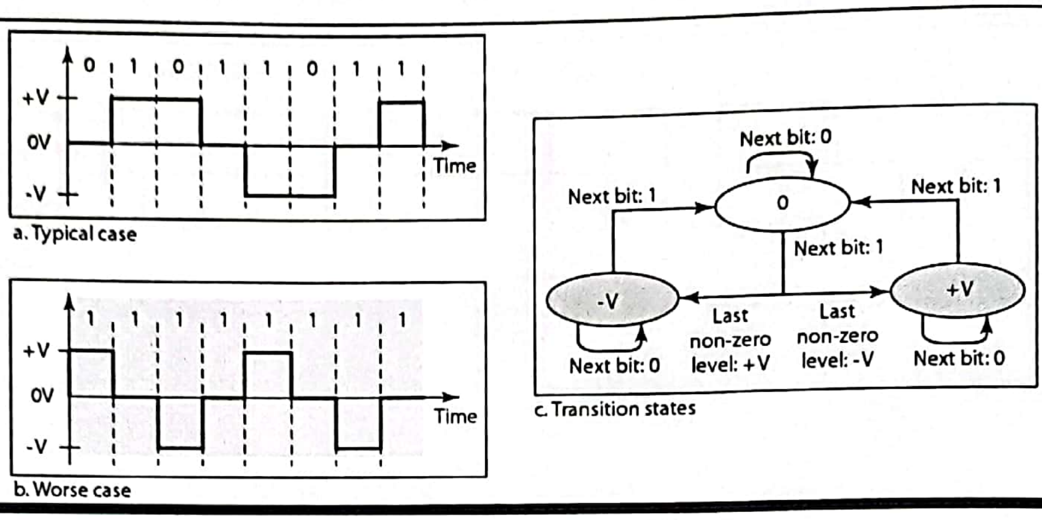
*بـ Appendix D موجود الجدول اللي بيمثل كل الـ Possibility الموجودة.

Figure 4.12 *Multilevel: 4D-PAM5 scheme*



- 4-dimentional five-level pulse amplitude modulation
- Level-0 is only used for error-correction
- Equivalent to 8B4Q scheme with $S_{max} = N \times 4/8 = N/2$
- Four signal are sent simultaneously over four different wires
- $S_{max} = N/8$
- Used in Gigabit LANs over 4 copper cables

Figure 4.13 *Multi-transition: MLT-3 scheme*



- If next bit is 0 ➔ no transition (problematic long string of 0's)
- If next bit is 1 and current level is not zero ➔ next level is 0
- If next bit is 1 and current level is zero ➔ alternate the nonzero level
- The worst case is a periodic signal with period of 4/N
- ➔ S_{max} = N/4

Table 4.1 *Summary of line coding schemes*

Category	*Scheme*	*Bandwidth (average)*	*Characteristics*
Unipolar	NRZ	$B = N/2$	Costly, no self-synchronization if long 0s or 1s, DC
Unipolar	NRZ-L	$B = N/2$	No self-synchronization if long 0s or 1s, DC
	NRZ-I	$B = N/2$	No self-synchronization for long 0s, DC
	Biphase	$B = N$	Self-synchronization, no DC, high bandwidth
Bipolar	AMI	$B = N/2$	No self-synchronization for long 0s, DC
Multilevel	2B1Q	$B = N/4$	No self-synchronization for long same double bits
	8B6T	$B = 3N/4$	Self-synchronization, no DC
	4D-PAM5	$B = N/8$	Self-synchronization, no DC
Multiline	MLT-3	$B = N/3$	No self-synchronization for long 0s

Block Coding

m bits يعمل عملية استبدال لل
→ n>m حيث n bits بـ

- The process of stuffing the bit stream with redundant bits in order to:
  - Ensure synchronization
  - Detect errors
- The bit stream is divided into groups of *m* bits (called blocks)
- Each group is substituted with a different (usually larger) group of *n* bits (a code) → n > m
- This is referred to as *mB/nB* coding

لعشان بنستخدم
Block coding

Note

Block coding is normally referred to as *m*B/*n*B coding; it replaces each *m*-bit group with an *n*-bit group.

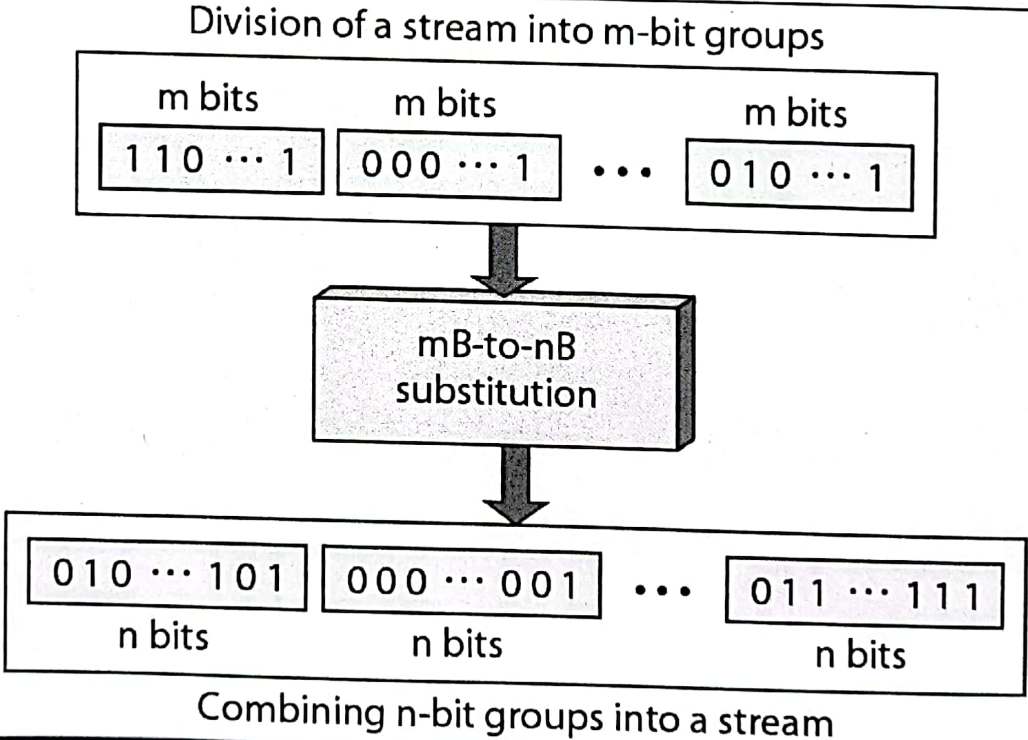
Steps in Block Coding Transformation

- Step 1: Division
  - The bit stream is divided into groups of m bits
- Step 2: Substitution
  - The m-bit groups are substituted with n-bit codes, where n>m
  - A number of n-bit codes are carefully chosen to ensure that the synchronization and error detection are achieved
  - Notice that at most only one half of the n-bit codes are needed. Why? because n is most larger than m

    آقل شي نزيد 1 bit .
- Step 4: Combination
  - The n-bit groups are combined together to form a new bit stream
- Step 3: Line Coding
  - A simple line coding scheme is used to convert the new bit stream into signals
  - No need for a complex line coding scheme since block coding ensures at least the synchronization

    بتحلها الـ Block coding

Figure 4.14 ***Block coding concept***



Common Block Codes

- 4B/5B Code → ضفنا 1bit زيادة
  - Every 4-bit block of data is substituted with a 5-bit codes
  - The 5-bit codes are line encoded with NRZ-I
  - Each code has no more than one leading 0 and no more than two trailing 0's (i.e.; no more than 3 consecutive 0's will ever be transmitted)
  - What about consecutive 1's? Why is it not handled?
  - 20% more bauds on NRZ-I due to using redundant bits
  - Unused codes provide a kind of error detection. How?
  - What about the DC component problem with NRZ-I?

1Mbps

4bits

Sub

5bits

? → 1.25 Mbps

"الrate اللي بتبعت فيه"

- 8B/10B Code
  - Same as 4B/5B except for the number of bits substituted
  - More codes are available for better error detection capability

Figure 4.15 *Using block coding 4B/5B with NRZ-I line coding scheme*

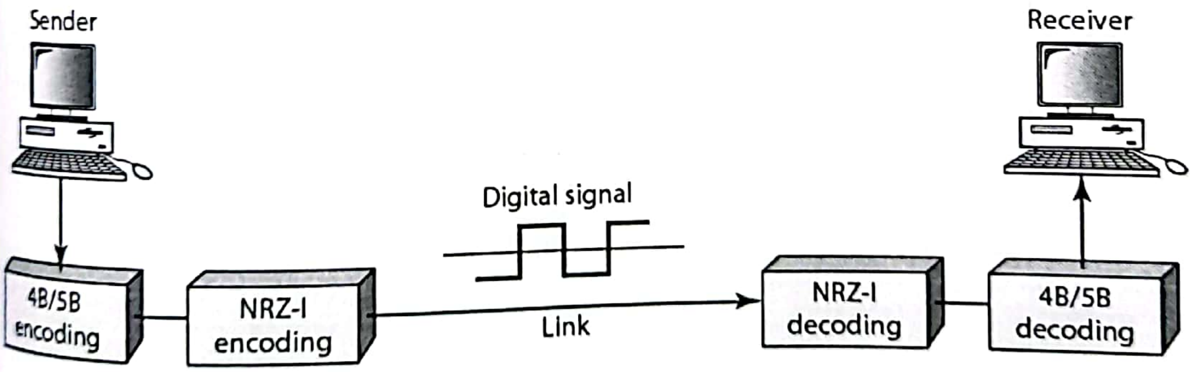
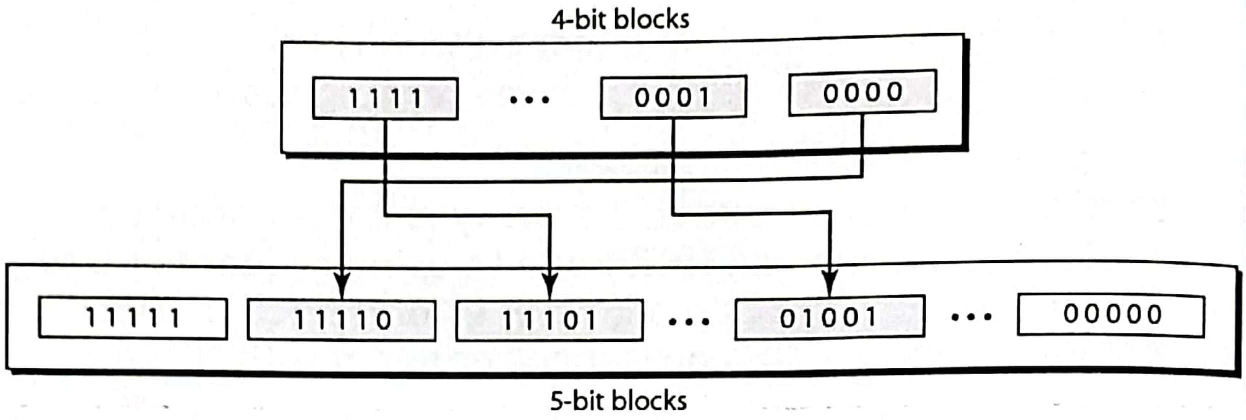


Figure 4.16 *Substitution in 4B/5B block coding*



تاني قاعدة ممنوع يجي 3 zero's ورا بعض من ال
least sig bit (يعني ممنوع أكثر من 2 zero's)

4.43

أول قاعدة ممنوع يجي 2 leading zero's ورا بعض
(يعني بال most significant bit ما بصير يجي 00)

لم يتم اختيارها بشكل عشوائي.

Table 4.2 *4B/5B mapping codes*

Data Sequence	*Encoded Sequence*	*Control Sequence*	*Encoded Sequence*
0000	11110	Q (Quiet) ← ارسالهم مش	00000
0001	01001	I (Idle) متكرر وغالبا ما برسلوا ورا بعض.	11111
0010	10100	H (Halt)	00100
0011	10101	J (Start delimiter)	11000
0100	01010	K (Start delimiter)	10001
0101	01011	T (End delimiter)	01101
0110	01110	S (Set)	11001
0111	01111	R (Reset)	00111
1000	10010		
1001	10011		
1010	10110		
1011	10111		
1100	11010		
1101	11011		
1110	11100		
1111	11101		

4.44

Example 4.5

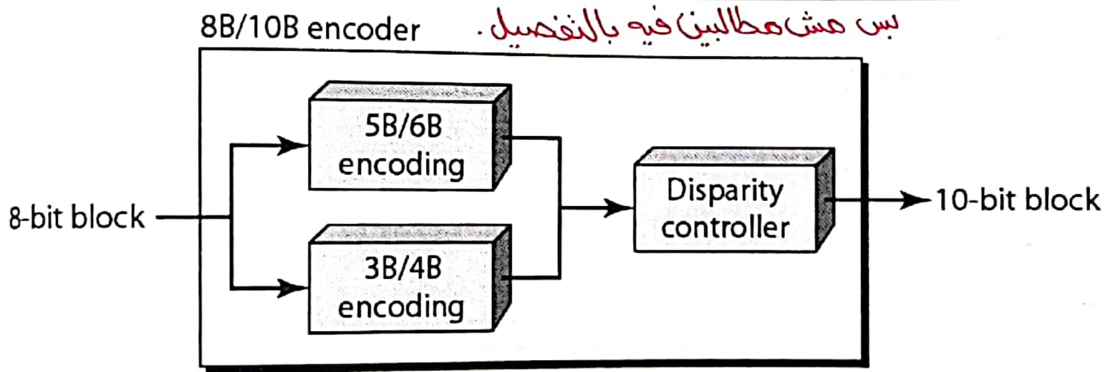
We need to send data at a 1-Mbps rate. What is the minimum required bandwidth, using a combination of 4B/5B and NRZ-I or Manchester coding?

Solution

First 4B/5B block coding increases the bit rate to 1.25 Mbps. The minimum bandwidth using NRZ-I is N/2 or 625 kHz. The Manchester scheme needs a minimum bandwidth of 1 MHz. The first choice needs a lower bandwidth, but has a DC component problem; the second choice needs a higher bandwidth, but does not have a DC component problem.

4.45

Figure 4.17 *8B/10B block encoding* → ممكن نفصل Combination
من ال blocks
بس مش مطالبين فيه بالتفصيل.



- It is a combination of 5B/6B and 3B/4B encoding for simpler mapping
- The disparity controller is used prevent long consecutive 0's or 1's
- If the bits in the current block creates a disparity that contributes to the previous disparity, the bits in the current block are complemented

4.46 - The coding has $2^{10} - 2^{8} = 768$ redundant code that can be used for disparity checking and error detection

4-2 ANALOG-TO-DIGITAL CONVERSION

*We have seen in Chapter 3 that a digital signal is superior to an analog signal. The tendency today is to change an analog signal to digital data. In this section we describe two techniques, **pulse code** ① **modulation** and **delta modulation**.* ②

↳ more complex

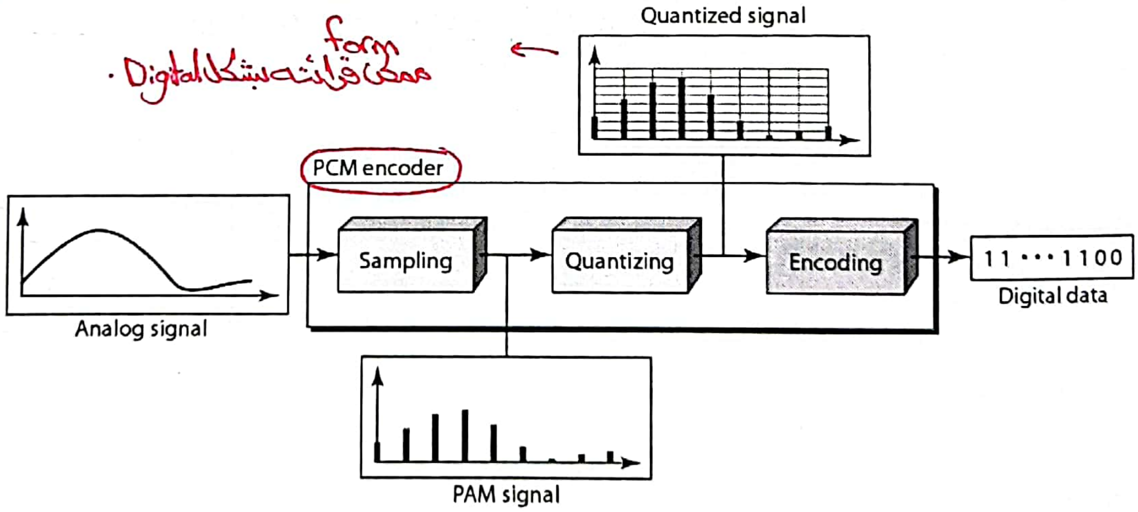
كلشي حابين نحوله لـ Digital
متل الصوت وهيك .

Topics discussed in this section:

Pulse Code Modulation (PCM)
Delta Modulation (DM)

Figure 4.21 ***Components of PCM encoder***

ممكن قرائته بشكل Digital form .



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

Sampling: Definition and Background

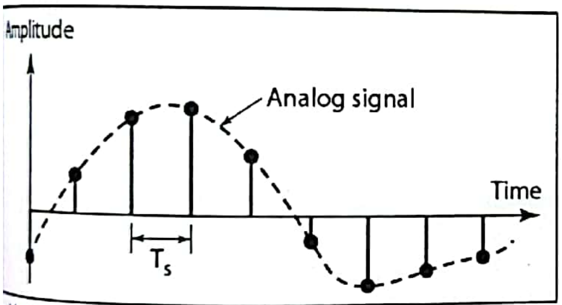
* Sampling is converting analog signals into digital by taking samples at certain uniform intervals called sampling interval (or sampling period), T_s
* The inverse of the sampling interval is called sampling rate (or sampling frequency), $f_s = 1/T_s$
* Sampling is also called Pulse Amplitude Modulation (PAM)
* The idea started by telephone carriers to provide long distance services

  البداية للفكرة

  * The analog voice signal loses power on long distance cables and therefore require amplifiers
  * Amplifiers distort the signal due to their own frequency spectrum and phase changes and they also add noise
  * Since digital signals are more immune to noise and distortion, digitization is used

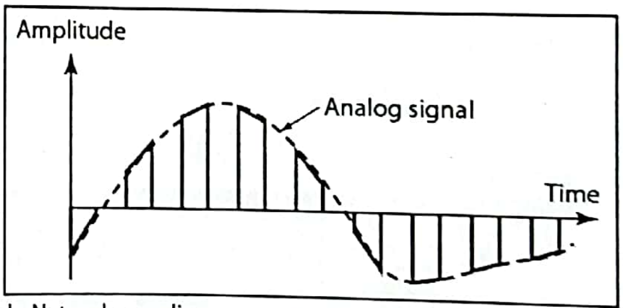
* اذا ماقدرت تعرف أعلى frequency موجودة عندك بال Signal ما بتعرف شو ال Sampling Rate.

Figure 4.22 ***Three different sampling methods for PCM***



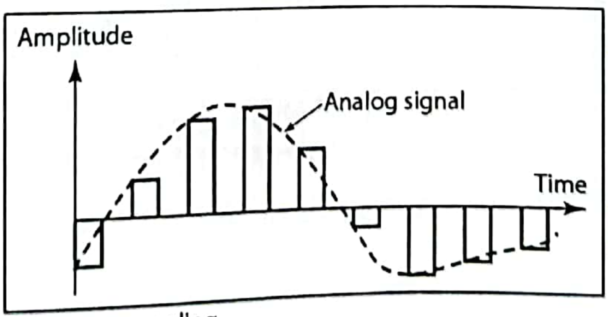
a. Ideal sampling

ما بنحصله بأرض الواقع، بس بنتعامل معه رياضياً.



b. Natural sampling

كأني بفتح بوابة لمدة معينة بعدين بسكرها وهكذا لتطلع ال Signal T_s



c. Flat-top sampling

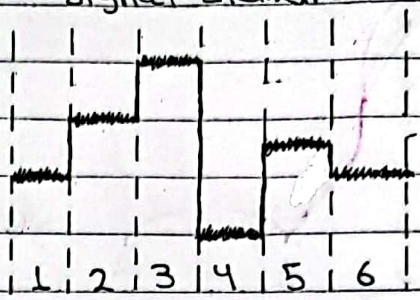
أو تفتح ال Switch وتشحن ال Capacitor ل Value معينة وتسكر ال Switch وهكذا. بتظل ثابتة

10/11/2022

*Summary for block coding Slide 4.41 :

m → n

Data Elements. Signal Elements.



$\underbrace{01101}_{m=5}$

5 levels ←

⟹ 2^5

↳ B = 0,1

*B: number of available data elements

← n = 6 →

*m: number of Data elements in a data pattern.

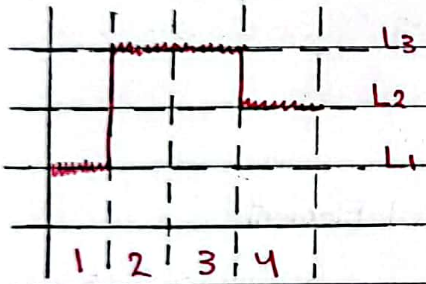
*L: number of available Signal levels.

*n: number of Signal elements in a pattern.

*B^m : number of all combinations of data pattern.

*L^n : number of all combinations of Signal pattern.

*Note: how we choose L^n ?



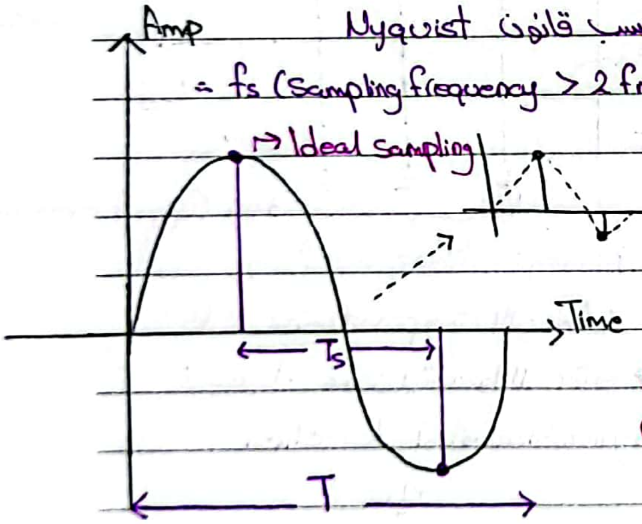
⟹ $\overline{L_3}\,\overline{L_2}\,L_1$ / $\overline{L_3}\,L_2 L_1$ / .....

⟹ L^n ⟹ 3^4

Slide 4.49:

Nyquist حسب قانون

= fs (Sampling frequency > 2 fmax) ⟹ $f = \frac{1}{T} = f_{max}$ / $f_s = \frac{1}{T_s}$



*افرض لو كانت الـSignal عبارة عن Composit

يعني فيها عدة frequencies وكانت الـBandwidth

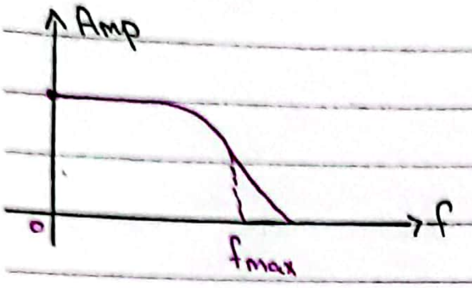
بتساوي B = 2 KHz شو الـSampling frequency ؟

① $f_s > 4$ KHz (low pass)

② f_s = ? (can't defined) (band pass)

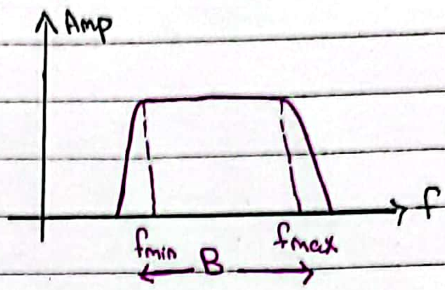
Five Apple

① low Pass :



$B = f_{max} - 0 = f_{max}$

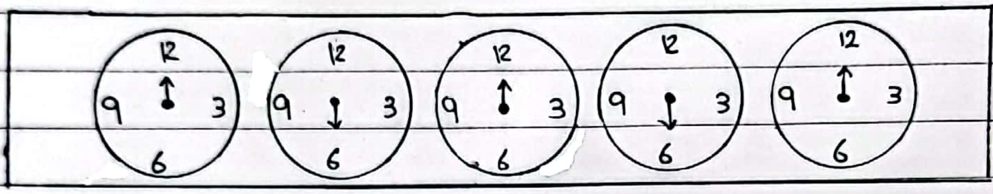
② Band Pass :



لـ ما بنعرف لإنصاف معنا f_{min}

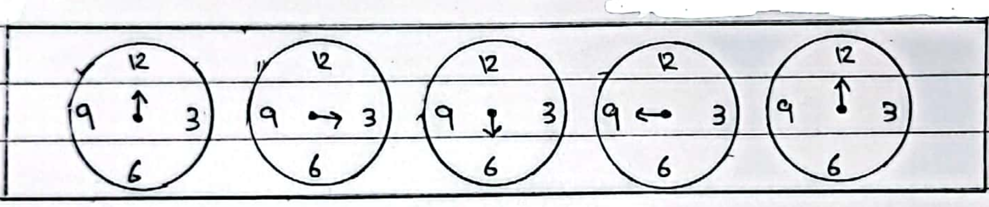
* مثال في الكتاب بنفس فكرة مثال 4.6 بـ Slide 54 :

" Sampling of a clock with only one hand "



بعمل Sampling كل دقيقة.

a. Sampling at Nyquist rate: $T_t = \frac{1}{2} T$ → مابقدر أحدد كيف بمشي العقرب

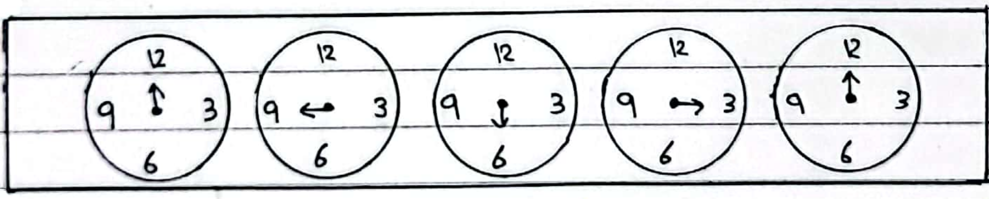


$\therefore f_s = 2 f_{max}$

$\rightarrow \frac{1}{T_s} = \frac{2}{T_{max}}$

$\rightarrow T_s = \frac{T_{max}}{2}$

b. oversampling (above Nyquist rate): $T_t = \frac{1}{4} T$ → قدرت أحدد الاتجاه



c. Undersampling (below Nyquist rate): $T_1 = \frac{3}{4} T$ → قدرت أحدد الاتجاه

Slide 4.59 :

$|error| \leq \frac{\Delta}{2} \rightarrow -\frac{\Delta}{2} \leq error \leq \frac{\Delta}{2}$

* كلما صغرنا Δ كل ما قل ال error.

Sampling Rate: Nyquist Theorem

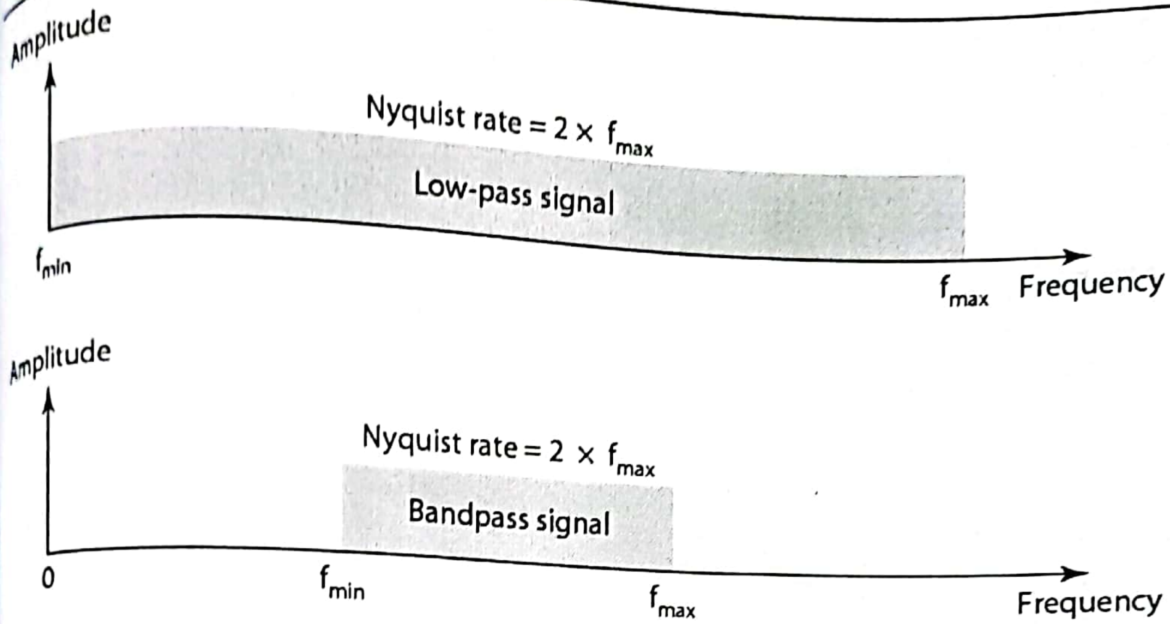
- Question: How many samples are needed to digitally reproduce the analog signal accurately?
  - Ideally, infinite number of samples

- Nyquist Theorem: the sampling rate should be at least twice the highest frequency component in the original analog signal

Note

According to the Nyquist theorem, the sampling rate must be at least 2 times the highest frequency contained in the signal.

Figure 4.23 ***Nyquist sampling rate for low-pass and bandpass signals***



4.53

Example 4.6

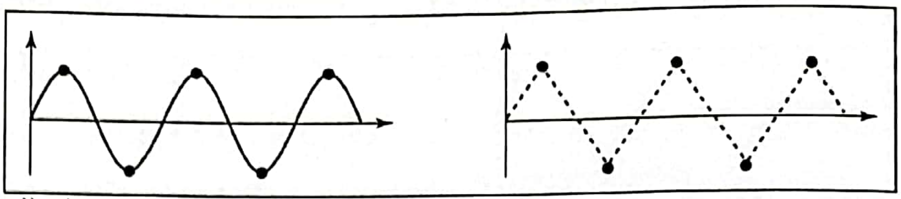
For an intuitive example of the Nyquist theorem, let us sample a simple sine wave at three sampling rates: $f_s = 4f$ (2 times the Nyquist rate), $f_s=2f$ (Nyquist rate), and $f_s = 4f/3$ (a little more than one-half the Nyquist rate). Figure 4.24 shows the sampling and the subsequent recovery of the signal.

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

4.54

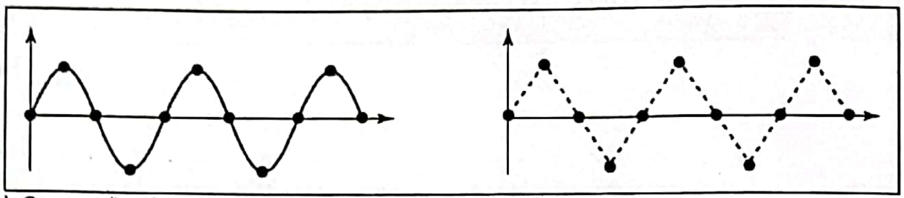
Figure 4.24 ***Recovery of a sampled sine wave for different sampling rates***

← بتقدر توصل فيه للـ Original Signal.



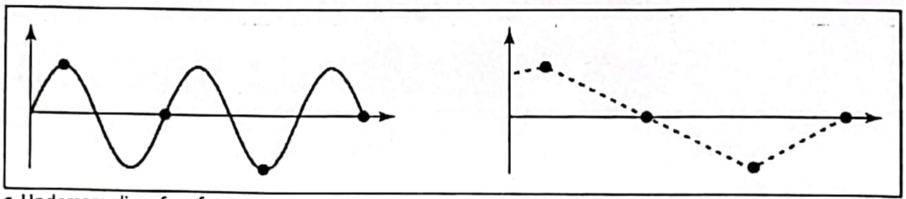
a. Nyquist rate sampling: $f_s = 2f$

← ممكن ما يكون مفيد لإنه باخذ زيادة Samples ما أثرت عالشكل.



b. Oversampling: $f_s = 4f$

← ما بتقدر توصل للـ Original Signal.



c. Undersampling: $f_s = f$

4.55

Example 4.9

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

* Note: بالأفلام بكون الـ Frame اللي بتاخذ بالتصوير 24 Frame بالثانية فإذا عجل السيارة بمشي أكثر من 12 لفة بالثانية حنشوفه كإنه برجع لورا.

Example 4.10

frequency أكثر من

ملش من الصفر

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

Solution

The bandwidth of a low-pass signal is between 0 and f, where f is the maximum frequency in the signal. Therefore, we can sample this signal at 2 times the highest frequency (200 kHz). The sampling rate is therefore 400,000 samples per second.

Example 4.11

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

Solution

We cannot find the minimum sampling rate in this case because we do not know where the bandwidth starts or ends. We do not know the maximum frequency in the signal.

Quantization

- Quantization: assigning values in a specific range of sampled instances
- Each value is translated into a binary equivalent number (i.e.; Binary Encoding)
- The binary digits are converted into digital signal using line coding
- Steps of quantization:
  - 1. Assume that the analog signal ranges between V_{min} and V_{max}
  - 2. Divide the range into L zones each of height Δ (delta)

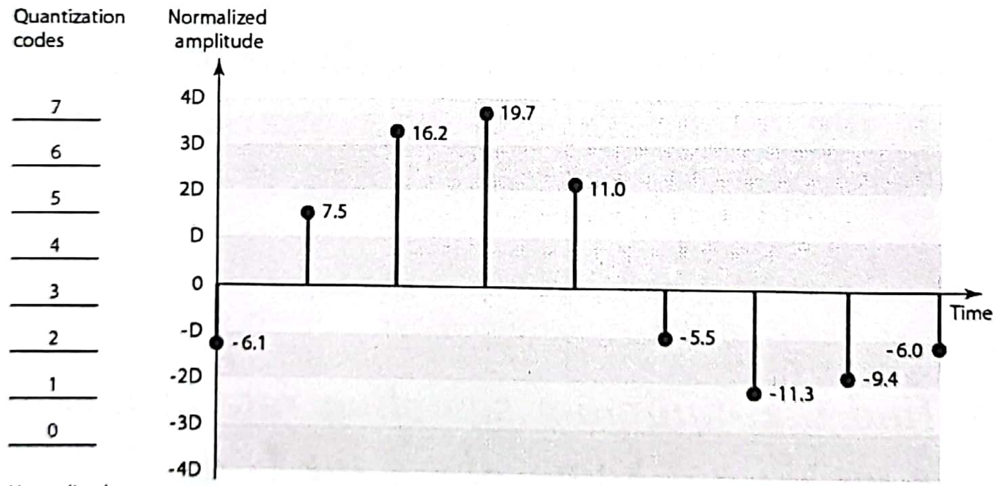
*قسمت ال range بشكل متساوي (uniform) ممكن أقسمه بشكل غير متساوي أيضاً.

$$\Delta = \frac{V_{max} - V_{min}}{L}$$

  - 3. Assign quantized values of 0 to L-1 to the midpoint of the zone
  - 4. Approximate the sample amplitude to the quantized value

لـ أي عملية Value تقع بين الفترة الصغيرة بقربها لل midpoint تاع هاي الفترة وهنا بعمل error.

Figure 4.26 ***Quantization and encoding of a sampled signal***



- V_{min} = -20 v
- V_{max} = +20 v
- L = 8
- Δ = 5 v

$\frac{V_{max} - V_{min}}{L}$

مقسوم Δ 2 ؟

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

Quantization Levels and Error

- The number of levels, L, depends on:
  - The amplitude range of the analog signal
  - The accuracy needed in recovering the signal
- Choosing low values of L may increase the quantization error if the signal changes a lot
- The quantization error for each sample is less than $\Delta/2$ ($-\Delta/2 \leq \text{error} \leq \Delta/2$)
- The contribution of the quantization error to the SNR_{dB} of the signal depends on L or n_b (the number of bits per sample) لو كان كبير بكون

عدد الـ bits كبير ← $n_b = \log_2 L$

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

كل ما تزيد عدد الـ bits الـ SNR بتتحسن،
الممانعة للـ noise بتكون أكبر لأنه.

4.61

Example 4.12

What is the SNR_{dB} in the example of Figure 4.26?

* لأحسب الـ SNR لازم أحسب الـ # of bits ، ولأحسب
الـ # of bits لازم احسب الـ # of levels .

Solution

We can use the formula to find the quantization. We have eight levels and 3 bits per sample, so

$$SNR_{dB} = 6.02(3) + 1.76 = 19.82 \text{ dB}$$

Increasing the number of levels increases the SNR.

بس بنزيد الـ bandwidth.

4.62

Example 4.13

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

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

* *Telephone companies usually assign 7 or 8 bits per sample.* *

* Frame Rate : how many frames per second.

4.63

→ Pulse Code Modulation

تعتبر عملية معقدة

PCM Bandwidth

بنبدأ من Zero ملم low-pass.

- Consider a low-pass analog signal
- Bit Rate = Sampling Rate x bits per sample

  $= f_s \times n_b$ ↳ how many Samples per second

  $= 2 \times B_{analog} \times \log_2 L$ (Nyquist Data Rate)

- $B_{min} = c \times N \times 1/r = c \times f_s \times n_b \times 1/r$

  $= c \times 2 \times B_{analog} \times n_b \times 1/r$
- When r=1 (for NRZ or Bipolar) and c=1/2,

$$B_{min} = n_b \times B_{analog}$$

4.64

Example 4.14

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

Solution

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

Sampling Rate = 4000 x 2 = 8000 samples per second
Bit Rate = 8000 x 8 = 64,000 bps = 64 kbps

4.65

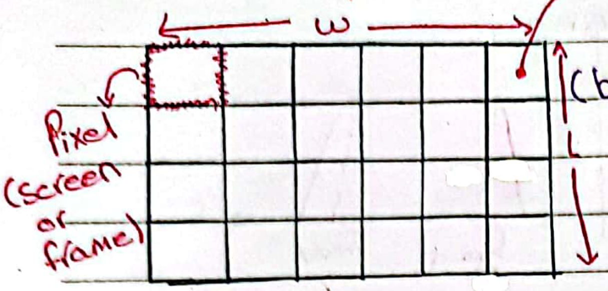
Example 4.15

We have a low-pass analog signal of 4 kHz. If we send the analog signal, we need a channel with a minimum bandwidth of 4 kHz. If we digitize the signal and send 8 bits per sample, we need a channel with a minimum bandwidth of 8 × 4 kHz = 32 kHz.

4.66

15/11/2022

* Solve the Quiz :



* احنا أخذنا الحالة الـ general بشكل عام
فهيك صارت حالة الـ black and white (b = 1)
حالة خاصة من الحالة العامة.

* Screen = Frame

(حل الطلب) : $*\, w * L * b$, bits / screen.

$*\, w * L$, Pixels / Screen.

$*\, fr$ of frames per second (Screens per Second)

A- Bandwidth ? (Data Rate / 2)

B- Data Rate ?

C- Lower Bound ? (for SNR) → in worst case = 3.

D- if $SNR_{dB} = 0$, Can we transmit data ? we can transmit but we can't recieve the right data.

E- BDP (Bandwidth Delay Product)

B- using, N (Data Rate) = $2B\log_2 L$, $\log_2 L = b$

→ $N = 2Bb$, but $N = \frac{(w*L*b)\text{ bits}}{1\text{ frame}} \times \frac{fr\text{ frame}}{1\text{ Second}}$

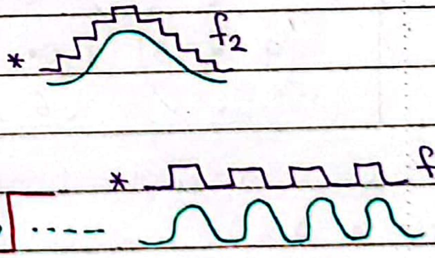
→ $N = w*L*b*fr$ (bps)

→ $w*L*\cancel{b}*fr = 2B\cancel{b}$

A- $B = \frac{w*L*fr}{2}$

or

worst case Senario : b = 1 / w B w B w B ....



$f_w = \frac{1}{T_w}$ ← T_w

→ $B = f_w - 0 = f_w$ → $N = \frac{2\text{ bits}}{1\text{ cycle}} \times \frac{1\text{ cycle}}{T_w}$ → $N = 2f_w = 2B$ → $N = 2B$

C - Maximum link Capacity (bps)

$$C = B \log_2 (1+SNR)$$

→ $2\cancel{B}b = \cancel{B}\log_2(1+SNR) \rightarrow 2b = \log_2(1+SNR) \rightarrow 2^{2b} = 1+SNR$

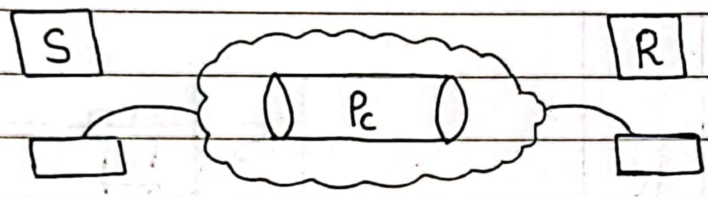
→ $\boxed{SNR = 2^{2b}-1}$ ∴ if $b=1 \rightarrow SNR=3$

D- $SNR_{dB} = 0 \rightarrow SNR = 1$

∴ $C = B\log_2(1+SNR) = B\log_2(1+1) \rightarrow 2\cancel{B} = \cancel{B}\log_2(1+1)$

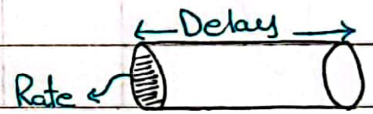
→ $2 = \log_2(1+1) \rightarrow 2 \neq 1$ → we can transmitt but can't recieve.

E-



Pc: Pipe or link capacity in Packets (Pkts)

P_c = Rate x Delay (BDP)



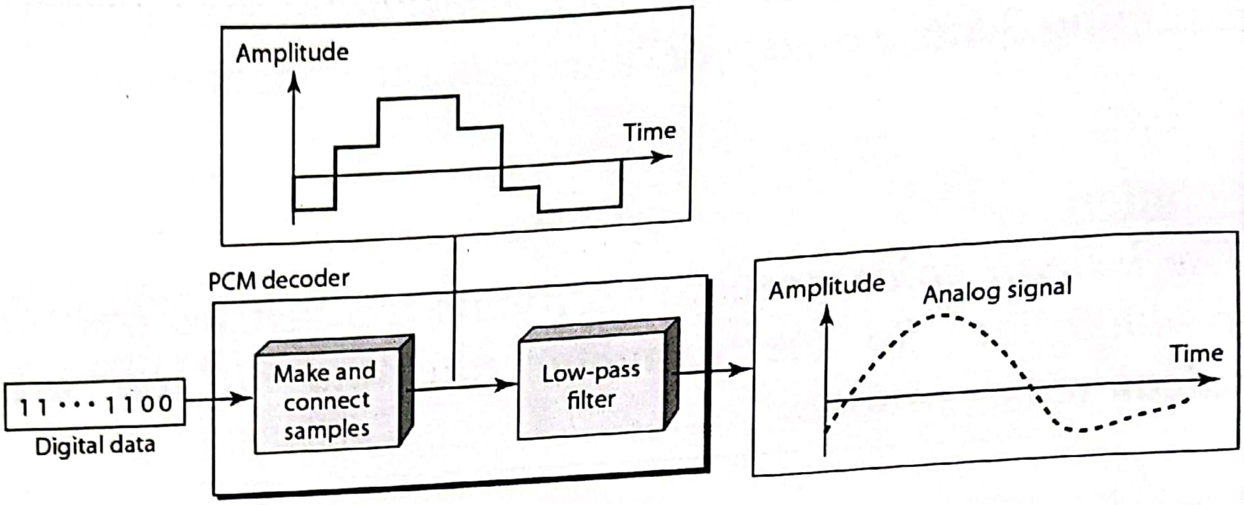
∴ If each Screen/frame is transmitted using "n" Packets.

→ $f_r \frac{\text{Screens}}{s} \times \frac{n \text{ Packets}}{1 \text{ screen}} = n f_r$ Pkts/s → D = ?

$$P_c = n f_r \times D \rightarrow D = \frac{P_c}{n f_r}$$

17/11/2022

Figure 4.27 *Components of a PCM decoder*



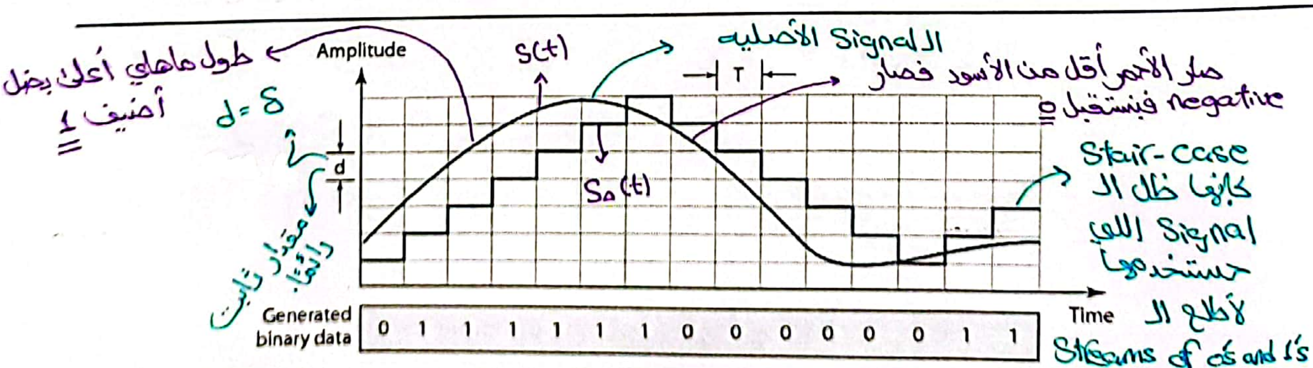
4.67

هي الطريقة البديلة والأسهل بدل ال PCM

Delta Modulation (DM)

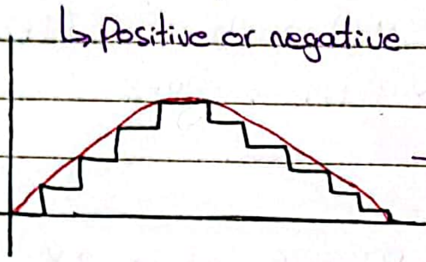
- PCM is a relatively complex A-to-D technique
- DM is a much simpler technique than PCM:
  - Finds the delta change of the current sample compared to the previous sample
  - If the current sample is larger, it sends a 1. Otherwise, it sends a 0

Figure 4.28 *The process of delta modulation*



4.68

③ Delay the Stair-case signal by T (Delay unit بالـ يكون عندي)
④ $S_\Delta(t) = S_\Delta(t-T) + \delta$ (update of Δ)
↳ Positive or negative

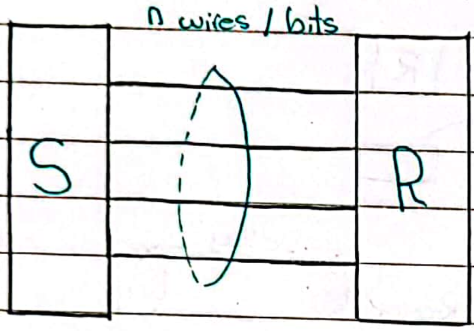


→ Slide 4.70

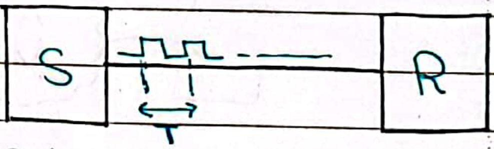
* Slide 4.74 :

Parallel :

n wires / bits



Serial :



Rate $= \frac{1}{T}$ bit → 1 bit بيبعت per T

clock cycle = T

Rate $= \frac{n}{T}$ → every T بيبعت n bits في

* nT_s seconds to transmit n bits.

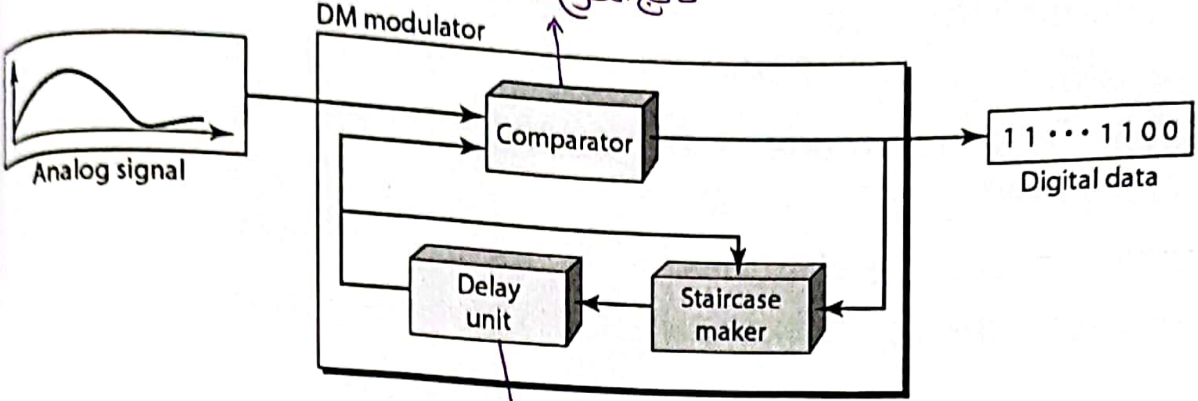
* T_p Seconds to transmit n bits.

← لو الـ T كانت متساوية يكون الـ Parallel أسرع ، لكن مرات بتكون الـ Serial أسرع
فما بنقدر نحدد بطريقة مطلقة مين الأفضل.

* هلأ في الـ Personal Computers الـ Connections بين الـ harddisk والـ cpu
وهيك Serial مش Parallel رغم إنها قبل كانت Parallel بس ثاتوا إنه الـ Serial
فيهم أفضل.

Figure 4.29 *Delta modulation components*

بلزمك معلومتين الأولى
ناتج الطرح (Δ)

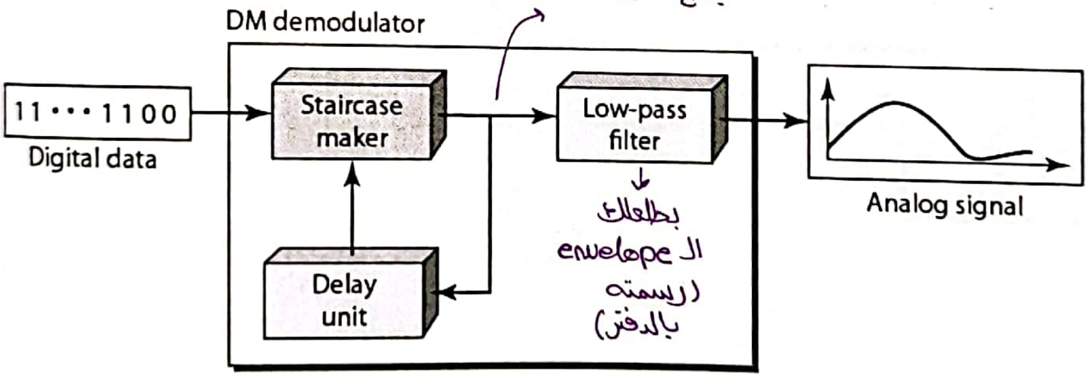


والثانية كشان يعرف T
شو بتساوي مشان ال Delay

Figure 4.30 *Delta demodulation components*

العملية العكسية:

بطلع ال Stair-case



بطلعك ال envelope
(رسمته بالدفتر)

4-3 TRANSMISSION MODES

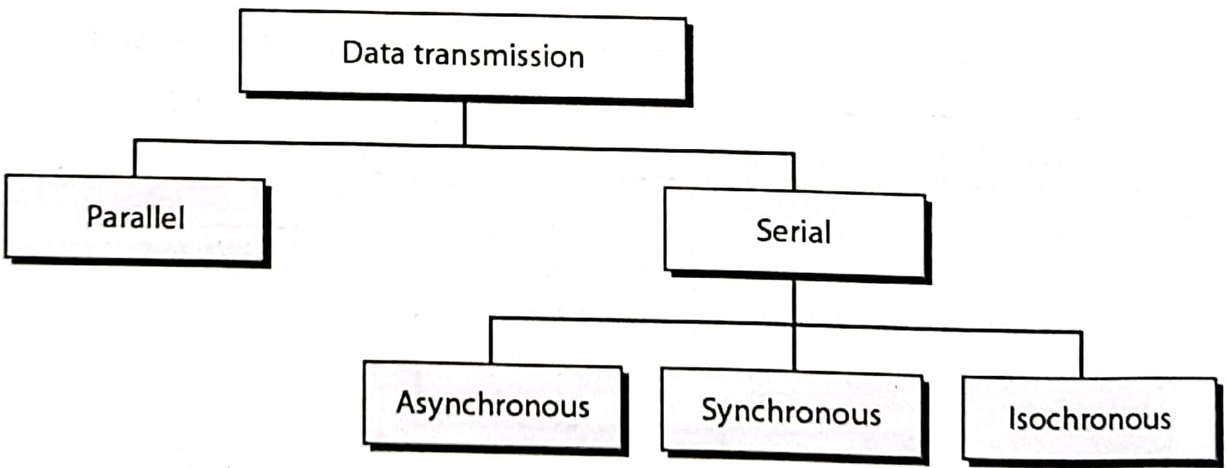
The transmission of binary data across a link can be accomplished in either parallel or serial mode. In parallel mode, multiple bits are sent with each clock tick. In serial mode, 1 bit is sent with each clock tick. While there is only one way to send parallel data, there are three subclasses of serial transmission: asynchronous, synchronous, and isochronous.

نوع ثالث وجديد من ال transmission

Topics discussed in this section:

Parallel Transmission
Serial Transmission

Figure 4.31 ***Data transmission and modes***

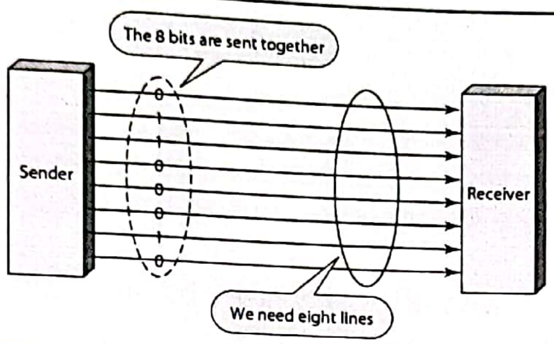


Delay between frames not bits.

Parallel Transmission

- Principle: use *n* wires to send *n* bits simultaneously → مع بعض
- Advantage: speed (*n* times faster than serial transmission)
- Disadvantage: cost and complexity due to the extra wiring لوجود الأسلاك
- Usually limited to short distances → ما بنفعه لمسافات طويلة عشان المشاكل
- Devices: older Centronics printers, internal data & address buses
- Adapters: PIA (Par. Interface Adapter), PPI (Par. Peripheral. Interface)

Figure 4.32 ***Parallel transmission***



3

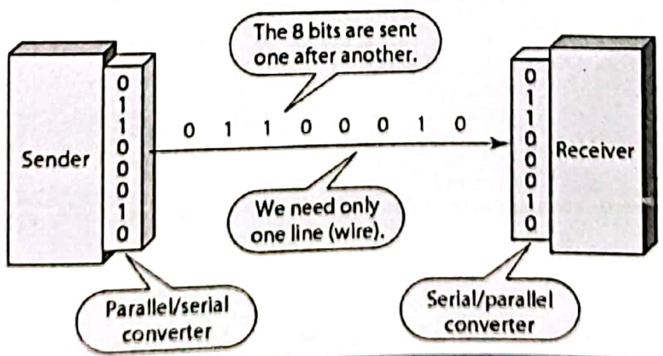
* مين الأسرع ال Parallel ولا ال Serial ؟ الجواب بالدفتر

like LCD

Serial Transmission

- Principle: use 1 wire to send 1 bit at a time
- Advantage: cost and simplicity (almost a factor of *n* less than parallel)
- Requires serial-to-parallel and parallel-to-serial conversion
- Devices: Peripheral devices (e.g. mouse & keyboard), modems, etc.
- Adapters: ACIA (Asynchronous Comm. Interface Adapter), UART (Universal Asynchronous Receiver Transmitter)

Figure 4.33 ***Serial transmission***

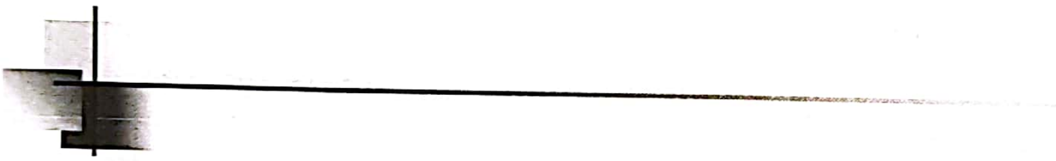


4

ع الي بدنا نعرفه في Start bit و Stop bit ومافي Clock ..

Asynchronous Serial Transmission

- The information is received and translated by agreed-upon patterns
- Usually, patterns are based on grouping the bits into bytes
- The sender handles each group independently
- Each group is sent whenever it is ready without regard to a timer
- To alert the receiver to the arrival of a new group, a "start bit" (usually a 0 bit) is added to the beginning of the group
- To let the receiver know that the byte has finished, 1 or more "stop bits" are appended to the end of the group
- Each group may be followed by a gap of random duration
- The gap can be an idle channel or a stream of stop bits
- The start and stop bits allow the receiver to synchronize with the data stream within the group
- "Asynchronous" means that the sender and receiver do not have to be synchronized at the group level, but at the bit level within the group
- The receiver counts n bits after the start bit and looks for the stop bit



Note

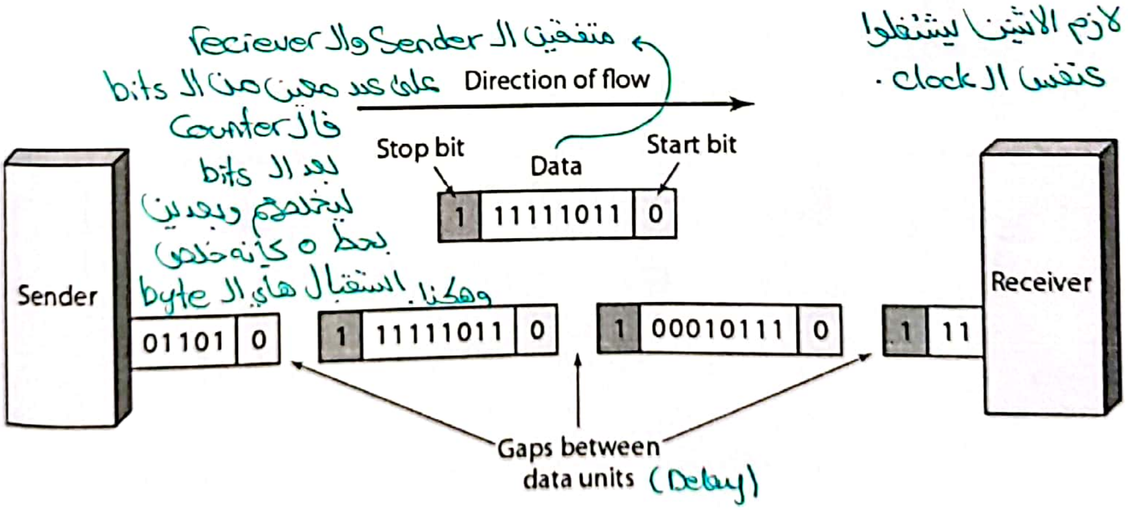
In asynchronous transmission, we send 1 start bit (0) at the beginning and 1 or more stop bits (1s) at the end of each byte. There may be a gap between each byte.

Note

Asynchronous here means "asynchronous at the byte level," but the bits are still synchronized; their durations are the same.

Figure 4.34 *Asynchronous transmission* *

* لازم الاثنين يشتغلوا بنفس الـ clock.



Synchronous Serial Transmission

مجموعة من ال bits

- The bit stream is combined into longer "frames"
- Each frame may contain multiple bytes without gaps
- Mainly, the data is sent as a continuous stream of bits
- The receiver decides how to group them (e.g.; into bytes, characters, numbers, etc.) for decoding purposes

بحسب ال Protocol

- If the sender sends the data in bursts, the gap must be filled with a special sequence of 0's & 1's (i.e.; idle)

لو بده يبعث bit فاضي

- Without start and stop bits, the receiver can't adjust its bit-level synchronization → خلال ال clock الشابك عالطرفين
- Therefore, strict timing between the sender and the receiver is required in order to receive the bits correctly
- The advantage of synchronous transmission is speed as there is no overhead of synchronization bits

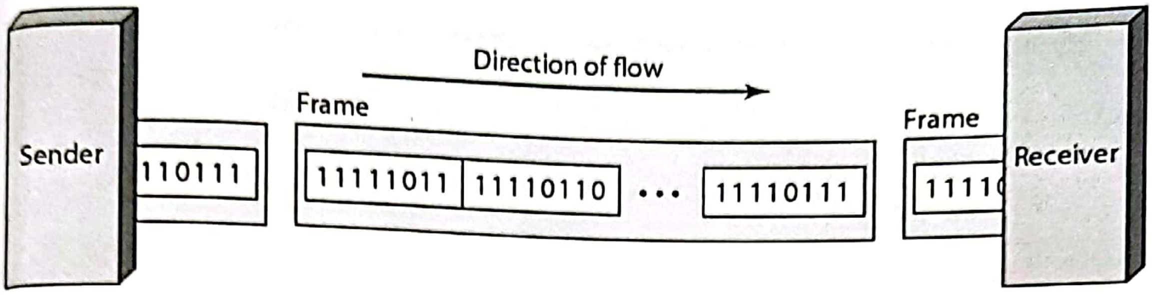
← أسرع، مافي زيادة لل bits ال Start وال Stop

4.79

Note

In synchronous transmission, we send bits one after another without start or stop bits or gaps. It is the responsibility of the receiver to group the bits.

Figure 4.35 *Synchronous transmission*



31

Isochronous Serial Transmission

audio and Video

- Used for real-time applications, where:
  - The synchronization between characters or bytes is not enough but rather the synchronization of the entire stream

    يفضل يا يكون ال Delay قد بعض يا مايكون في Delay نهائيا.

  - The delay between frames must be equal or none

    ↳ not bits

    التغير بال Delay بين ال Frames اسمه Jitter.

  - The data is received at a fixed rate
- Examples: real-time audio and video streaming

32

* كل اللي أخذناه بـ Ch4 كان Digital transmission لكن بـ Ch5
حناخد الـ Analog transmission اللي بنحتاجه عشان الـ broadband.

Chapter 5

Analog Transmission

5-1 DIGITAL-TO-ANALOG CONVERSION

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

بسبب إنه ال Channel تبعتنا محدودة الbandwidth فما بنقدر نبعت ال Square wave لأنه فيها عدد هائل من ال Frequency.

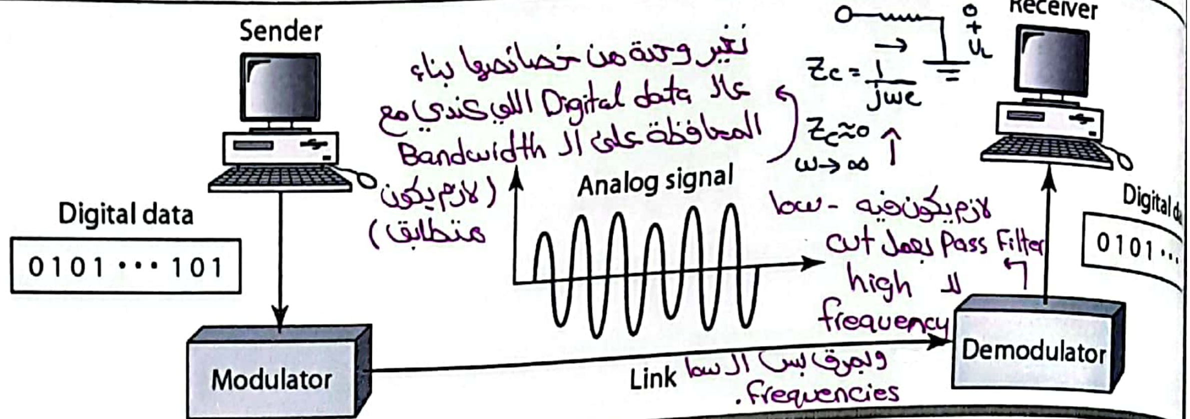
Topics discussed in this section:

Aspects of Digital-to-Analog Conversion
Amplitude Shift Keying
Frequency Shift Keying
Phase Shift Keying
Quadrature Amplitude Modulation

Figure 5.1 *Digital-to-analog conversion*

LPF

f_c



$Z_c = \frac{1}{j\omega C}$

$Z_c \approx 0$, $\omega \to \infty$

v_L

تغير وحدة من خصائصها بناء على الـ Digital data اللي عندي مع المحافظة على الـ Bandwidth

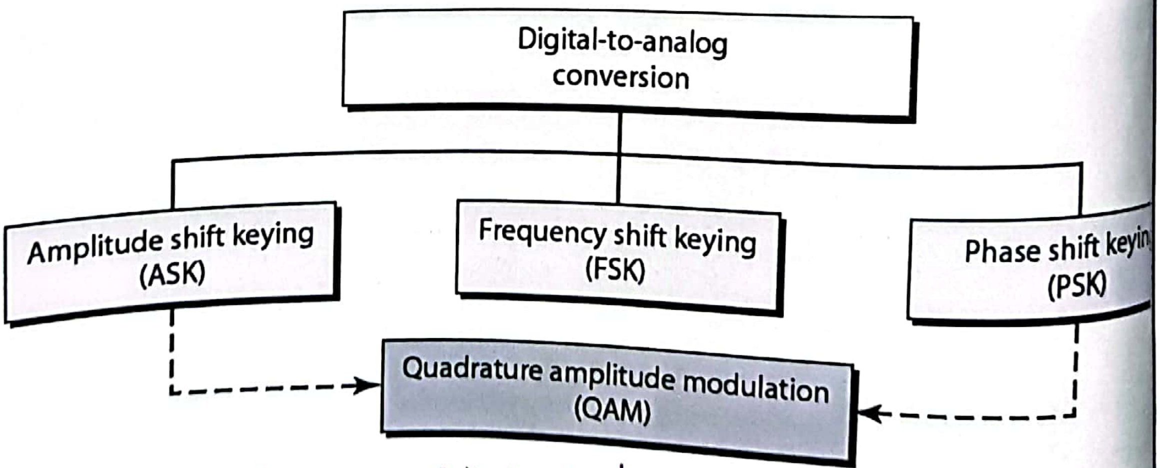
(لازم يكون متطابق)

لازم يكون فيه low-cut بعمل Pass Filter للـ high frequency

ويمرق بس الـ low frequencies.

- Digital-to-analog modulation (or shift keying): changing one of the characteristics of the analog signal based on the information of the digital signal (carrying digital information onto analog signals)
- Remember: changing any of the characteristics of the simple signal (amplitude, frequency, or phase) would change the nature of the signal to become a composite signal — 1 frequency (not composite)
- بس تتخل within the bandwidth of the link.
- The digital information can be carried as predefined changes to one or more of the characteristics (e.g., no-change = 0 and some-change = 1)

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



له بيزيد رفع الـ "Data Rate"

Aspects of Digital-to-Analog Conversion

- Data Element vs. Signal Element
- Data Rate vs. Signal Rate
  - $S = N/r$
  - $r = \log_2 L$, where L is the number of signal elements
- Bandwidth:
  - The required bandwidth for analog transmission of digital data is proportional to the signal rate
- Carrier Signal:
  - The digital data changes the carrier signal by modifying one of its characteristics
  - This is called modulation (or Shift Keying)
  - The receiver is tuned to the carrier signal's frequency

Note

Bit rate is the number of bits per second. Baud rate is the number of signal elements per second.

In the analog transmission of digital data, the baud rate is less than or equal to the bit rate.

Example 5.1

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 ca find the value of N from

$$S = N \times \frac{1}{r} \quad \text{or} \quad N = S \times r = 1000 \times 4 = 4000 \text{ bps}$$

Example 5.2

An analog signal has a bit rate of 8000 bps and a baud rate of 1000 baud. How many data elements are carried by each signal element? How many signal elements do we need?

Solution

In this example, S = 1000, N = 8000, and r and L unknown. We find first the value of r and then the va of L.

$$S = N \times \frac{1}{r} \longrightarrow r = \frac{N}{S} = \frac{8000}{1000} = 8 \text{ bits/baud}$$

$$r = \log_2 L \longrightarrow L = 2^r = 2^8 = 256$$

نغير ال Amplitude لل Carrier.

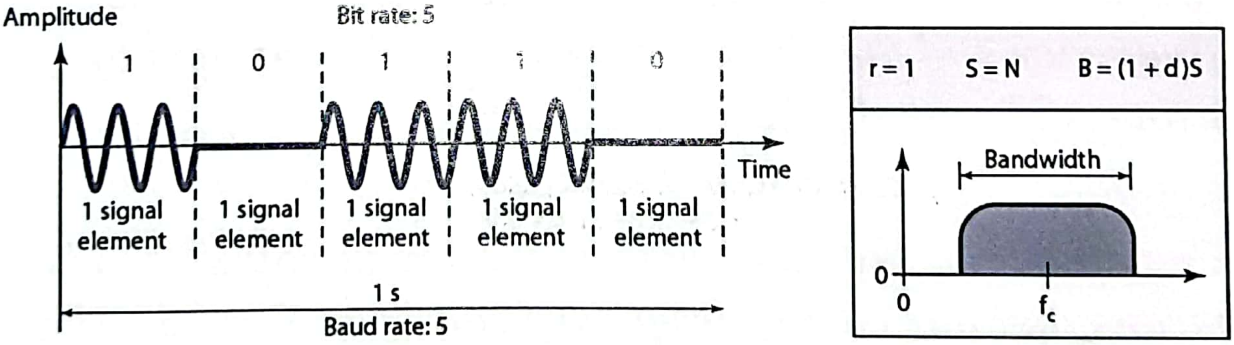
Amplitude Shift Keying (ASK) = OOK

- ASK: varying the peak amplitude of the carrier signal to represent binary 1 or 0
- Other signal characteristics remain constant as the amplitude changes
- During each bit duration, the peak amplitude remains constant
- Transmission medium noise is usually additive (i.e.; affects the amplitude), therefore, ASK is very susceptible to noise interference
- Binary ASK (BASK) or On/Off Keying (OOK) modulation technique:
  - * One of the binary bits is represented by no voltage
  - * Advantage: requires less transmission energy compared to two-level techniques
- BASK spectrum is most significantly between [f_c - S/2, f_c + S/2], where f_c is frequency of the carrier signal and S is the baud rate → ال Parameter العلاقة بال Modulation والـ Sampling.
- The ASK bandwidth is B_{BASK} = (1+d) S , where d is a factor related to the modulation process, of which the value is between 0 and 1 تعوض مكان ال C اللي كان بال S
- The min. bandwidth required to transmit an ASK is equal to the baud rate (يعني هون صرنا ناخذ ال Cases بعين الاعتبار)
- The baud rate is the same as the bit rate ➔ N = S

* d = 0 → Min Bandwidth → Min B = Signal Rate.
* d = 1 → Max Bandwidth → Max B = 2 * Signal Rate.

* Signal rate = Data rate → السبب إنه r=1 حسب المعادلة.

Figure 5.3 *Binary amplitude shift keying*



* ASK-T1 : بتوضح كيف بتم ضرب ال 2 Signal من خلال ال Resistor.
* Bandwidth → Symmetric about carrier frequency
↳ around

20/11/2022

Chapter 5 :

* Analog Signal → $S(t) = A \sin(2\pi f_c + \phi)$

① Amplitude ; ② frequency ; ③ phase

↳ بيغير خاصية أو أكثر بال Analog Signal بحيث إنك تنقل Digital information عليها مع المحافظة على ال bandwidth .

* Analog Transmission : (5.5) (5.6)

- Data Element : Smallest piece of information to be exchange (bit = في حالتنا)
- Signal Element : Smallest unit of a Signal that is constant
  - ↳ * In Analog Signal, the nature of the Signal element is different. we usually refer to the type of Signal element.
- Data Rate Vs Signal Rate :

$S = N * \frac{1}{r}$ baud (الوحدة) / $r = \frac{\#\text{ of data elements}}{\#\text{ of signal element}}$

↳ * ماذا في حالة ال C في ال analog ما فيها

لأنه حاليا بنتعامل مع 1 signal digital Signal (لأنه ال C بنفيد حال ال case الي بنبعت فيها 0's و 1's) .

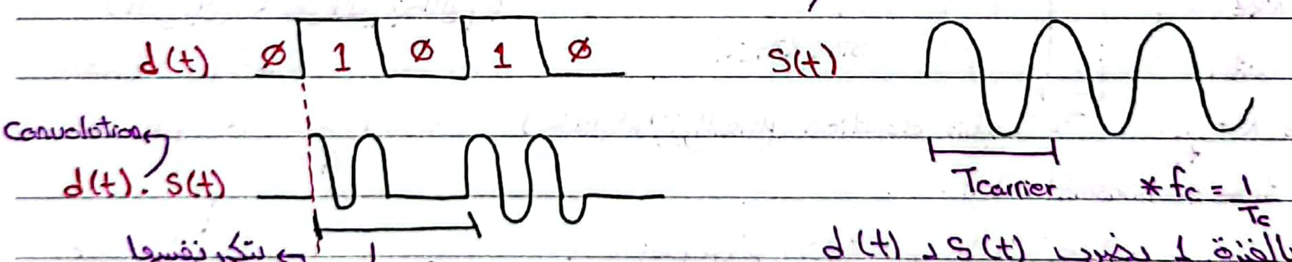
* ↑ Signal element (not data element) → ↑ Bandwidth Requirement → ↑ # of changes in Signal (rising and falling Edges)

- Baud Rate : also called Simple Rate (Signal Rate),

(Signal) $S \le N$ (Data) $\Rightarrow \frac{1}{r} = \frac{S}{N} \le 1 \rightarrow r \ge 1$

* 5.9 :

carrier frequency عالي



بتكرر نفسها

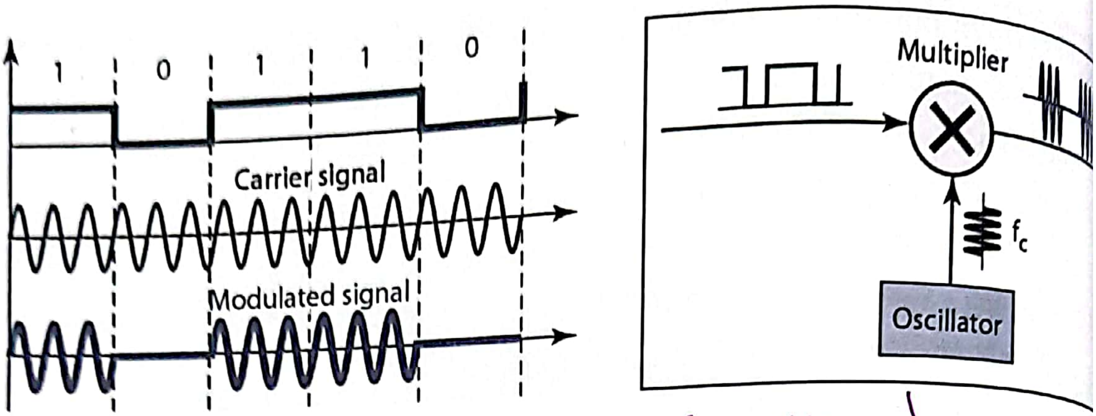
$* f_c = \frac{1}{T_c}$

* بالفترة 1 بضرب S(t) بـ d(t)
وبمنطقة الصفر بتساوي صفر .

"بضل بقسمها هون"

Five Apple

Figure 5.4 *Implementation of binary ASK*



لـ عمليًا لو بدي أبنيها بال Circuit عشان أضرب ال 2 signals

5.11

Example 5.3

لـ نبعث ال Data باتجاه واحد.

We have an available bandwidth of 100 kHz whic spans from 200 to 300 kHz. What are the carrie frequency and the bit rate if we modulated our data b using ASK with d = 1?

Solution

Carrier = Middle of the bandwidth = (200+300) / 2 = 250

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

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

لـ نفسه 50 baud لأنه S=N .

5.12

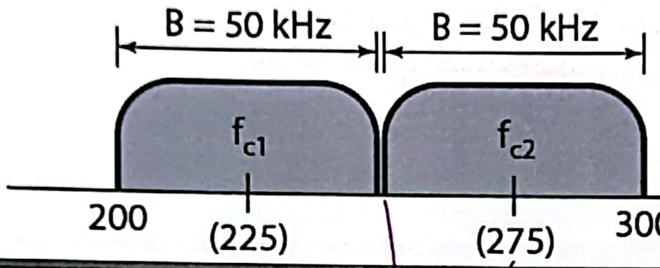
Example 5.4

لازم أخصص band للإرسال و band للاستقبال .

أرسل واستقبل مع بعض

In data communications, we normally use full-duplex links with communication in both directions. We need to divide the bandwidth into two with two carrier frequencies, as shown in Figure 5.5. The figure shows the positions of two carrier frequencies and the bandwidths. The available bandwidth for each direction is now 50 kHz, which leaves us with a data rate of 25 kbps in each direction.

Figure 5.5 *Bandwidth of full-duplex ASK used in Example 5.4*



*Frequency division Multiplexing: you own part of the bandwidth all the time.

*Time division Multiplexing: you own all the bandwidth at part of the time.

carrier carrier

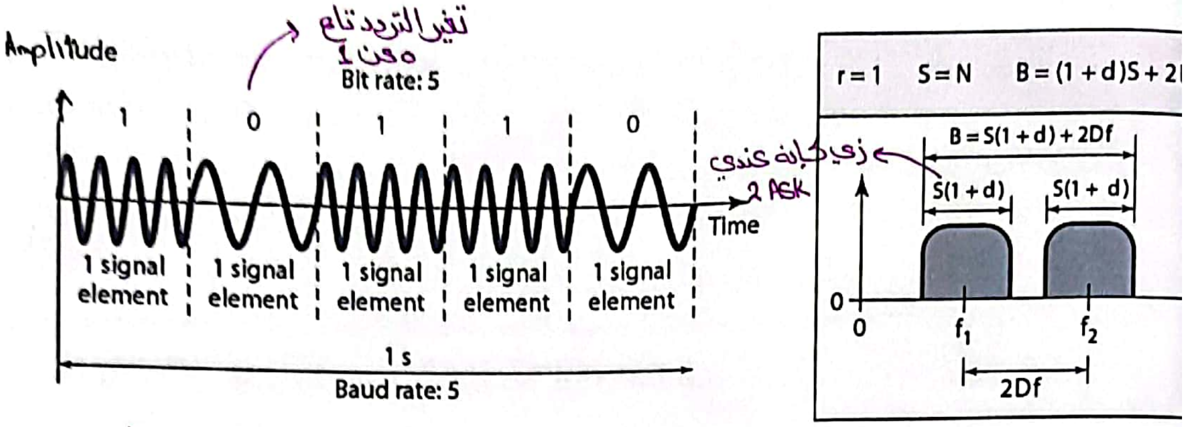
مسافة عشان مايصير بينهم تداخل

Frequency Shift Keying (FSK)

- FSK: varying the frequency of the carrier signal to represent binary 1 or 0
- Other signal characteristics remain constant as the frequency changes
- During each bit duration, the frequency remains constant
- FSK is mostly immune (مقاوم) to the transmission medium additive noise interference since FSK only cares about Frequency changes
- The Binary FSK (BFSK) can be thought of as two ASK signals, each with its own carrier frequency f_1 and f_2 * 00 → f_{c1} , 01 → f_{c2}
- f_1 and f_2 are $2\Delta f$ apart (ممكن ابعث 2 bits بس مبدأياً خليها 1 bit .)
- The BFSK required bandwidth is $B_{BFSK} = (1+d)\,S + 2\Delta f$
- What is the minimum value of $2\Delta f$?
- The baud rate is the same as the bit rate → $N = S$

* Minimum Bandwidth = 2S

re 5.6 *Binary frequency shift keying*



Continuous → يعني ال Signal الأصلية عبارة عن non-Periodic و Composite Signals.

Example 5.5

We have an available bandwidth of 100 kHz which spans from 20(...) to 300 kHz. What should be the carrier frequency and the bit rate i(...) we modulated our data by using FSK with d = 1?

" يكون دايمًا بالوسط "

Solution

This problem is similar to Example 5.3, but we are modulating b(...) using FSK. The midpoint of the band is at 250 kHz. We choose 2Δ(...) to be 50 kHz; this means

$$\boxed{B = (1 + d) \times S + 2\Delta f = 100 \Rightarrow 2S = 50 \text{ kHz} \quad S = 25 \text{ kbaud} \quad N = 25 \text{ kbps}}$$

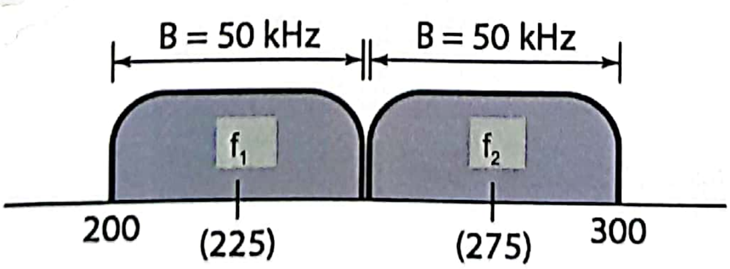
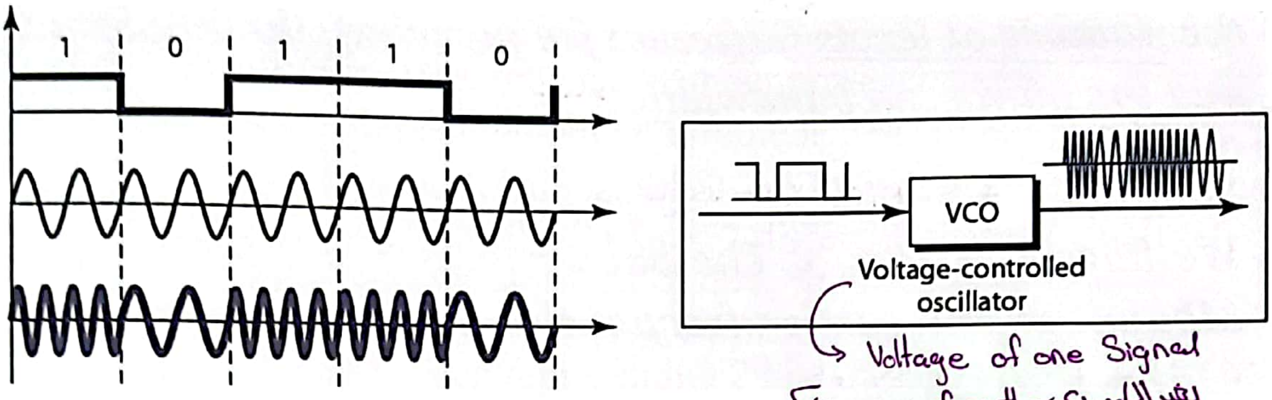


Figure 5.7 *Implementation of BFSK*



Voltage of one Signal
يغير ال Frequency of another Signal

Multi-Level FSK (MFSK)

بزيد → البت أكثر من
ال bit rate بأكثر من Pattern
frequency

00→ f_{c1}, 01→ f_{c2}, 10→ f_{c3}, 11→ f_{c4}

- More than two frequencies can be used to represent more than one bit each
  الضريبة اللي بدفعها إنه الـbandwidth عندي بزيد لإنه عدد ال freq بزيد
  - * For example: four different frequencies can be used to send 2 bits at a time
- However, each adjacent pair of frequencies must be $2\Delta f$ apart
- The MFSK required bandwidth is

عدد ال levels

لو كان 2 → وكأنه رجعنا لل FSK

$$B_{MFSK} = (1+d)\,S + (L-1)\,2\Delta f$$

- When d=0, the minimum Bandwidth $B_{MFSK} = L \times S$

بيعتمد بوضوح عال # of levels.

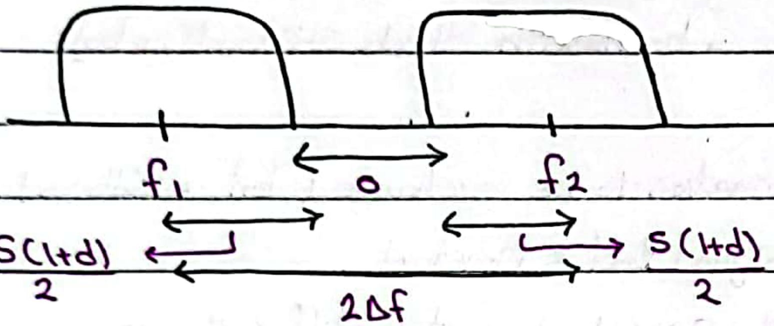
22/11/2022

* Revision :

Carrier frequency : $S(t) = \underset{①}{A} \sin(2\pi \underset{②}{f_c} + \underset{③}{\phi})$

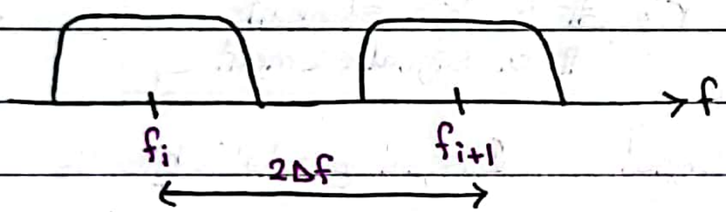
أي وحدة بتغيرهم من الثلاثة هدول ممكن نعمل Modulation

* 5.14 : Minimum value of $2\Delta f$?



$\therefore 2\Delta f = S(1+d) \rightarrow$ if $d = 0$, $2\Delta f = S$

* 5.18 : Minimum Bandwidth :



* $B_{MFSK} = (1+d)S + (L-1)2\Delta f$

$\rightarrow d = 0 \rightarrow B_{MFSK} = S + LS - S = L \times S$

$S + (L-1)S$

if $2\Delta F = S$

Example 5.6

We need to send data 3 bits at a time at a bit rate of 3 Mbps. The ~~carrier~~ center frequency is 10 MHz. Calculate the number of levels →= L *(different frequencies), the baud rate, and the minimum bandwidth.* → L×S

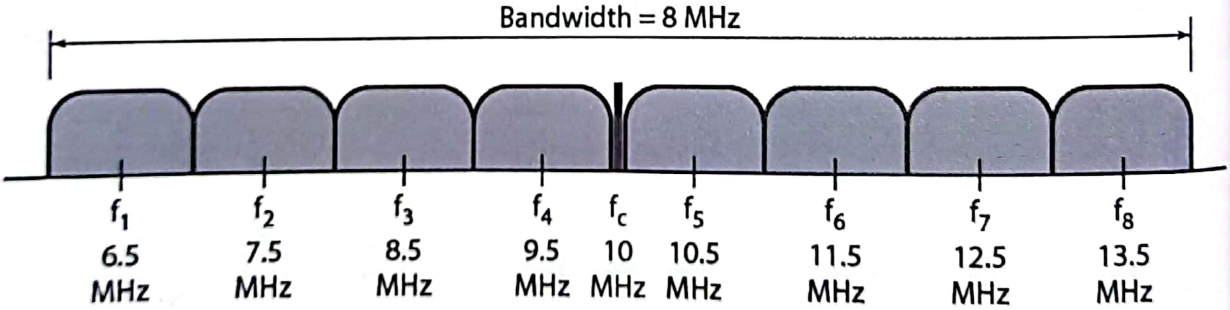
Solution   * S = N/r   * r = $\log_2 L$

We have L = 2^3 = 8. The baud rate is S = 3 MHz/3 = 1 Mbaud ➔ the carrier frequencies must be 1 MHz apart ($2\Delta f$ = 1 MHz). → = S

The bandwidth is B = 8 × 1MHz = 8 MHz

Figure 5.8 shows the allocation of frequencies and bandwidth.

Figure 5.8 *Bandwidth of MFSK used in Example 5.6*



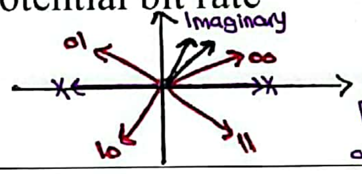
* أكثر واحد بتعرض للـ noise هو الـ ASK لإنو الـ Amplitude بضل يتغير.

Phase Shift Keying (PSK)

- PSK: varying the phase of carrier signal to represent binary 1 or 0
- Other signal characteristics remain constant as the phase changes
- During each bit duration, the phase remains constant
- PSK is mostly immune (ممانع) to the transmission medium additive noise interference since PSK only cares about phase changes
- PSK spectrum and bandwidth requirements are similar to ASK
- PSK is better than both ASK and FSK. Why?
- 2-PSK or Binary PSK (BPSK): uses 2 different phases (usually 0 and 180) each representing 1 bit of data ➔ N = S
- 4-PSK or Quad PSK (QPSK): uses 4 different phases (e.g.; 45, -45, 135, and -135) each representing 2 bits of data ➔ N = 2 S
- PSK is represented by a Constellation (or Phase-State) diagram
- PSK is limited by the ability of the receiver to distinguish small phase difference, therefore, limiting the potential bit rate

مرات بكون صعب جداً الـ reciever إنك تميز بين Phase و Phase.

الرسم الكوكبي

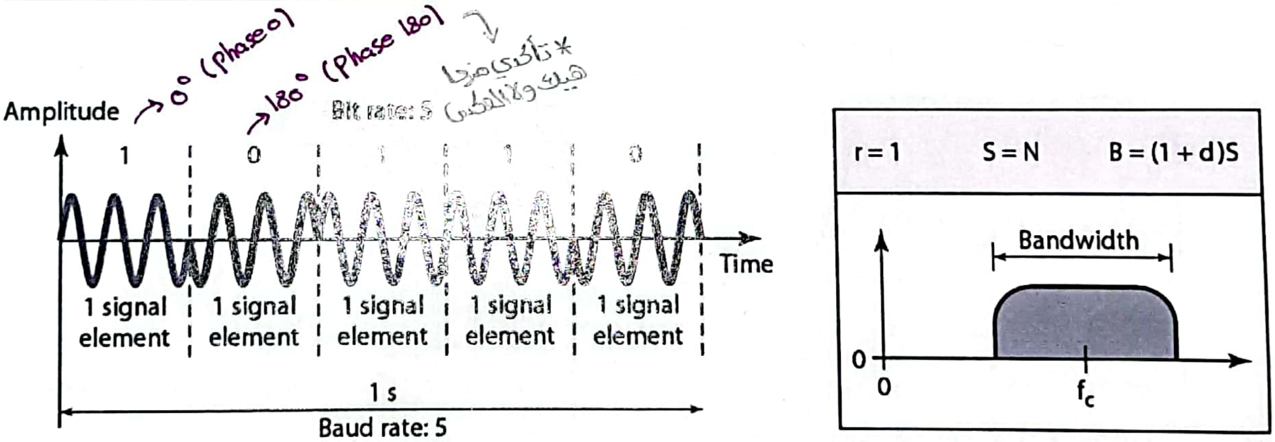


5.21

هلا حنحرك الخطين (Vector) البنفسجي بكون زدت الـ data rate بس صار عندي 4 phases والتفكير المنطقي أكثر إنك تقسم كمان الـ time زي بالأسود (real) فبصير يبعث عدد أكبر من الـ bits.

* الـ PSK أفضل من الـ FSK لأنها بتحتاج Frequency واحد والـ Bandwidth محدد وأقل وممانع للـ noise. (كأنه ماخذ ميزات الـ ASK والـ FSK سوا)

Figure 5.9 *Binary phase shift keying*



← ميزة الـ PSK على الـ ASK رغم إنه نفس هاي الرسمة؟ بنفوق عليه بإنه ممانع للـ noise

5.22

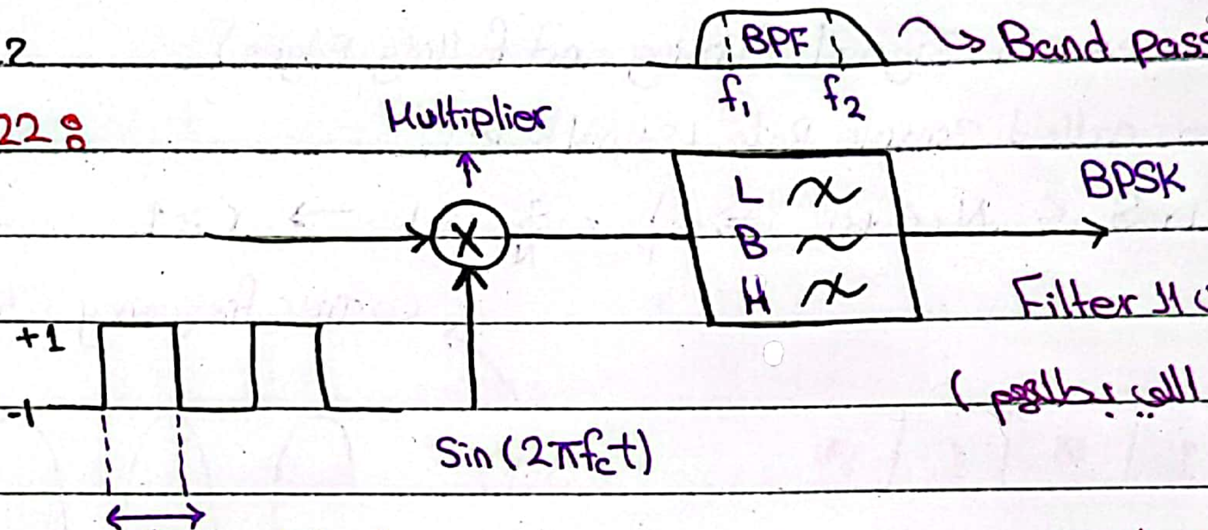
24/11/2022

*** Slide 5.22:**

1- BPSK:
↳ binary PSK

Polar NRZ ←

Signal with bit rate $N = \frac{1}{t_b}$



Band pass Filter ↓ الناتج بندخله عليه عشان يطلعلي من ال Bands Filter (اللي بالوسط هو اللي بطلعهم)

t_b ↳ bit duration (خلالها ما بتغير ال phase)

↳ time to transmit 1 bit.

← بميزهم بوضع إشارة عالي ما بمرقه :

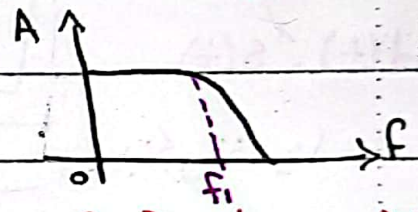
(LP)
L ~
B ≁
H ≁

or

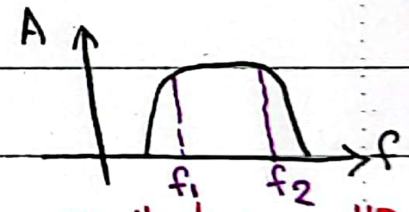
(BP)
L ≁
B ~
H ≁

↓ 3 types of Filter

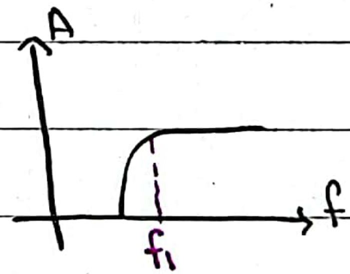
***Note:** 1- low pass LP



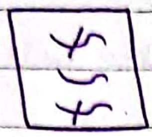
2- Band pass BP



3- High pass HP

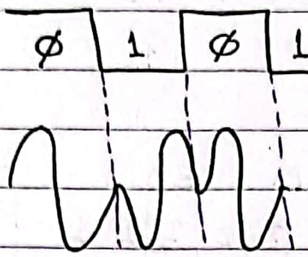


Five Apple

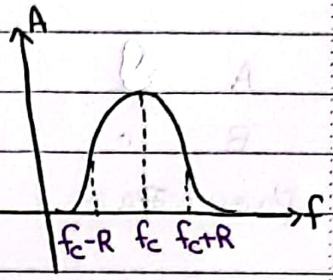


→ BPSK V(t) * بدنا نشوف الناتج الي بطلعولنا *

* Alternative 0 and 1 :



Time Domain ← … Frequency Domain

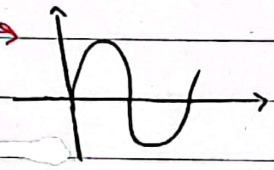


→ In Math : $V(t) = A\sin(2\pi f_c t)$, binary Ø.

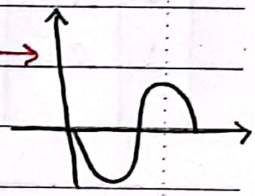
$V(t) = -A\sin(2\pi f_c t) = A\sin(2\pi f_c + \pi)$, binary 1

or $V(t) = \pm A\sin(2\pi f_c t)$, binary Ø or 1.

* Note : * Phase Ø, b = 0 →

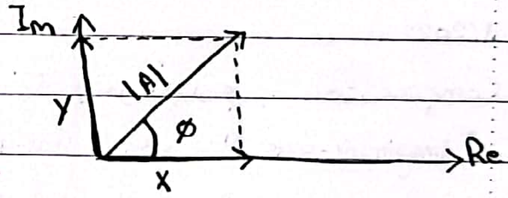


* phase 180°, b = 1 →



* Note : phase Diagram : $A\sin(\omega_c t + \phi)$

$A\angle\phi \iff X + jY$



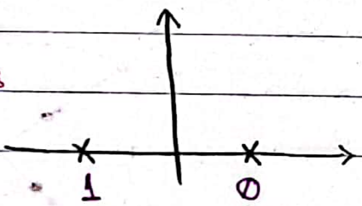
* $A = \sqrt{X^2 + Y^2}$

* $\phi = \tan^{-1}\left(\frac{Y}{X}\right)$

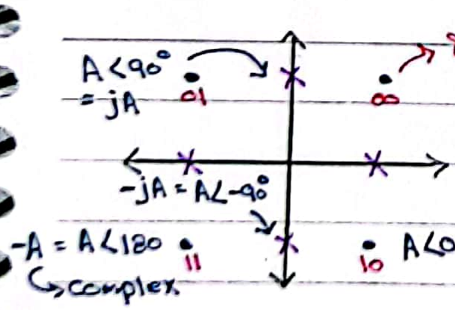
* Phase Diagram :

1 Amplitude ←

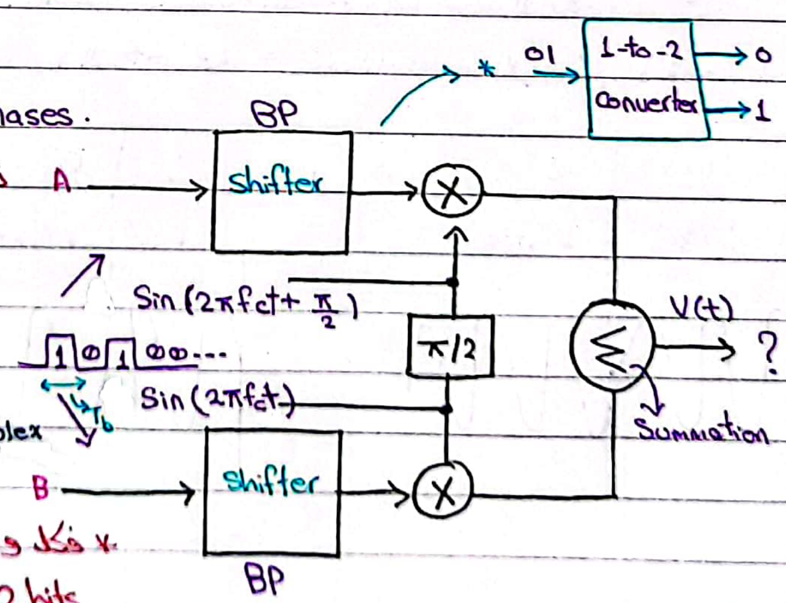
and 2 phases



2- QPSK : we use 4 phases.



* phase Diagram *



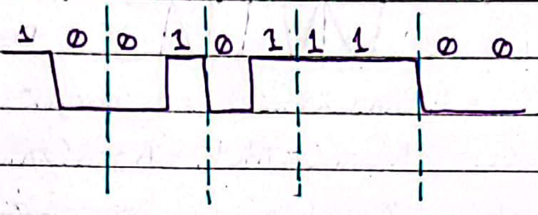
* كل وحدة من ال 4 نقاط جوا مرسومة بتمثل

. 2 bits

* Suppose we have the following Sequence of bits : 1001011100

* بدي اكمل table لأفصل وأمين ال bits عن بعض.

	10	01	01	11	00
A	1	0	0	1	0
B	0	1	1	1	0
Phase	$7\pi/4$	$3\pi/4$	$3\pi/4$	$5\pi/4$	$\pi/4$



*

A	B	Phase
0	0	$\pi/4$
0	1	$3\pi/4$
1	0	$7\pi/4$
1	1	$5\pi/4$

27/11/2022

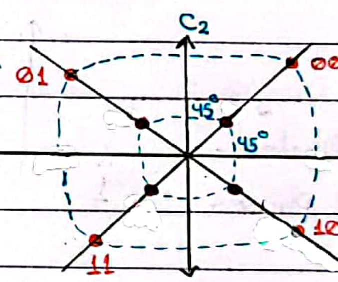
لتقسيم ال phases فقط.

* كل Compination من ال 4 bits اللي مشروحين فوق بقدر أعبرهم ب Compination من 2 vectors المسافة بين انهم 90° (واحد عال real-axis وواحد عال Imaginary-axis :

هدول الأربعة الهم نفس الـ Amplitude بس 4 phases

* لو بدي أزيد ال data rate بزيد عدد ال phases أو النقاط (هيك بكون لعبت بال Phases وال Magnitude) زي اللون الزهري عالرسمة. الـ Magnitude عرضة للـ noise مرات.

* ال data rate : عدد ال Symbols اللي ببعثهم لكل Signal element واحد.



* $\frac{2\pi}{2^b}$ ← # of bits

↓

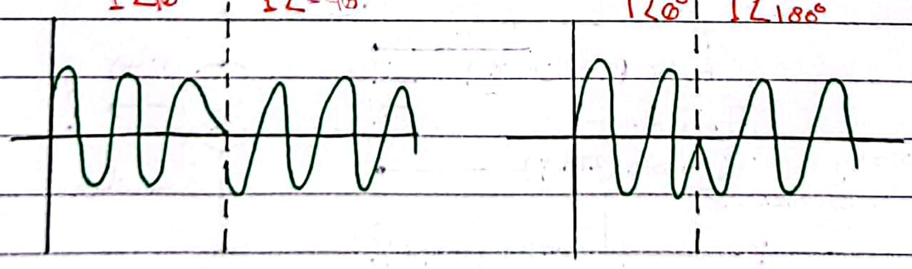
عبارة عن Sin بس بحسب اله تردد مقداره f_{c_1}.

A : +1 −1 1 0

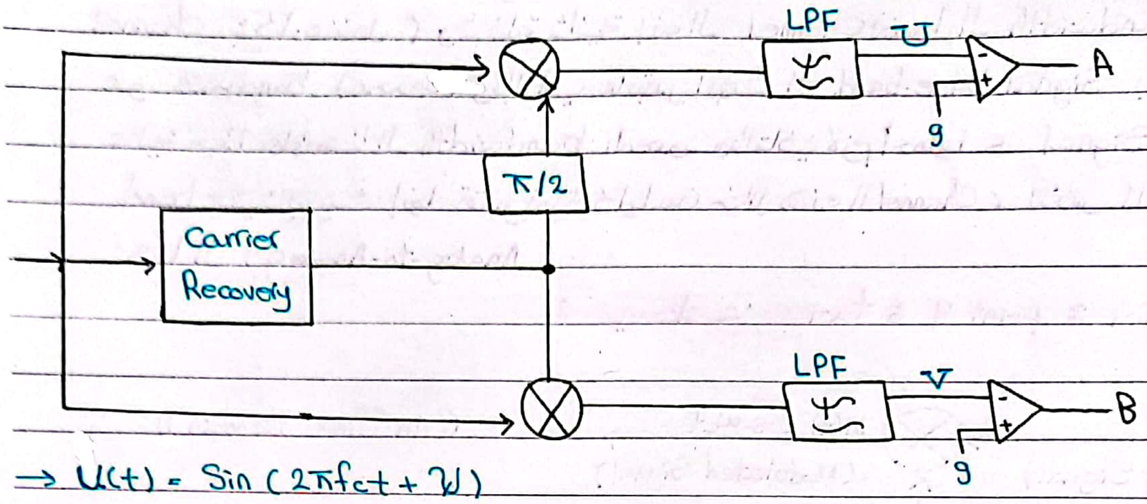
$1\angle 90^\circ$ | $1\angle -90^\circ$

B : +1 −1 1 0

$1\angle 0^\circ$ | $1\angle 180^\circ$



* 4-PSK Demodulator :



→ $U(t) = \sin(2\pi f_c t + \psi)$

* Case-1 : $u(t)\,[\sin(2\pi f_c t)]$

$$= \sin(2\pi f_c t + \psi)\,\sin(2\pi f_c t)$$

using trigometric identity $= \underbrace{\frac{1}{2}\cos\psi - \frac{1}{2}\cos(4\pi f_c t + \psi)}$

↓

$\boxed{\text{LPF}}$ → "تيقص العالي"

↓

$$= \frac{1}{2}\cos\psi$$

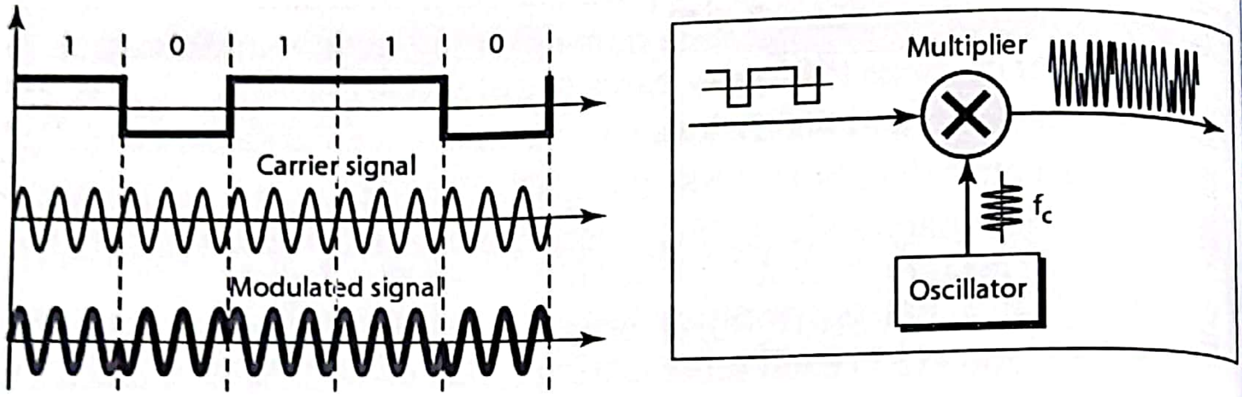
* Case-2 : $\sin(2\pi f_c t + \psi)\,[\sin(2\pi f_c t + \frac{\pi}{2})]$

$$= \frac{1}{2}\cos(\psi - \frac{\pi}{2}) - \frac{1}{2}\cos(4\pi f_c t + \psi + \frac{\pi}{2})$$

using LPF $\Rightarrow \frac{1}{2}\cos(\psi - \frac{\pi}{2})$

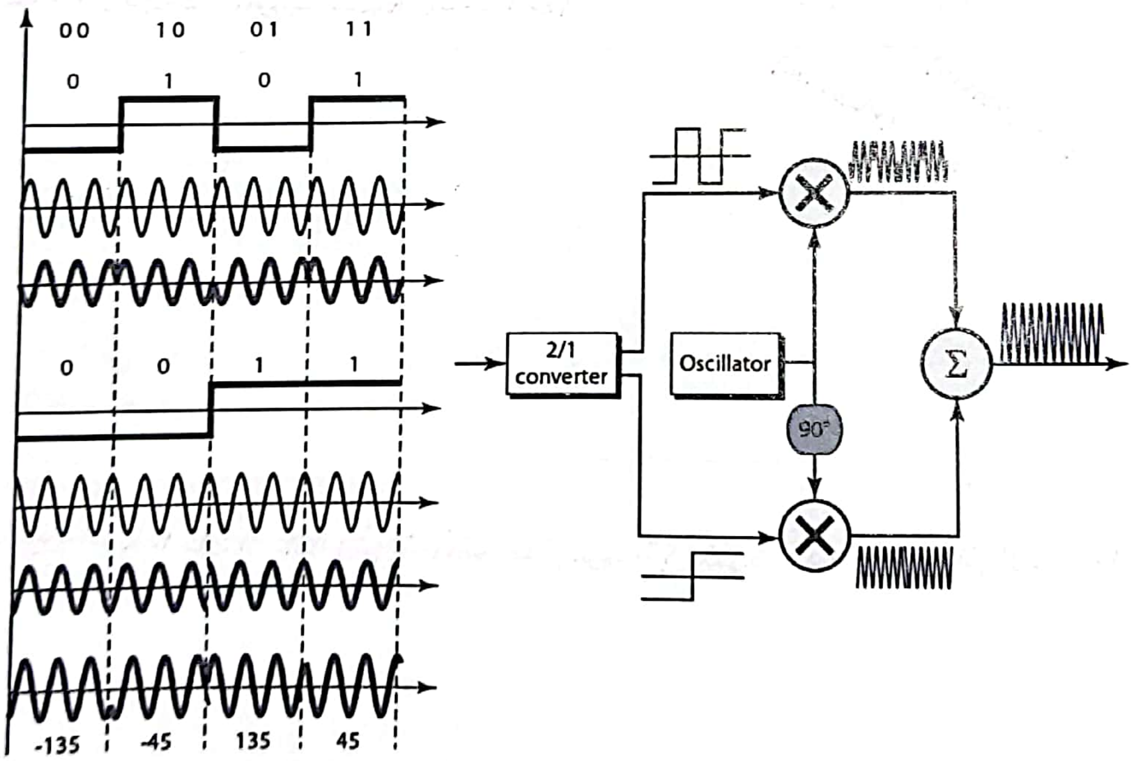
ψ	U	V	A	B
$\pi/4$	$\frac{1}{2\sqrt{2}}$	$\frac{1}{2\sqrt{2}}$	0	0
$3\pi/4$	$\frac{1}{2\sqrt{2}}$	$\frac{-1}{2\sqrt{2}}$	0	1
$5\pi/4$	$\frac{-1}{2\sqrt{2}}$	$\frac{-1}{2\sqrt{2}}$	1	1
$7\pi/4$	$\frac{-1}{2\sqrt{2}}$	$\frac{1}{2\sqrt{2}}$	1	0

Figure 5.10 *Implementation of BPSK*



5.23

Figure 5.11 *QPSK and its implementation*



5.24

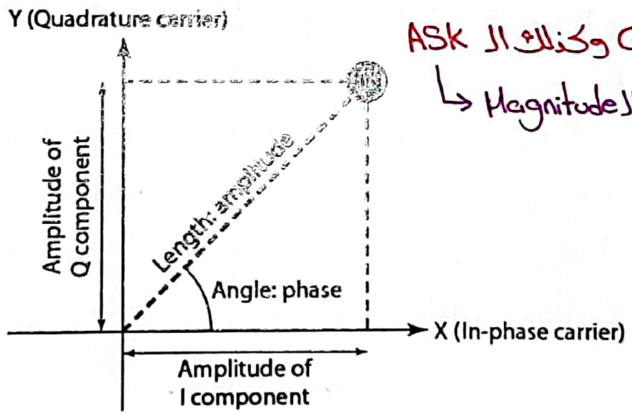
Example 5.7

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

Solution

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

Figure 5.12 *Concept of a constellation diagram*



* بنقدر نرسم ال QPSK وكذلك ال ASK
بنغير فقط ال Magnitude ←

- Used to define the amplitude and phase of a signal element when using two carriers or when dealing with multi-level shift keying
- A signal element is represented by a dot in the diagram, of which:
  - The projection on the X axis defines the peak amplitude of the in-phase component
  - The projection on the Y axis defines the peak amplitude of the quadrature component
  - The length of the line connecting the point to the origin is the peak amplitude of the signal element

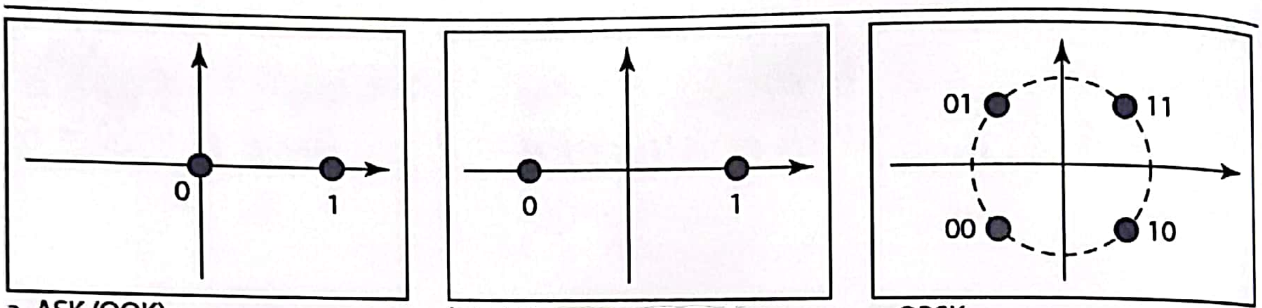
Example 5.8

Show the constellation diagrams for an ASK (OOK), BPSK, and QPSK signals.

Solution

Figure 5.13 shows the three constellation diagrams.

Figure 5.13 *Three constellation diagrams*



5.27

Quadrature Amplitude Modulation (QAM)

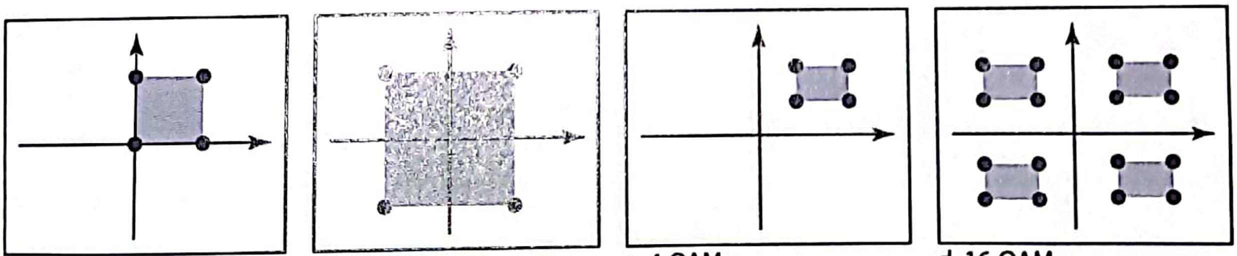
- QAM: Varying both the peak amplitude and the phase of the carrier signal to represent binary combination
- During each bit duration, the phase and amplitude remain constant
- Theoretically, any number of measurable changes in phase and amplitude can be combined to give several variations in the signal
- The number of phase shifts is always greater than the amplitude shifts. Why?
- The greater the ratio of phase to amplitude shifts, the better the noise immunity
- QAM spectrum and bandwidth requirements are similar to ASK and PSK

5.28

Note

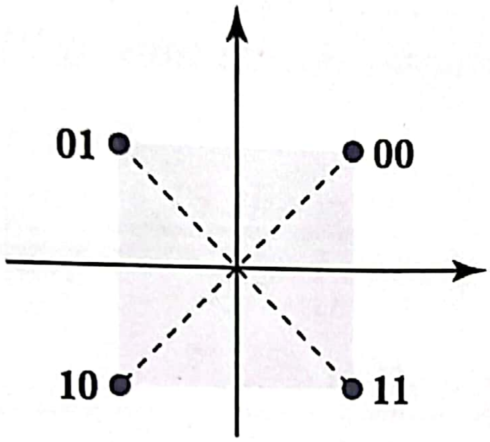
Quadrature amplitude modulation is a combination of ASK and PSK.

Figure 5.14 *Constellation diagrams for some QAMs*

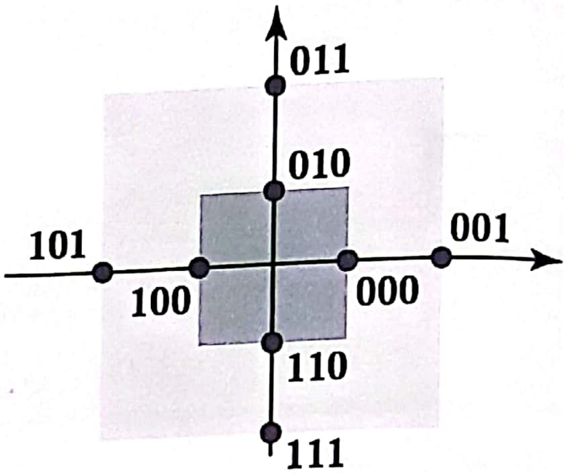


a. 4-QAM   b. 4-QAM   c. 4-QAM   d. 16-QAM

Examples of 4-QAM and 8-QAM constellations



4-QAM
1 amplitude, 4 phases



8-QAM
2 amplitudes, 4 phases

Magnitudes

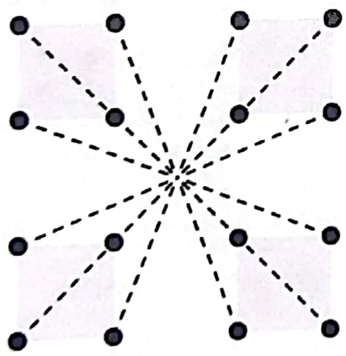
لو أخذنا ال X-axis لحالة :

→ Phase ثابت / 2 Amplitudes

8-QAM → $2^3 = 8$ → 3 bits

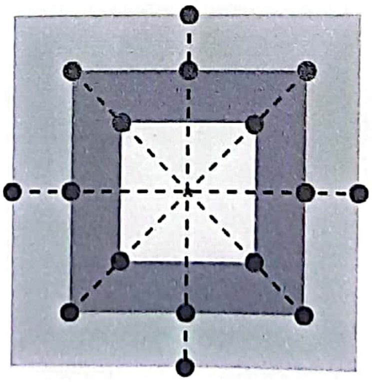
Examples of 16-QAM constellations

* 3 amplitudes, 12 phases



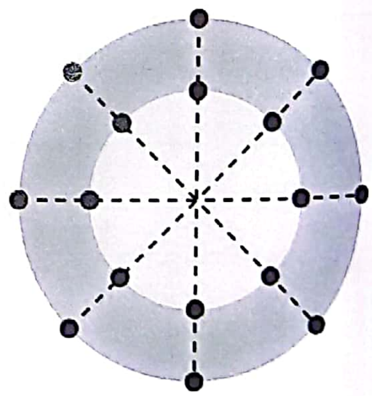
16-QAM

* 4 amplitudes, 8 phases



16-QAM

* 2 amplitudes, 8 phases



16-QAM

Example

A constellation diagram consists of eight equally spaced points on a circle. If the bit rate is 4800 bps, what is the baud rate? → Amplitude ثابت / Phase بتغير

Solution

* 8-QAM → $2^3 = 8$ → 3 bits

* $N = 4800$ bps , $S = ?$, * $r = \frac{3}{1}$
* $\frac{N}{S} = r$ → $S = \frac{N}{r} = \frac{4800}{3} = 1600$ baud

The constellation indicates 8-PSK with the points 45 degrees apart. Since $2^3 = 8$, 3 bits are transmitted with each signal unit. Therefore, the baud rate is

$$4800 / 3 = 1600 \text{ baud}$$

Example

* $S = 1000$ baud, $N = ?$, $r = ?$
* 16-QAM → 4 bits → $r = \frac{4}{1} = 4$
* $N = Sr = 1000 * 4 = 4000$ bps

Compute the bit rate for a 1000-baud 16-QAM signal.

مناسب لأنه بزيد ال data rate بس مش ع حساب
ال bandwidth.

Solution

A 16-QAM signal has 4 bits per signal unit since

$$\log_2 16 = 4.$$

Thus,

$$(1000)(4) = 4000 \text{ bps}$$

Example

Compute the baud rate for a 72,000-bps 64-QAM signal.

Solution

A 64-QAM signal has 6 bits per signal unit since

$$\log_2 64 = 6.$$

Thus,

$$72000 / 6 = 12{,}000 \text{ baud}$$

Bit and baud rate comparison

Modulation	Units	Bits/Baud	Baud rate	Bit Rate
ASK, FSK, 2-PSK	Bit	1	N	N
4-PSK, 4-QAM	Dibit	2	N	2N
8-PSK, 8-QAM	Tribit	3	N	3N
16-QAM	Quadbit	4	N	4N
32-QAM	Pentabit	5	N	5N
64-QAM	Hexabit	6	N	6N
128-QAM	Septabit	7	N	7N
256-QAM	Octabit	8	N	8N

5-2 ANALOG-TO-ANALOG CONVERSION

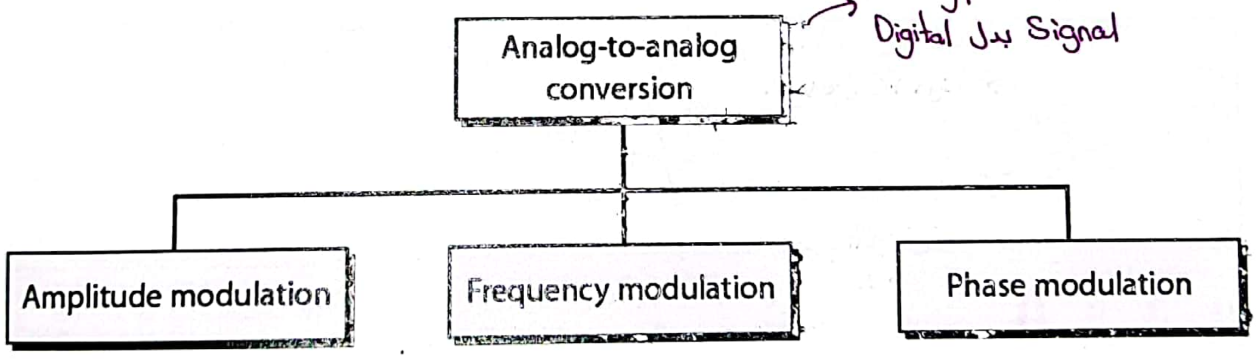
Analog-to-analog conversion is the representation of analog information by an analog signal. One may ask why we need to modulate an analog signal; it is already analog. Modulation is needed if the medium is bandpass in nature or if only a bandpass channel is available to us.

Topics discussed in this section:

Amplitude Modulation
Frequency Modulation
Phase Modulation

Figure 5.15 *Types of analog-to-analog modulation*

Analog باستخدام Digital بدل Signal



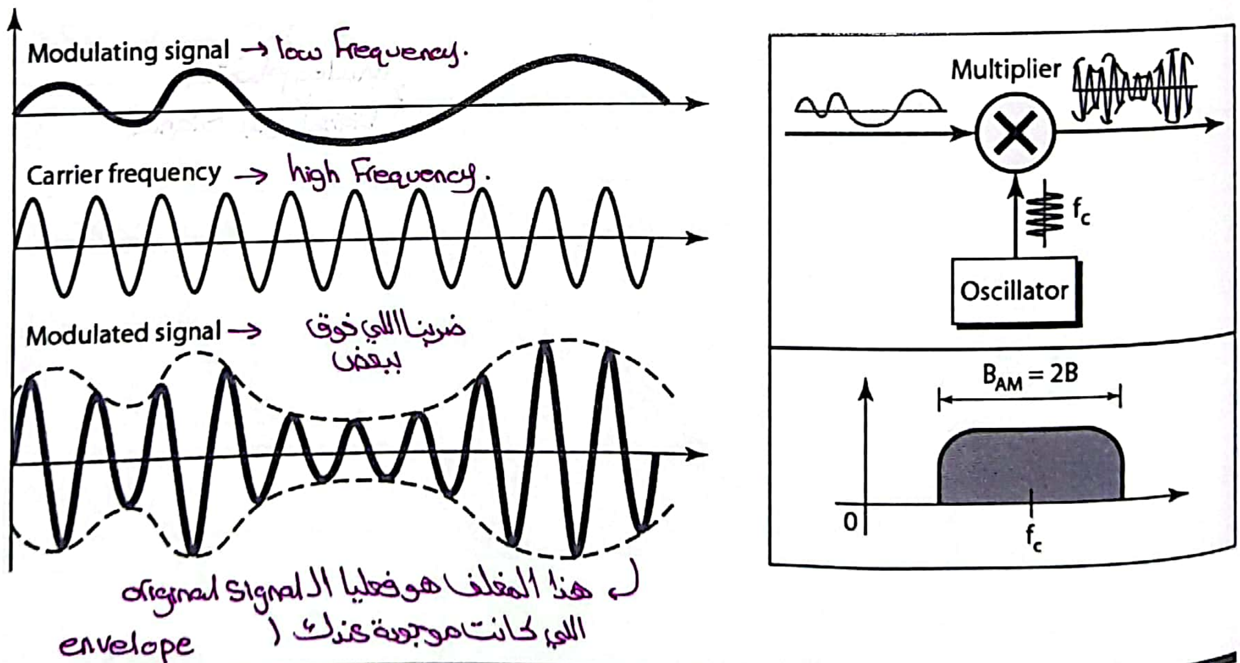
Amplitude Modulation (AM)

← بالعادة بتكون Voice signal أو Video Signal

- * The amplitude of the carrier (or modulated) signal changes with the amplitude changes of the modulating signal
- * The frequency and phase of the modulated signal remain constant
- * The modulating signal becomes the envelope (i.e.; the outer shape) of the modulated signal
- * The bandwidth of the AM signal is equal to twice the bandwidth of the modulating signal الصوت بال radio مثلاً ما بوصل كامل زي الصوت بالطبيعة (8 kHz)
- * The bandwidth of an audio signal (voice and music) is 5 kHz
- * AM radio stations are assigned 10 kHz band per channel لكل قناة بال radio بنعطي 10kHz) محددة من قبل جهات معنية ← Standard)
- * Every other band is used as a guard band to prevent interference among adjacent channels لـ مساحة عشان ما يتداخلوا مع بعض 2 bandwidth.

مرات بكون الصوت بالتلفزيون واضح كثير وهذا بكون بسبب إنه في كثير Digital signals بتطلع بعدد عالي من ال bits بس هون احنا بنحكي Analog.

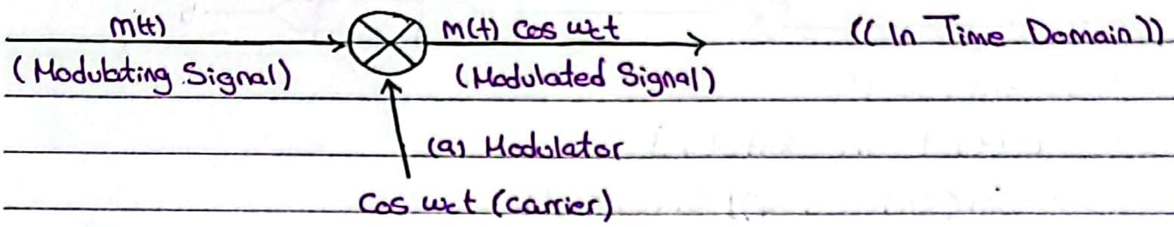
Figure 5.16 *Amplitude modulation*



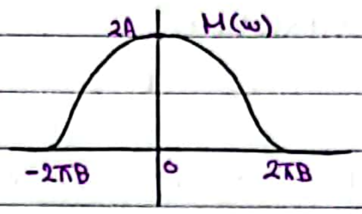
29/11/2022

* تحويل الـ digital إلى Analog بنحتاجه لنقدر نعمل Multiplexing (نحط كل channel بشكل منفصل) وشغلة ثانية إنو الـ Channel نفسها الـ bandwidth تبعها هو bandpass (محدود) وبالتالي مابقدر ابعت الـ base-band Signal بـ Frequencies عالية جدا على هذا الـ bandwidth المحدود وبالتالي لازم أحملها ع Signal ثانية الها band معين بحيث إنها تقدر تمرق كلها من خلال هذه الـ Channel ، لنفس الفوائد في الـ digital هنا الـ (Analog-to-Analog).

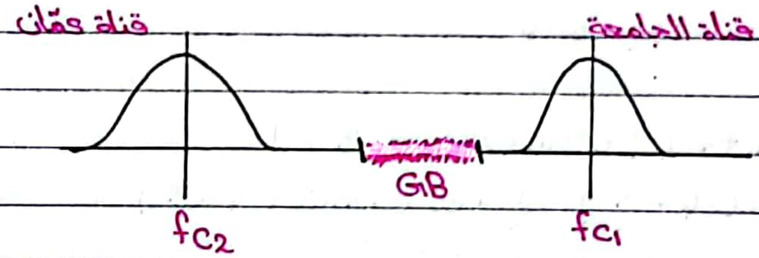
* Slide 5.39 : point 4 : twice ليه طلعت :



→ Frequency Domain :



* Slide 5.39 : Point 7 : guard band :



* ممكن يجي واحد يخرب استقبال قناة الـ radio بإنه يبث noise بـ Power عالي على نفس الـ Frequency فيكون خرب عليك استقبال الإشارة ، بـ ch6 حنعرف كيف حنتفادى هذا الموضوع بإني أغير الـ Carrier Frequency تبعي خليه يروح مكان ثاني بس لازم أغير برضه الـ reciever عشان نلقط بس برضه ممكن يلقط معي ويخرب علي فالحل إني أظل أغير الـ Freq. بشكل عشوائي randomly من جهة الـ Sender بس الـ reciever كيف ممكن يظل يلقط التغيير؟
pseudo random ← بستخدم عشان يطلعلي The same sequence of random # بكل run مش زي الـ random العادي ، فهيك بصير أصعب عليه يخترقك إذا ما عرف الـ Sequence .

Note

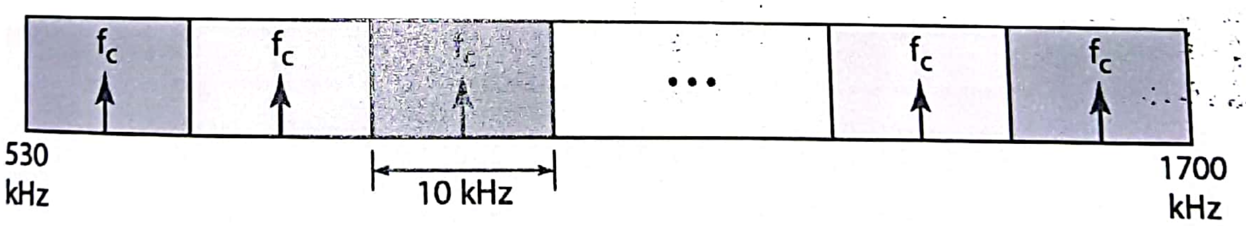
The total bandwidth required for AM can be determined from the bandwidth of the audio signal: $B_{AM} = 2B$.

« أهم نقطة »

Figure 5.17 *AM band allocation*

guard band

بالعادة في GB بين كل قناة وقناة



← بتغير ال Frequency بدل ال Amplitude

Frequency Modulation (FM)

- The frequency of the carrier (or modulated) signal changes with the amplitude changes of the modulating signal
- The peak amplitude and phase of the modulated signal remain constant

← بالكتب بالعادة 8 بس لإنه مش كل شيئ دقيق 100% .

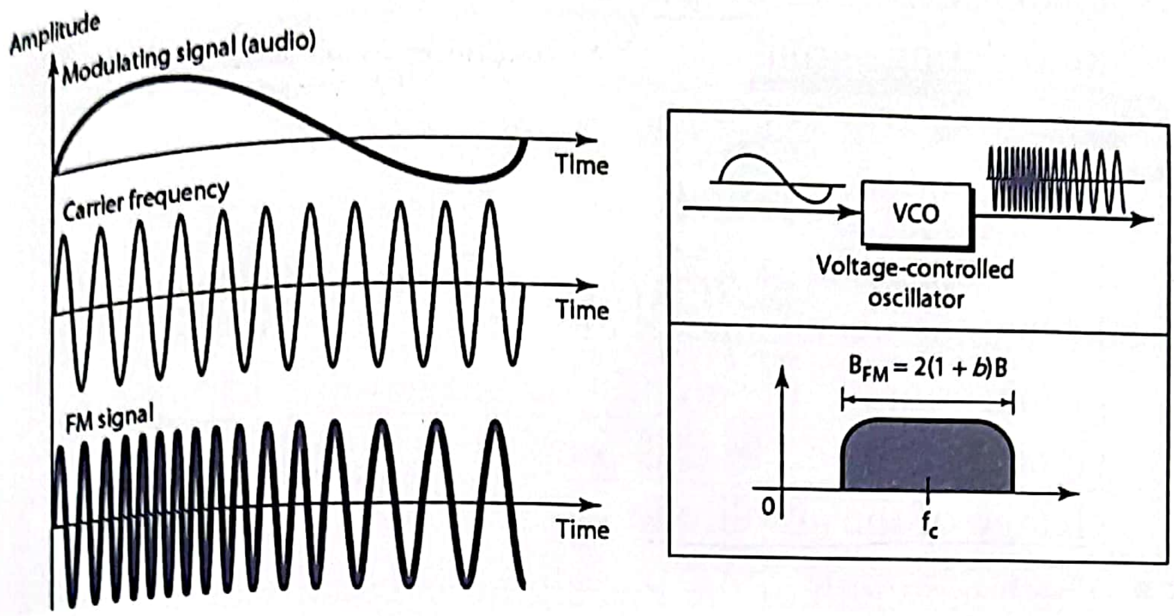
- The bandwidth of the FM signal is usually about 10 times the bandwidth of the modulating signal
- The bandwidth of a stereo audio signal (voice and music) is about 15 kHz
- FM radio stations are assigned 200 kHz band per channel
- Every other band is used as a guard band to prevent interference among adjacent channels

Note

The total bandwidth required for FM can be determined from the bandwidth of the audio signal: $B_{FM} = 2(1 + \beta)B$.

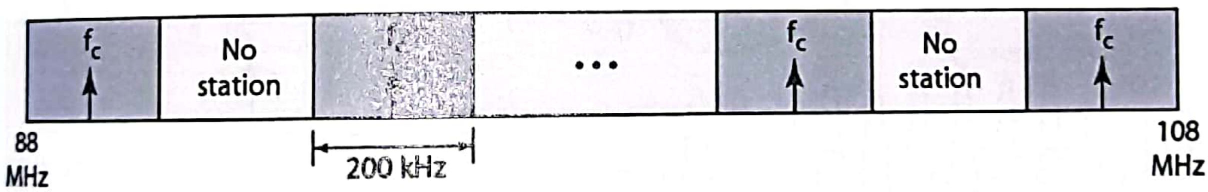
← Parameter متعلق بال Modulation ، بالعادة = 4 .

Figure 5.18 *Frequency modulation*



5.45

Figure 5.19 *FM band allocation*



5.46

Phase Modulation (PM)

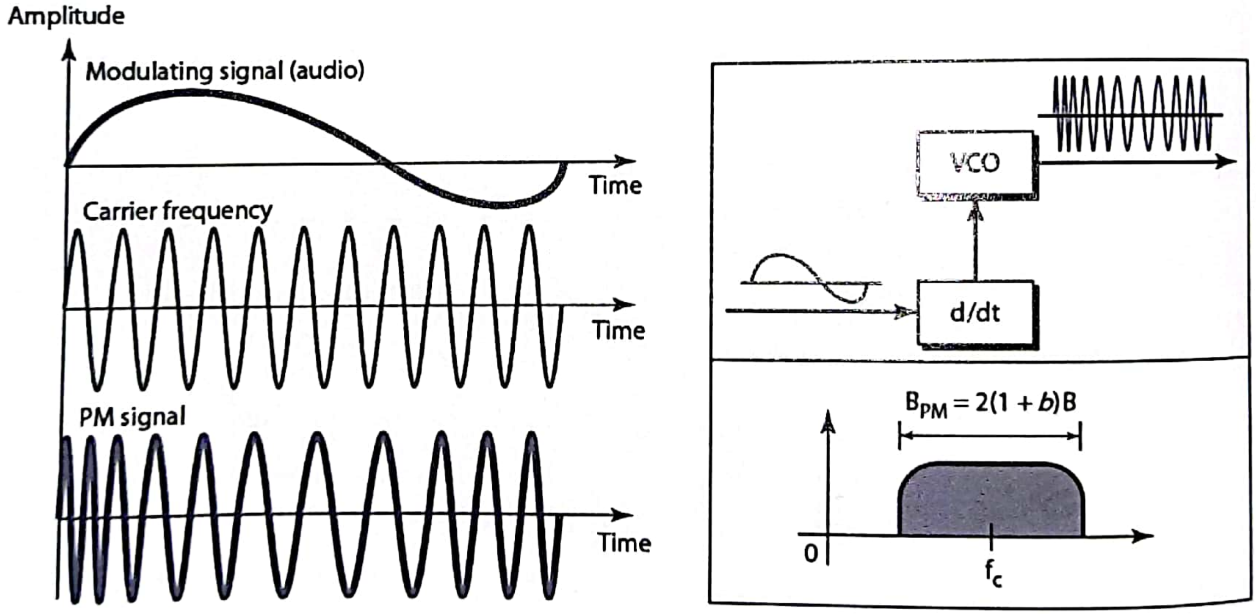
- The phase of the carrier (or modulated) signal changes with the amplitude changes of the modulating signal

بعتمد عمقدار تغير الـ amplitude (derivative)

- The peak amplitude and frequency of the modulated signal remain constant
- PM is the same as FM except that in FM, the instantaneous change in the carrier frequency is proportional to the amplitude of the modulating signal, while in PM it is proportional to the rate of change of the amplitude
- The bandwidth of the PM signal is usually abut 6 times the bandwidth of the modulating signal

بالوسط بين AM و FM

Figure 5.20 *Phase modulation*



Note

*The total bandwidth required for PM can be determined from the bandwidth and maximum amplitude of the modulating signal:

$$B_{PM} = 2(1 + \beta)B.$$

$\underline{\underline{2}}$

Data Communications and Networking Fourth Edition

Forouzan

Chapter 6

Bandwidth Utilization: Multiplexing and Spreading

الـ link الـbandwidth
كبير كثير فمش منطق
يكون فيها بس user واحد
فحيكون عنّا أكثر من user
يعبي الـ link والـbandwidth
كامل.

لـ حكون بدي أزيد الـbandwidth
لأغراض مختلفة مثلاً لزيادة الـSecurity

* link ← الها bandwidth كثير كبير.

* Channel ← تحت استخدام الـ user (في حالة الـ audio مثلاً 4 kHz).

Note

الاستخدام الحكيم

Bandwidth utilization is the wise use of available bandwidth to achieve specific goals.
Efficiency can be achieved by multiplexing
Privacy and anti-jamming can be achieved by spreading.

6-1 MULTIPLEXING

* ***Whenever the bandwidth of a medium linking two devices is greater than the bandwidth needs of the devices, the link can be shared. Multiplexing is the set of techniques that allows the simultaneous transmission of multiple signals across a single data link. As data and telecommunications use increases, so does traffic.***

<u>Topics discussed in this section:</u>

1. * **Frequency-Division Multiplexing**
2. * **Wavelength-Division Multiplexing**
3. * **Synchronous Time-Division Multiplexing**
4. * **Statistical Time-Division Multiplexing**

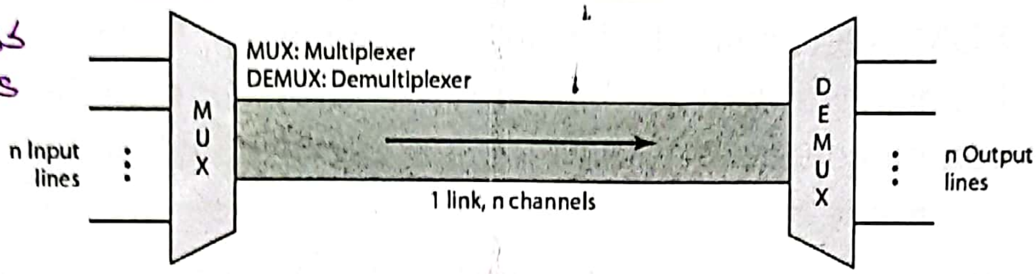
Multiplexing

أكثر من Device بدهم
يطلعوا ع link واحدة
بس الها bandwidth كبير.

* ▪ If the link bandwidth is greater than the bandwidth needed by a single device, then the link can be shared by multiple devices → bandwidth sharing
* ▪ Multiplexing is the set of techniques used to simultaneously transmit multiple signals across a single data link
* ▪ Multiplexing creates multiple transmission channels over a single communication link → حصة أو أكثر من ال bandwidth لل user
* ▪ Several recent transmission media such as optical fibers and satellite microwave links have much higher bandwidth than the average bandwidth needed for most applications
* ▪ It is essential to optimize the utilization of a given bandwidth in order to <u>reduce costs</u> and prevent wastage. That is, to improve the cost-effectiveness and utilization of the resources

Figure 6.1 *Dividing a link into channels*

كل User
Channel نعيّن
وحدة (أبسط
(Senario

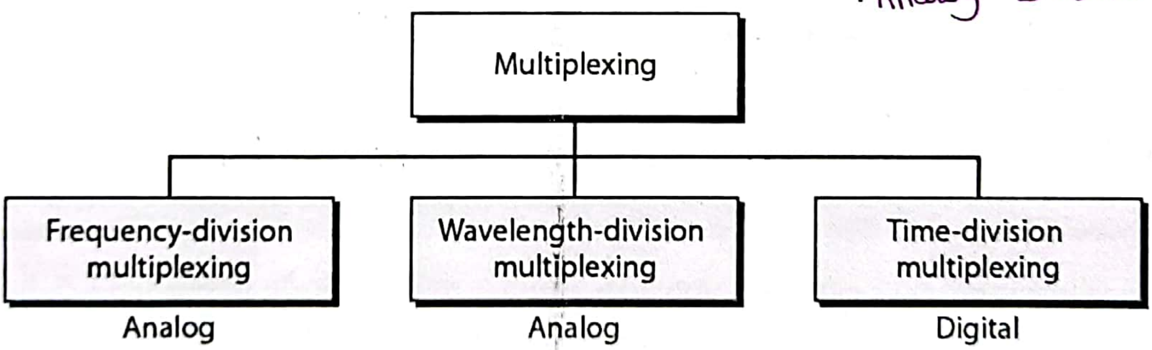


- * The multiplexer (MUX) combines several channels into one stream (many-to-one conversion)
- * The demultiplexer (DEMUX) separates the stream back into the original channels (one-to-many conversion)
- * The link refers to the physical medium
- * The channel refers to the portion of the link that carries the transmission between any pair of lines

حصة الUser من الlink

Figure 6.2 *Categories of multiplexing*

*حتى الDigital ممكن
نحولها ل Analog.



لـ كل الSignals
بتكون Digital

Frequency Division Multiplexing (FDM)

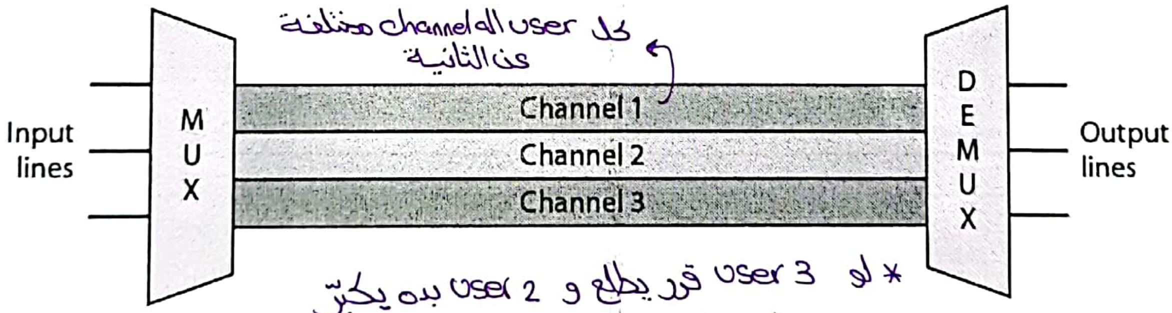
- FDM is an analog technique that can be applied when the bandwidth of the link in Hertz is greater than the combined bandwidths of the individual signals to be transmitted
- Each signal is modulated with a different carrier frequency
- The modulated signals are then combined to form a composite signal, which is transmitted over the link
- The carrier frequencies should be separated enough using guard bands to prevent overlapping or interfering with neighboring signals

مسافة ضيق بين الـ Channels الهدف منها ما تتداخل الـ Channels مع بعض.

6.7

كيف أقدر أطلع أكثر من user أو Channel في band وتطلعهم كلهم ضمن الـ bandwidth تبع الـ Link.

Figure 6.3 *Frequency-division multiplexing*



كل user الـ channel مختلفة عن الثانية

* لو user 3 قرر يطلع و user 2 بده يكبّر حصته بقدر ياخذ حصة user3 بس أكيد بده يدفع ثمن هاي الزيادة أكيد.

6.8

Note

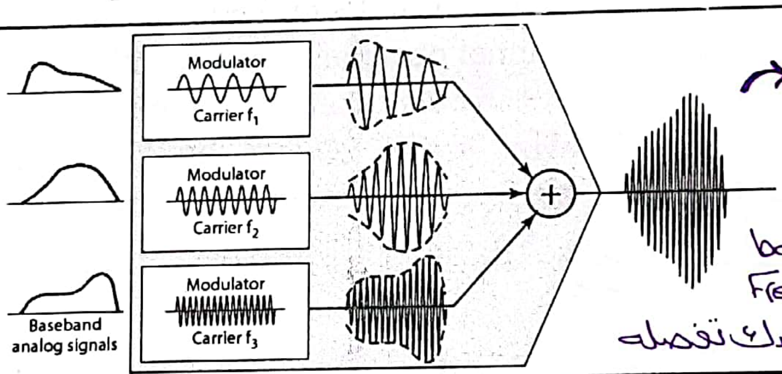
ميزتها مش معقدة

* **FDM is an analog multiplexing technique that combines analog signals.**

* في مشكلة حنكتشفها بالسايمنت الثاني : كيف نحل مشكلة إنه اذا اجاتنا Phase 180 زيادة حيصير استقبال ال bits كله غلط فكيف نحله ؟

6.9

Figure 6.4 *FDM process*



لو بدي أرجع أفصلهم عن بعض بستخدم low pass Filter أو band-pass Filter centered الـ Frequency تبع الـ Carrier الي بدك تفصله

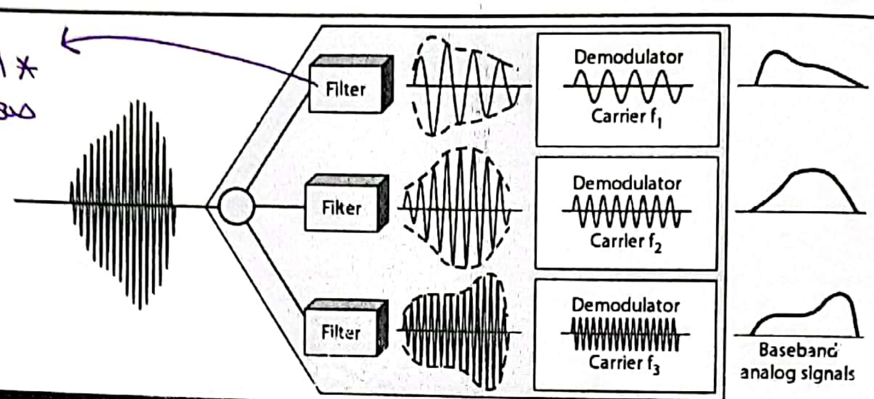
- * The input signals have similar frequency ranges and bandwidth requirements
- * Each signal is shifted in frequency away from the other signals
- * The modulated input signals are then combined to form a single composite signal, whose bandwidth is equal to, at least, the sum of the all input signals' bandwidths

مش exactly عشان ال guard band الموجودين بين ال users.

6.10

Figure 6.5 ***FDM demultiplexing example*** In Time Domain

Design Process الـ *
صعبة بس مجرد ما نتنفذ
خلص بصير الوضع
تمام.



- * A series of filters are used to decompose the multiplexed signal into its originally modulated signals
- * The individual modulated signals are demodulated and shifted back to their original frequency ranges
- * The original signals are then passed to their intended recipients

6.11

Example 6.1

→ Size of bandwidth = 4 kHz

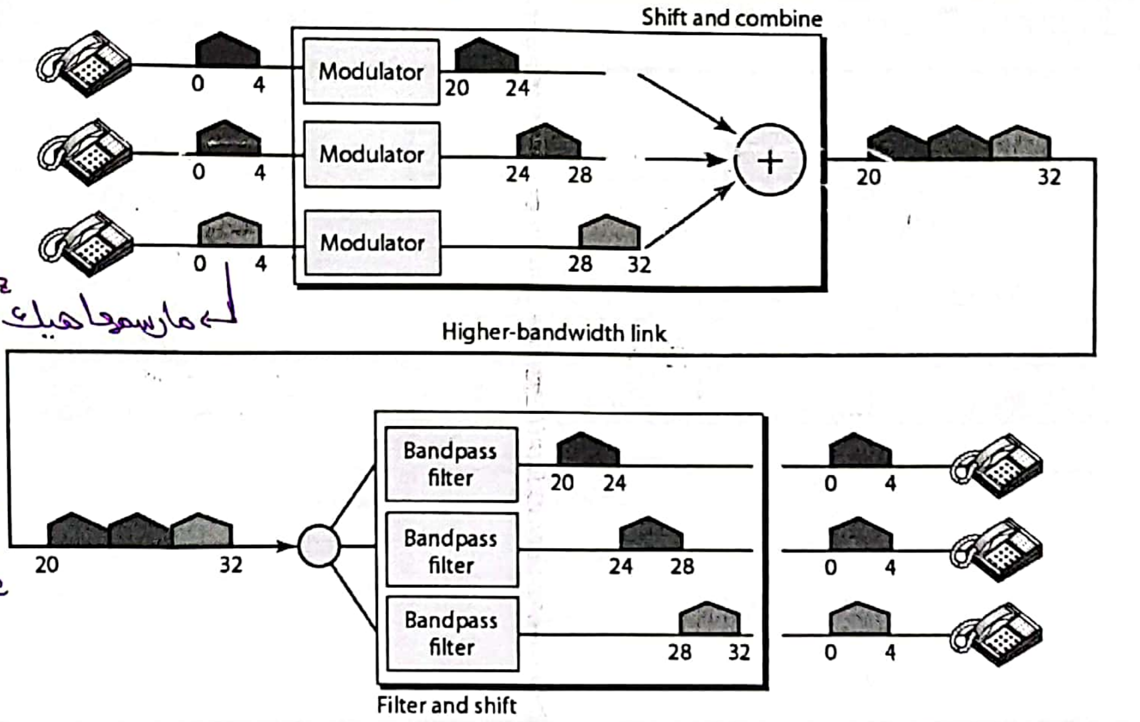
Assume that a voice channel occupies a bandwidth of 4 kHz. We need to combine three voice channels into a link with a bandwidth of 12 kHz, from 20 to 32 kHz. Show the configuration, using the frequency domain. Assume there are no guard bands. → ملزقين ببعض

Solution

We shift (modulate) each of the three voice channels to a different bandwidth, as shown in Figure 6.6. We use the 20- to 24-kHz bandwidth for the first channel, the 24- to 28-kHz bandwidth for the second channel, and the 28- to 32-kHz bandwidth for the third one. Then we combine them as shown in Figure 6.6.

6.12

Figure 6.6 *Example 6.1*



4kHz

ما رسمها هيك ؟

4kHz

لأنه ال Signal مش Periodic عشان نرسمها هيك هون ال Signal non-Periodic

6.13

Example 6.2

Five channels, each with a 100-kHz bandwidth, are to be multiplexed together. What is the minimum bandwidth of the link if there is a need for a guard band of 10 kHz between the channels to prevent interference?

Solution

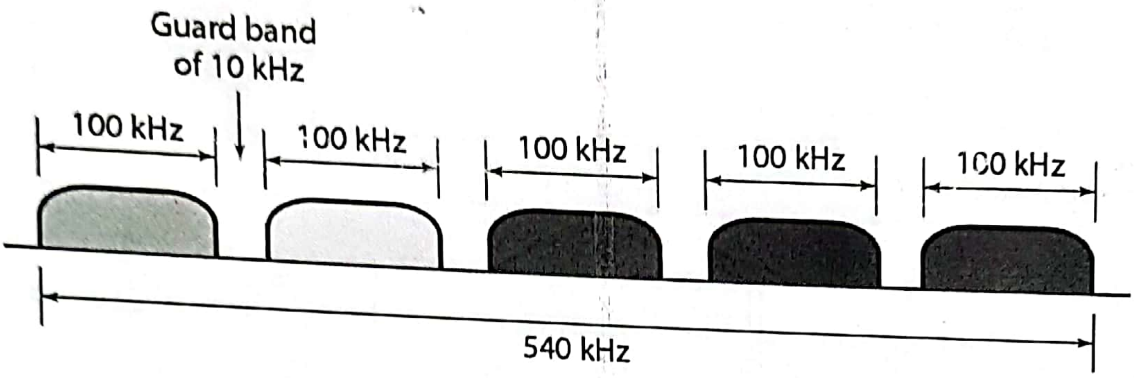
For five channels, we need at least four guard bands. This means that the required bandwidth is at least

$$5 \times 100 + 4 \times 10 = 540 \text{ kHz},$$

as shown in Figure 6.7.

6.14

Figure 6.7 *Example 6.2*



6.15

Example 6.3

Data Rate

Four data channels (digital), each transmitting at 1 Mbps, use a satellite channel of 1 MHz. Design an appropriate configuration, using FDM.

↳ Bandwidth of the channel

Solution

The satellite channel is analog.

We divide it into four channels, each channel having a 250-kHz bandwidth.

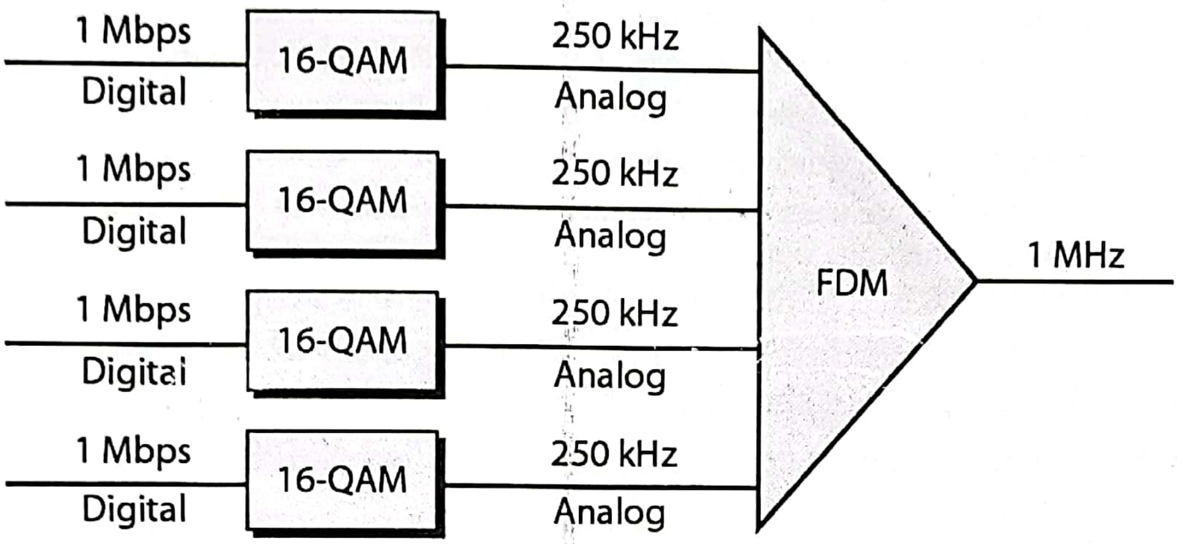
Each digital channel of 1 Mbps is modulated such that each 4 bits are modulated to 1 Hz.

One solution is to use 16-QAM modulation.

Figure 6.8 shows one possible configuration.

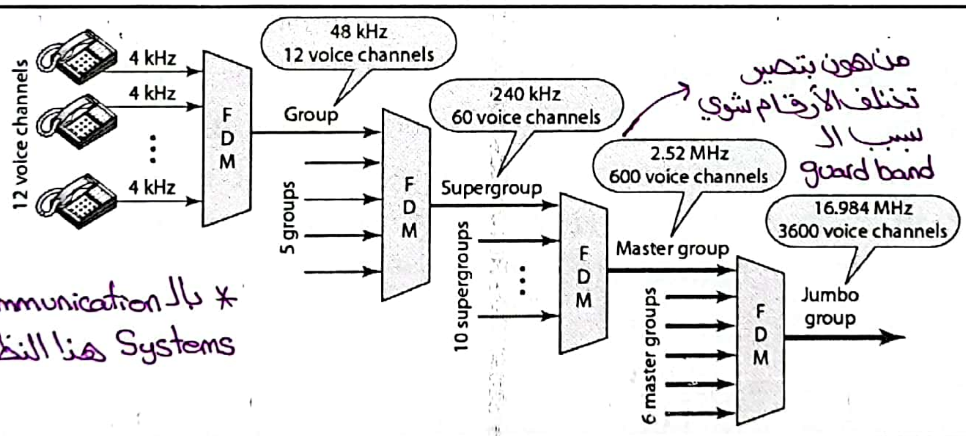
6.16

Figure 6.8 *Example 6.3*



* النظام المستخدم في الأردن هو الـ Standard الأوروبي.

Figure 6.9 *Analog hierarchy* → حاجتك للـ bandwidth بتزيد



من هون بتصير تختلف الأرقام شوي بسبب الـ guard band

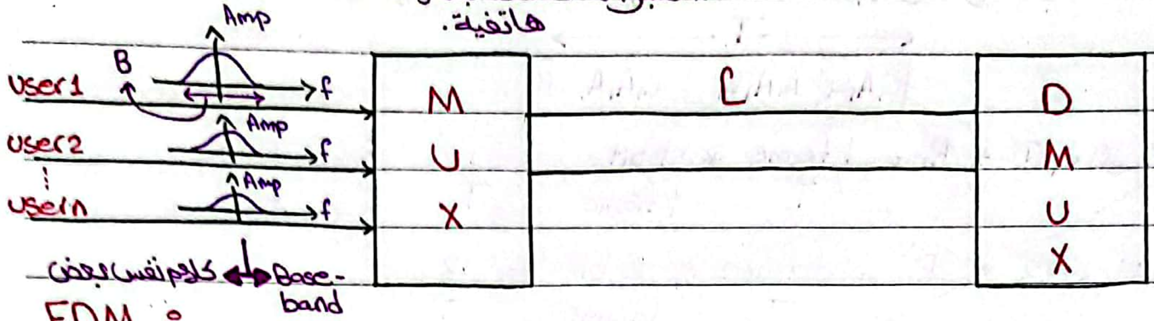
* بالـ Telecommunication Systems هذا النظام المستخدم.

- * Used by telcos to optimize the efficiency of the infrastructure
- * Combines multiple low bandwidth lines into higher bandwidth
- * Fewer but larger lines are used: → عدد أقل بس bandwidth عالي
  - ✓ Advantages: optimized installation cost and easier long haul management
  - ✓ Disadvantages: less diversity and more down lines in case of a failure

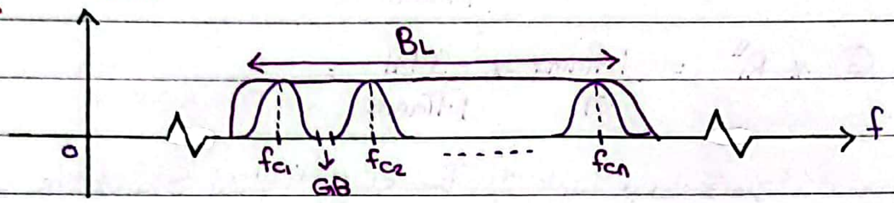
← لو واحد خرب كلهم بخربوا.

1/12/2022 chapter 6

*Slide 6.7 : * نعتبر انه ال users بيعملوا مكالمات هاتفية.



FDM :

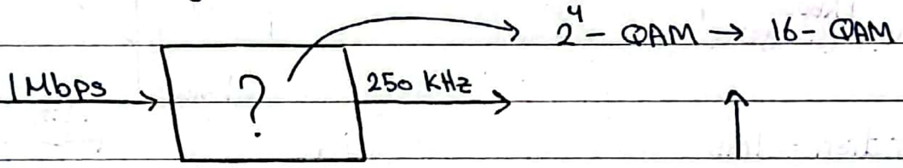


* Part of the bandwidth مخصص الك all the time لـ بتزيد مع ازدياد ال users.

* ال Time-Division Multiplexing : you have all the bandwidth at part of time.

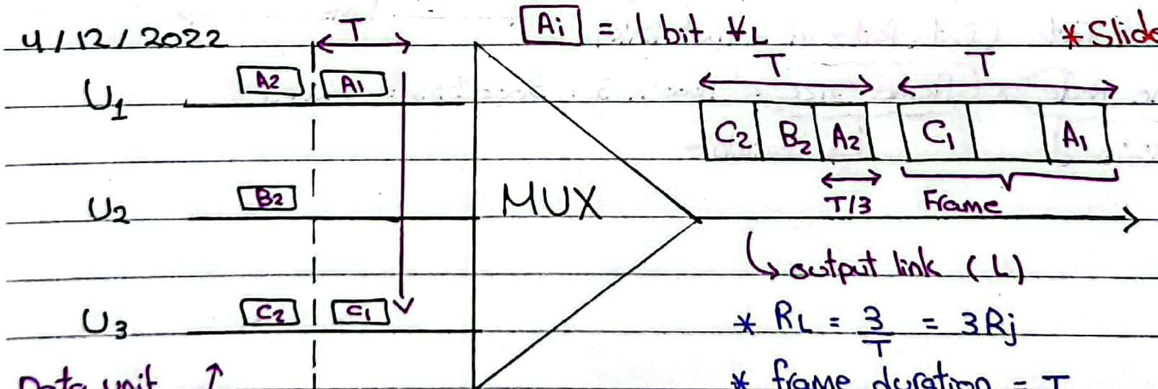
*Slide 6.16 : Example 6.3 :

Given that we are using QAM:



* N = 1 Mbps * S = 250 kHz * $\frac{N}{S} = r \rightarrow r = \frac{1\text{ Mbps}}{250\text{ kHz}} = 4$ bits.

4/12/2022 $\boxed{A_i}$ = 1 bit $\forall L$ *Slide 6.26



* $R_L = \frac{3}{T} = 3R_j$

* frame duration = T

* $R_1 = \frac{1\text{ bit}}{T} = \frac{1}{T}$ bps $= R_2 = R_3$ *Data unit duration:

* let R_j be the data rate of the j^{th} user.
→ in input link = T
→ in output link = T/3.

* انت ك user ممنوع تاخذ حصة غيرك من ال users إلا لو اشتريت ال Time slot تاع غيرك :

$$\xleftrightarrow{\quad T \quad}$$

$C_1A_1C_1$	$A_1A_1B_1$	$A_1A_1A_1$

① $* \; R_1 = \dfrac{1 \text{ frame}}{T} * \dfrac{1 \text{ bit}}{1 \text{ frame}} = \dfrac{1}{T}$ ← اللون الزودي

② $* \; R_1' = \dfrac{1 \text{ frame}}{T} * \dfrac{2 \text{ bit}}{1 \text{ frame}} = \dfrac{2}{T}$ ← باللون الأزرق

③ $* \; R_1'' = \dfrac{1 \text{ frame}}{T} * \dfrac{3 \text{ bit}}{1 \text{ frame}} = \dfrac{3}{T}$ ← باللون الزهري

* في Standards لشراء ال Time slot بال link بندفع قيمتها كل شهر مثلاً ، ممكن تشتري أكثر من Time slot ومو شرط يكونوا جمب بعض . (ال Standard هنا اوروبي مش امريكي)

← بالعادة بيستخدم مع الـ Analog بس ممكن بيستخدم مع الـ Digital لأنه مرنا نقدر نحولها للـ Analog

Other Applications of FDM

- * AM and FM radio broadcasting:
  - Air is the transmission medium
  - A limited bandwidth is assigned for each modulation technique
  - Each station shifts its signal into a predefined band
  - Guard bands are automatically predefined
  - All signals are transmitted over the air simultaneously as if they are a single composite signal
  - The radio receiver filters out the composite signal to a single desired modulated signal, which is demodulated
- * TV broadcasting: similar to AM and FM in concept
- * First generation of cellular telephones (called AMPS):
  - ✓ 3 kHz voice signal is FM modulated
  - ✓ Two 30 kHz channels are assigned for each user to receive and transmit
  - ✓ Carrier frequencies are dynamically assigned/reassigned based on usage

6.19

*range of AM: 530 → 1700 KHz

* شوفي مثال الكتاب عالريموت (مثال عال FDM).
لو كانوا عشان يفحصوا لو الريموت شغال
أو لا كانوا يسلطوه عالراديو ويكبسوا أي كبسة
فلو طلع صوت الكبسة بالراديو بكون الريموت شغال.

Example 6.4

The Advanced Mobile Phone System (AMPS) uses two bands. The first band of 824 to 849 MHz is used for sending, and 869 to 894 MHz is used for receiving. Each user has a bandwidth of 30 kHz in each direction. How many people can use their cellular phones simultaneously?

Solution ↳ " كله بيه يكون فل "

* الـ Users هون random and Independent.

Each band is 25 MHz (849 – 824 = 25 & 894 – 869 =25)
If we divide 25 MHz by 30 kHz, we get 833.33 → worst case
In reality, the band is divided into 832 channels
Of these, 42 channels are used for control, which means only 790 channels are available for cellular phone users

Independent and random ↵

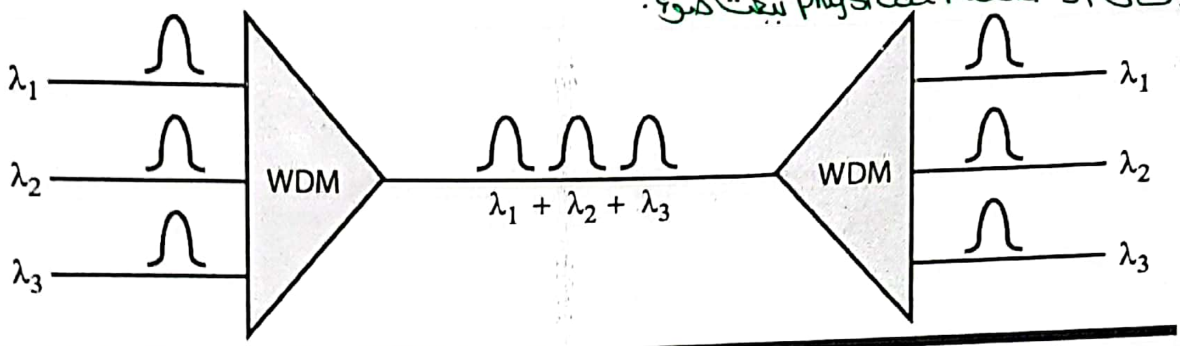
6.20

الترددعالي جداً ←

← يعتمد خصائص الضوء.　　　* الضوء هو electromagnetic wave

Figure 6.10 ***Wavelength-division multiplexing (WDM)*** → مبني على أطوال موجية

لو كان ال Physical Media بيبعث ضوء.



- * Designed for the high-bandwidth fiber-optic cable
- * Similar in concept to FDM except that it is used to mux and demux optical signals transmitted over fiber-optic channels
- * Optical signals are very high frequency signals that are usually defined by their wavelengths

مستخدم فيهم

المنشور (Prism) هو الي بنتعامل معه بالضوء.

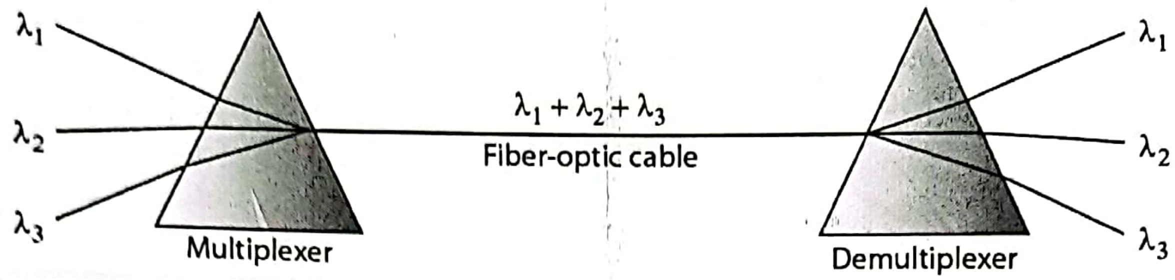
6.21

Note

WDM is an analog multiplexing technique to combine optical signals.

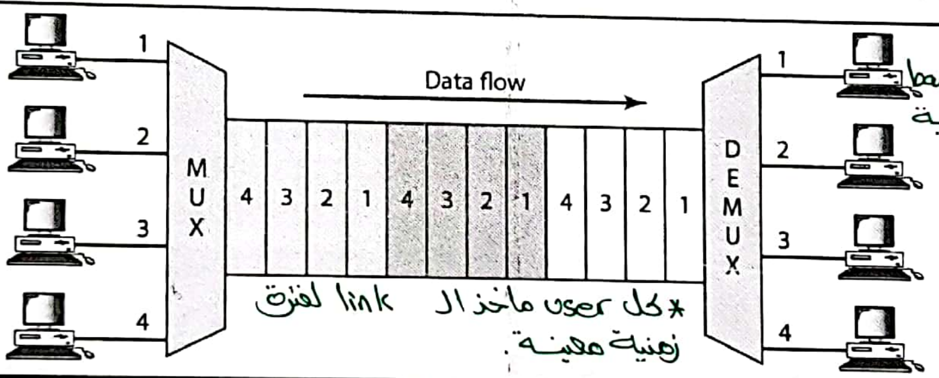
6.22

Figure 6.11 *Prisms in wavelength-division multiplexing and demultiplexing*



- Multiple narrow band light signals are combined into a single wide band light signal using a prism
- The prism bends a beam of light based on the signal frequency and the angle of incidence
- The prism is used both as mux and demux

Figure 6.12 Time Division Multiplexing (*TDM*) مستخدم بكثافة بال Telecommunication System



* بتعطيك كل ال bandwidth بال link لفترة زمنية معينة *

*كل user ماخذ ال link لفترة زمنية معينة.

- * TDM is a $\boxed{\text{digital process}}$ that allows several connections to time-share a high-bandwidth link
- * The whole bandwidth is used in full by each connection only during its time slots → حصة زمنية
- * The input devices are allowed to access the link based on either a synchronous (or sequential) or a statistical (or probabilistic) fashion ↓

عندك data أو ماعندك بكون الك Time slot بس لو ماعندك data فبضيع ال Time عالفاضي.

عندك data بيعطوك Time slot وبتبعت ماعندك ما بيعطوك Time slot (أفضل بالعادة بكون)

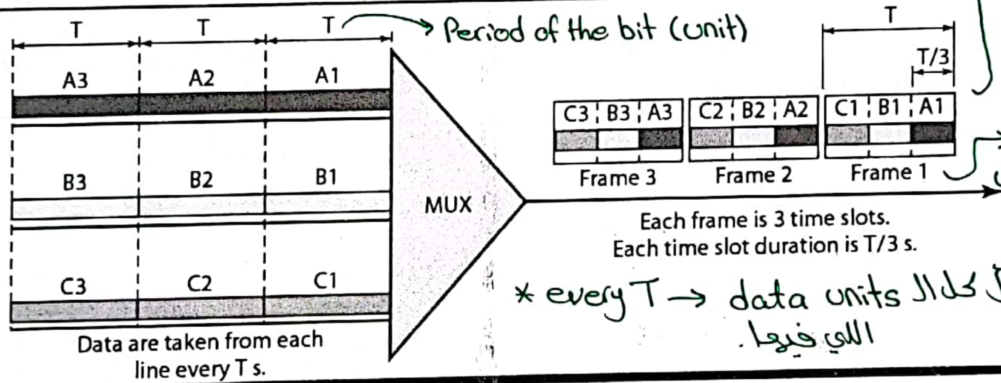
Note

TDM is a digital multiplexing technique for combining several low-rate channels into one high-rate one.

Time sharing على الـ based

6.25

Figure 6.13 *Synchronous time-division multiplexing*



Period of the bit (unit)

الـ Rate تبع user A, B, C متساوي.

* لو وحدة من الأجزاء تاعت الـ Fram ماكان الها data بضل الجزء من الـ Time محجوز الـها (الدفتر)

تحوي كل الـ data units تبعت الـ user

* every T → data units بنقل كل الـ اللي فيها.

* ■ The data flow of each connection is divided into units
* ■ Data units are combined into frames
* ■ The frame consists of, at least, one data unit from each connection
* ■ For *n* connections, the frame has at least *n* time slots
* ■ The data rate of the link is *n* times faster than each connection and hence the unit duration is *n* times shorter

* Frame duration تلقائيا تساوي الـ data unit duration on Input link.

6.26

لـ * العلاقة بين الـ time والـ rate علاقة عكسية *

* أحسن الـ Performance الـ TDM لما يكون عندك heavy load مش light load.

لازم ال input
وال output
يكونوا متناسقين

لأنه ال Time slot بضل موجود
لل user حتى لو ما في
Data

Note

In synchronous TDM, the data rate of the link is *n* times faster, and the unit duration is *n* times shorter.

Example 6.5

In Figure 6.13, the data rate for each input connection is ~~3~~ 1 kbps. If 1 bit at a time is multiplexed (a unit is 1 bit), what is the duration of (a) each input slot, (b) each output slot, and (c) each frame?

Solution

We can answer the questions as follows:

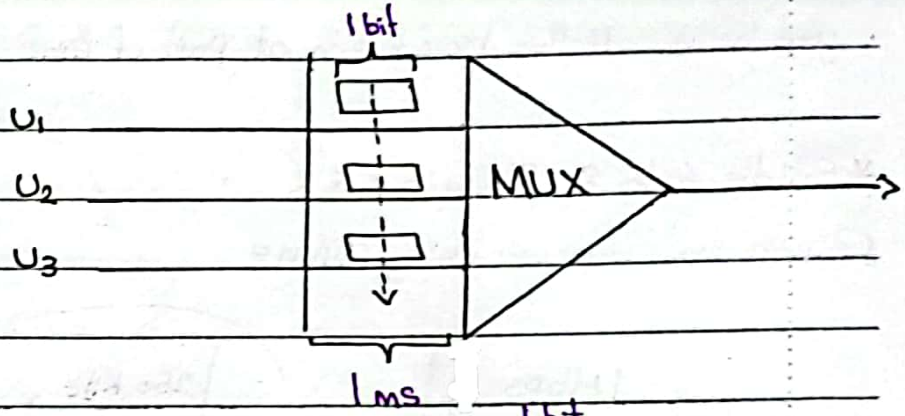
a. The data rate of each input connection is 1 kbps. This means that the bit duration is 1/1000 s or 1 ms. The duration of the input time slot is 1 ms (same as bit duration).

b. The duration of each output time slot is one-third of the input time slot. This means that the duration of the output time slot is 1/3 ms.

c. Each frame carries three output time slots. So the duration of a frame is 3 × 1/3 ms, or 1 ms. The duration of a frame is the same as the duration of an input unit.

8/12/2022

Example 6.5, Slide 6-28 :



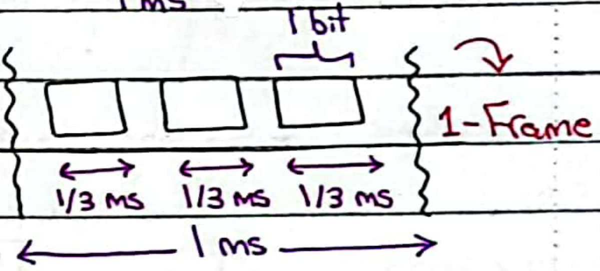
* Input data rate = 1 kbps

→ bit duration = $\frac{1}{1\ \text{kbps}}$ = 1 ms

أ خلال هاي ال 1ms بكون 3 بت :Frame

① Frame duration = 1ms

② Frame rate = $\frac{1\ \text{Frame}}{1\ \text{ms}}$ = 1000 frame/s



③ output data rate (Data Rate of output link)

= Frame Rate × Frame Size = 1000 × 3 = 3000 bps = 3 kbps

fixed دايمًا ↳ Variable

Example 6.6

Figure 6.14 shows synchronous TDM with a data stream for each input and one data stream for the output. The unit of data is 1 bit. Find (a) the input bit duration, (b) the output bit duration, (c) the output bit rate, and (d) the output frame rate.

Solution

We can answer the questions as follows:

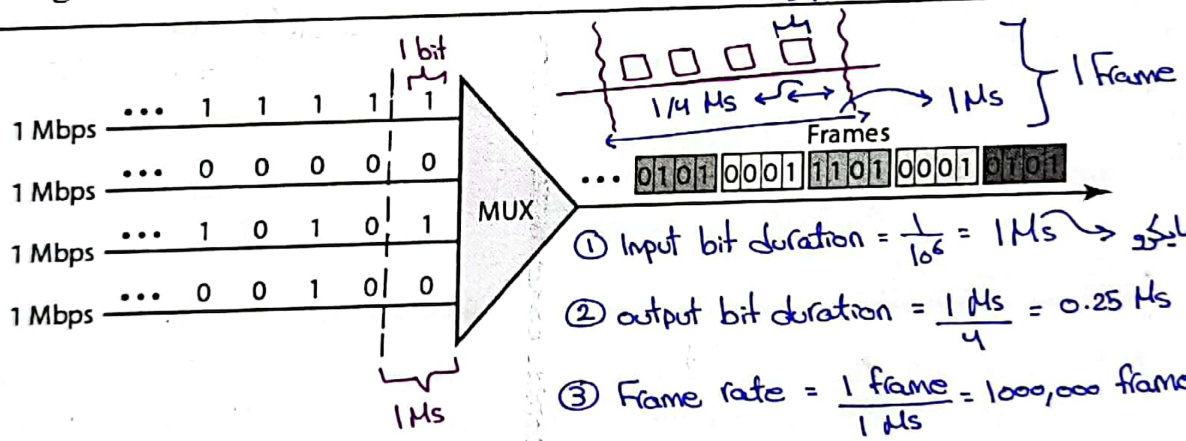
a. The input bit duration is the inverse of the bit rate: 1/1 Mbps = 1 μs.

b. The output bit duration is one-fourth of the input bit duration, or ¼ μs.

c. The output bit rate is the inverse of the output bit duration or 1/(¼ μs) or 4 Mbps. This can also be deduced from the fact that the output rate is 4 times as fast as any input rate; so the output rate = 4 × 1 Mbps = 4 Mbps.

d. The frame rate is always the same as any input rate. So the frame rate is 1,000,000 frames per second. Because we are sending 4 bits in each frame, we can verify the result of the previous question by multiplying the frame rate by the number of bits per frame.

Figure 6.14 *Example 6.6*



① Input bit duration = $\frac{1}{10^6}$ = 1 μs طايرة

② output bit duration = $\frac{1\ \mu s}{4}$ = 0.25 μs

③ Frame rate = $\frac{1\ \text{frame}}{1\ \mu s}$ = 1000,000 frame/s

④ output data rate = frame rate x frame size
= $10^6 \times 4$ = 4 Mbps
لـ ميجا

Example 6.7

Four 1-kbps connections are multiplexed together. A unit is 1 bit. Find (a) the duration of 1 bit before multiplexing, (b) the transmission rate of the link, (c) the duration of a time slot, and (d) the duration of a frame.

Solution

We can answer the questions as follows:

a. The duration of 1 bit before multiplexing is 1 / 1 kbps, or 0.001 s (1 ms).

b. The rate of the link is 4 times the rate of a connection, or 4 kbps.

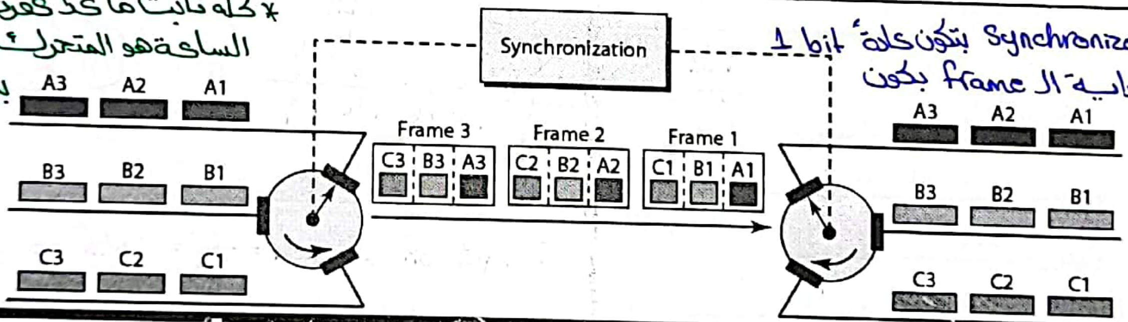
c. The duration of each time slot is one-fourth of the duration of each bit before multiplexing, or 1/4 ms or 250 μs. Note that we can also calculate this from the data rate of the link, 4 kbps. The bit duration is the inverse of the data rate, or 1/4 kbps or 250 μs.

d. The duration of a frame is always the same as the duration of a unit before multiplexing, or 1 ms. We can also calculate this in another way. Each frame in this case has four time slots. So the duration of a frame is 4 times 250 μs, or 1 ms.

كيف ابني أميز ال Frames اللي جاية من ال users عن بعض؟ ممكن يصير في لخبطة بين اللي انبعت واللي وصل فلتجنب هالاشي بضيف كانة Synchronize bit .

Figure 6.15 ***Interleaving***

* كله ثابت ما عدا عقرب الساعة هو المتحرك فلازم نسق بين ال input وال output لتكون القراءة صحيحة



* ال Synchronize bit بتكون عادةً 1 bit بنهاية أو بداية ال frame بكون متفق عليها وقيمتها

- TDM can visualized as two synchronized fast-rotating switches, one for muxing and the other for demuxing
- The process of inserting one unit of data per connection in each frame is called Interleaving and the data unit is called the interleaved unit

* بتحل ال Synchronize بس بتعمل مشكلة ال overhead.
data زيادة بنبعت ما كان لازم تنبعت فبالتالي بستهلك من ال bandwidth ال link .

Example 6.8

Four channels are multiplexed using TDM. If each channel sends 100 bytes/s and we multiplex 1 byte per channel, show the frame traveling on the link, the size of the frame, the duration of a frame, the frame rate, and the bit rate for the link.

Solution

- The multiplexer is shown in Figure 6.16.
- Each frame carries 1 byte from each channel;
- The size of each frame, therefore, is 4 bytes, or 32 bits.
- Because each channel is sending 100 bytes/s and a frame carries 1 byte from each channel, the frame rate must be 100 frames per second.
- The bit rate is 100 × 32, or 3200 bps.

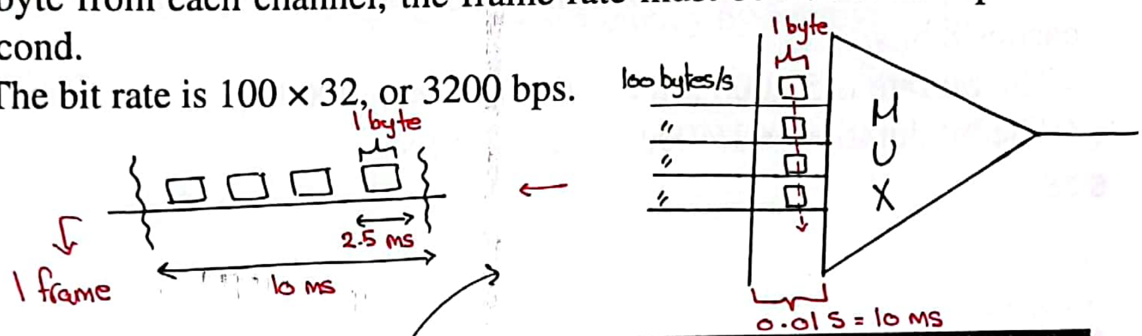
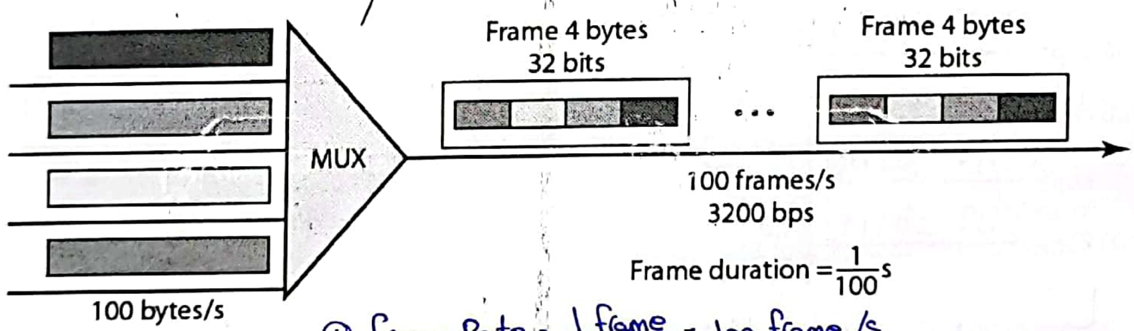


Figure 6.16 ***Example 6.8***



① frame Rate = $\frac{1 \text{ frame}}{10 \text{ ms}}$ = 100 frame /s

② frame size = 4 bytes = 4x8 = 32 bits

③ output data rate = frame rate x frame size = 100(32)
= 3200 bps

Example 6.9

A multiplexer combines four 100-kbps channels using a time slot of 2 bits. Show the output with four arbitrary inputs.

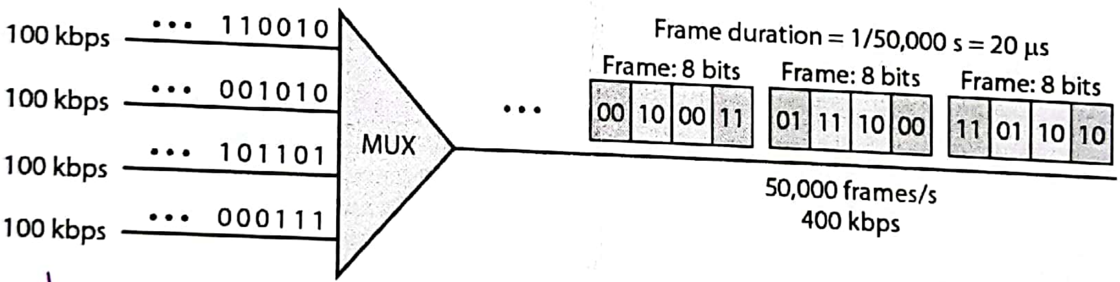
- What is the frame rate?
- What is the frame duration?
- What is the bit rate?
- What is the bit duration?

Solution

Figure 6.17 shows the output for four arbitrary inputs.

- The link carries 50,000 frames per second.
- The frame duration is therefore 1/50,000 s or 20 μs.
- The frame rate is 50,000 frames per second, and each frame carries 8 bits;
- The bit rate is 50,000 × 8 = 400,000 bits or 400 kbps.
- The bit duration is 1/400,000 s, or 2.5 μs.

Figure 6.17 *Example 6.9*



لو كانوا مختلفين بعمل
إضافة للأقل عشان يصيروا
نفس الاشي.

* Example 6.9, Slide 6.35 :

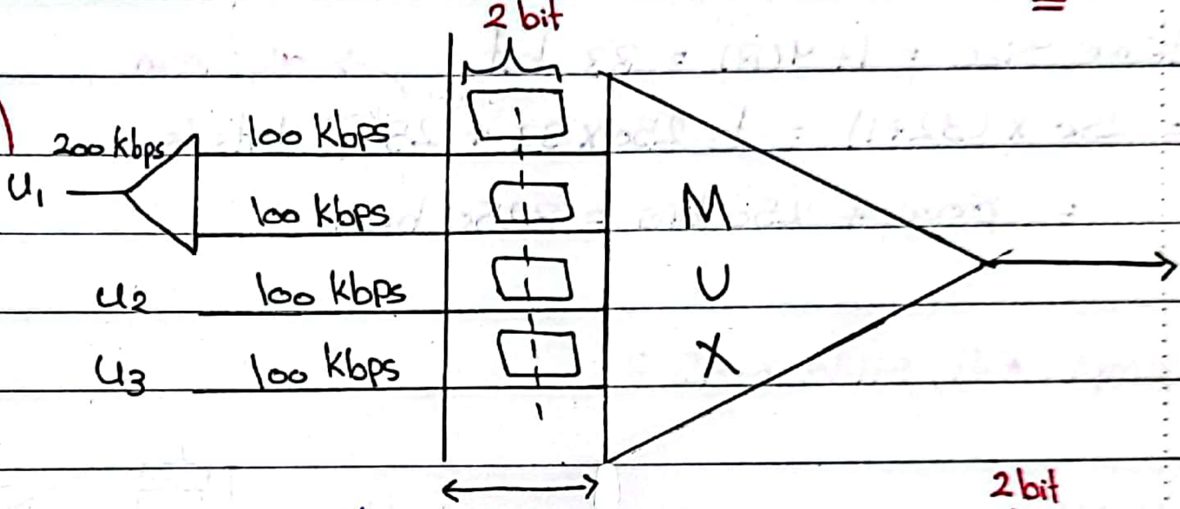
* $2x+x+x = 4x$

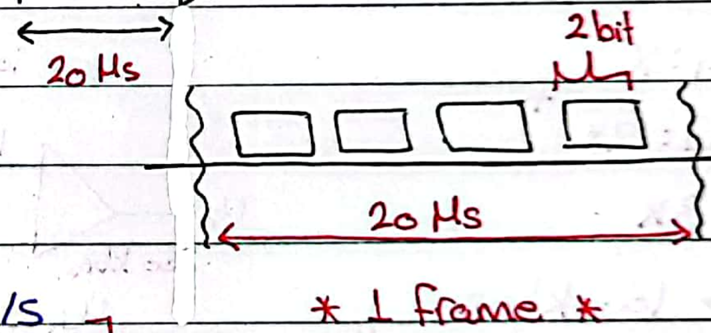
ممكن يكونوا هيك

أو 2 users عادي بدخل

الجلزي ماهو.

(Solutions :)





① bit duration $= \frac{1}{100 \times 10^3} = 0.01\,ms = 10\,\mu s$

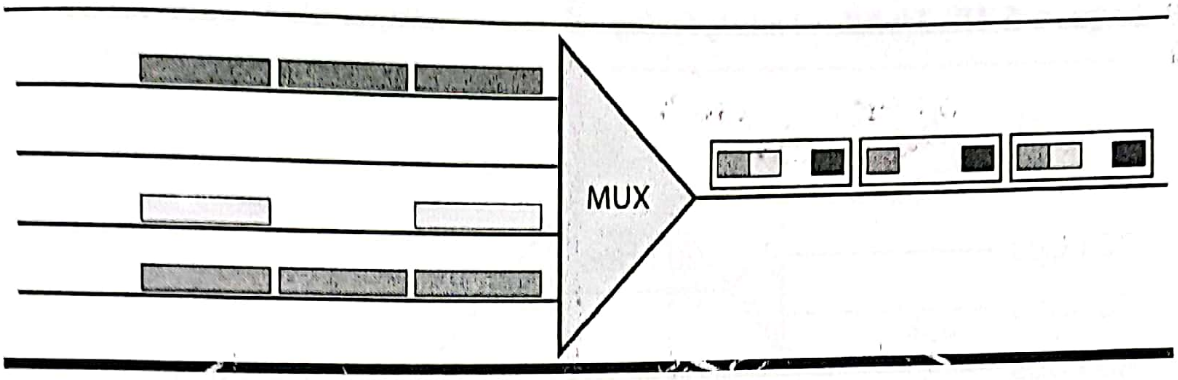
② Data unit duration $= 20\,\mu s$

③ frame rate $= \frac{1\ frame}{20\,\mu s} = 50{,}000$ frame/s → fixed

④ frame size $= 4(2) = 8$ bits

⑤ output Data Rate $= (50{,}000)(8) = 400{,}000$ bps

Figure 6.18 *Empty slots*



- Synchronous TDM may not be very efficient
- If no data is available at the input source, then the corresponding time slot in the output frame will be empty → wasting resources
- Statistical TDM can improve the efficiency by removing the empty slots

Non-Homogeneous TDM

هل ممكن الـ rates الي فايتة تكون مختلفة؟

- Can we multiplex devices that have different data rates?
- The answer is yes. How?
  - Faster devices may use more time slots per frame than slower devices

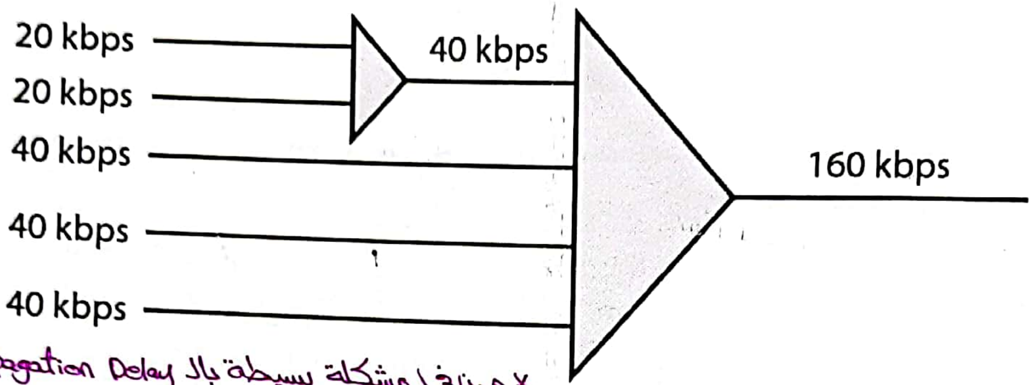
    بنعطيهم حصة أكبر من الـ Frame
  - But remember that in a TDM system:
    - The number of multiplexed devices, the number of time slots per frame, and the time slot duration in the frame are always fixed
  - Therefore, for TDM to work optimally, the different data rates must be integer multiples of each other. In this case, each device uses a number of time slots that corresponds to the ratio of its data rate to data rate of the slowest device

    بحل المسألة بدلالة الثوابت هدول.
  - For example: for three devices with data rates x, $2x$, and $6x$, the devices are assigned 1, 2, and 6 time slots per frame respectively.

    بنروح عأقل واحد بنعرفه X والباقي بكونوا حصص من هذا الـ X
  - What if data rates are not integer multiples of each others? Use Bit Padding (or Pulse Stuffing)

    لـ ليكون عدد صحيح
  - Bit Padding is a technique used by the multiplexer to force the data rates of the devices to be integer multiples of each other by adding extra dummy bits in the source streams of the slow device

Figure 6.19 *Multilevel multiplexing*



Propagation Delay هون في مشكلة بسيطة بال *
كثير عالية بتكون.

Figure 6.20 *Multiple-slot multiplexing*

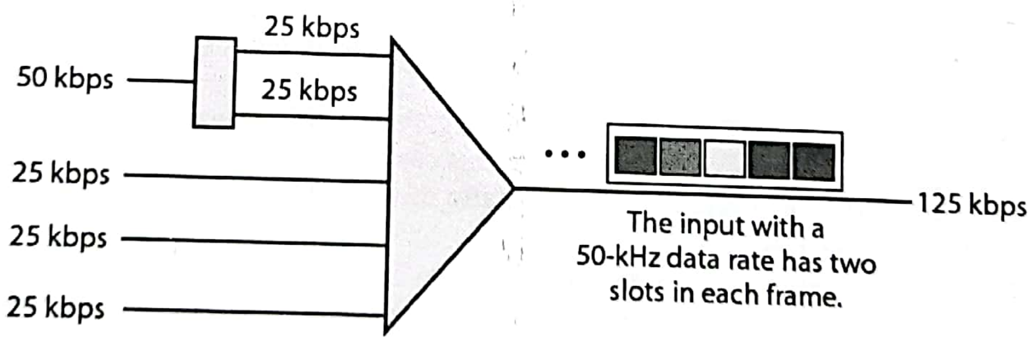
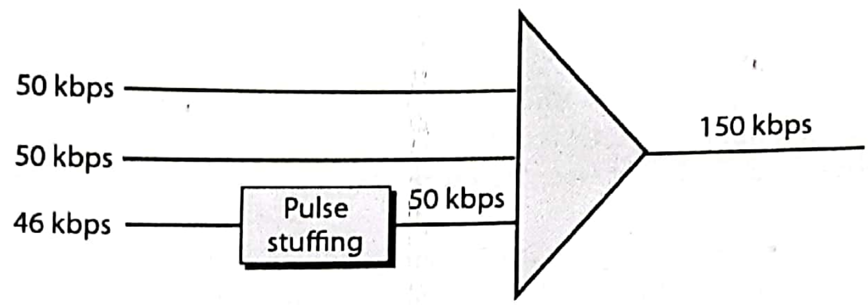
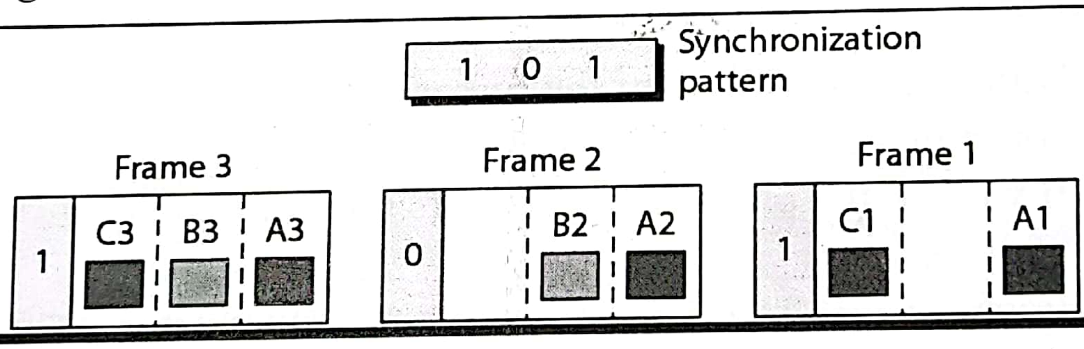


Figure 6.21 *Pulse stuffing*



6.41

Figure 6.22 *Framing bits* → Frame لبقدر يعرف وين بداية ونهاية الـ



- The major issue with TDM is the synchronization between the multiplexer and the demultiplexer
- The lack of synchronization may cause the data to be delivered to the wrong channel
- Therefore, one or more synchronization bits (called framing bits) are usually added to the beginning of each frame
- The framing bits usually follow a predefined pattern known by both sides such alternating 1's and 0's

6.42

Example 6.10

8 bit

We have four sources, each creating 250 characters per second. If the interleaved unit is a character and 1 synchronizing bit is added to each frame, find (a) the data rate of each source, (b) the duration of each character in each source, (c) the frame rate, (d) the duration of each frame, (e) the number of bits in each frame, and (f) the data rate of the link.

Solution

We can answer the questions as follows:

a. The data rate of each source is $250 \times 8 = 2000$ bps = 2 kbps.

b. Each source sends 250 ch/sec; therefore, the duration of a character is 1/250 s, or 4 ms.

c. Each frame has one character from each source, which means the link needs to send 250 frames/sec to keep the transmission rate of each source.

d. The duration of each frame is 1/250 s, or 4 ms. Note that the duration of each frame is the same as the duration of each character of each source.

e. Each frame carries 4 characters and 1 extra synchronizing bit. This means that each frame is $4 \times 8 + 1 = 33$ bits.

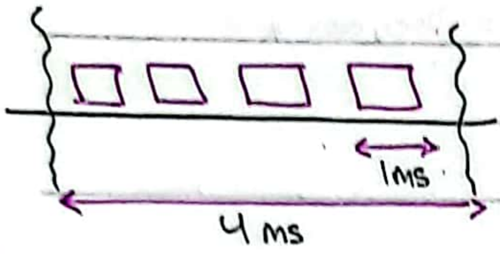
f. The link sends 250 frames per second and each frame contains 33 bits. This means that the data rate of the link is $250 \times 33 = 8250$ bps.

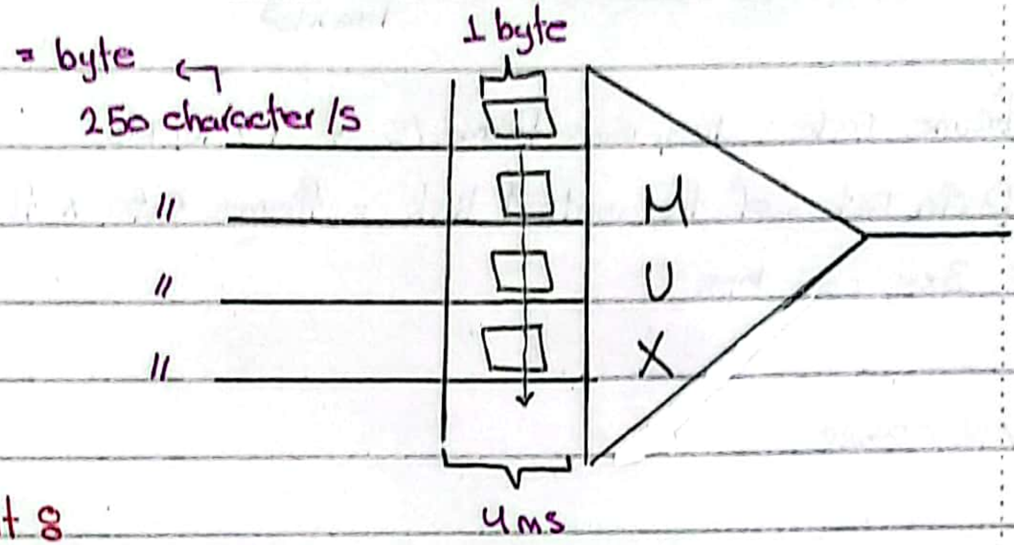
Example 6.10 (continued)

- Note that the bit rate of the link is greater than the combined bit rates of the four channels.
- If we add the bit rates of the fours channels, we get 8000 bps.
- Because 250 frames are traveling per second and each contains 1 extra bit for synchronization, we need to add 250 to the sum to get 8250 bps.

11 / 12 / 2022

* Example 6.10, Slide 6.43





A- without Synchronising bit 8

1- Input data rate = $250 \frac{bytes}{s} \times \frac{8\ bit}{byte} = 250 \times 8 = 2000$ bits/s

2- Duration of Data unit at the Input link = $\frac{1}{250}$ = 4 ms

number of users

Synch bit + $\hat{n}$ * (size of Data Unit)

frame size = 0 + 4 (8) = 32 bit

3- Frame duration = 4 ms

5- frame rate = $\frac{1\ frame}{4\ ms}$ = 250 frame/s

bps بتاعت الـ ↰ fixed ↰ Variable ↰

4- output Data rate = frame rate * frame size = 250 * 32 = 8000 bps

B- with Synchronising bit 8

* frame Size = 1+4(8) = 33 bit

$= 250 \times (32+1) = [\underbrace{250 \times 32 + 250}_{}]$ bits/s overhead

= 8000 + 250 bps = 8250 bps.

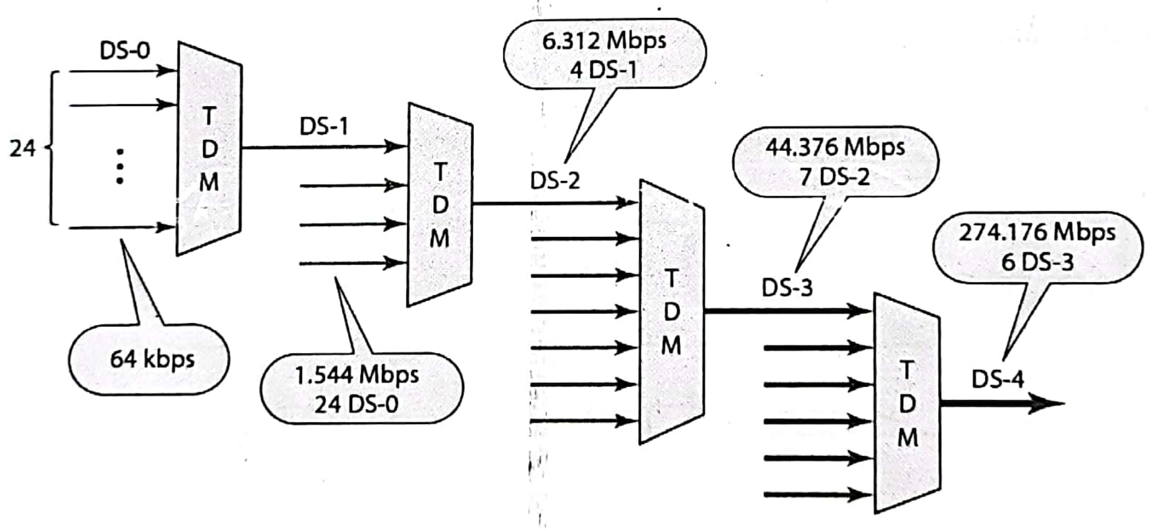
Example 6.11

Two channels, one with a bit rate of 100 kbps and another with a bit rate of 200 kbps, are to be multiplexed. How can this be achieved? What is the frame rate? What is the frame duration? What is the bit rate of the link?

Solution

- We can allocate one slot to the first channel and two slots to the second channel.
- Each frame carries 3 bits.
- The frame rate is 100,000 frames per second because it carries 1 bit from the first channel.
- The bit rate is 100,000 frames/s × 3 bits per frame, or 300 kbps.

Figure 6.23 ***Digital hierarchy***



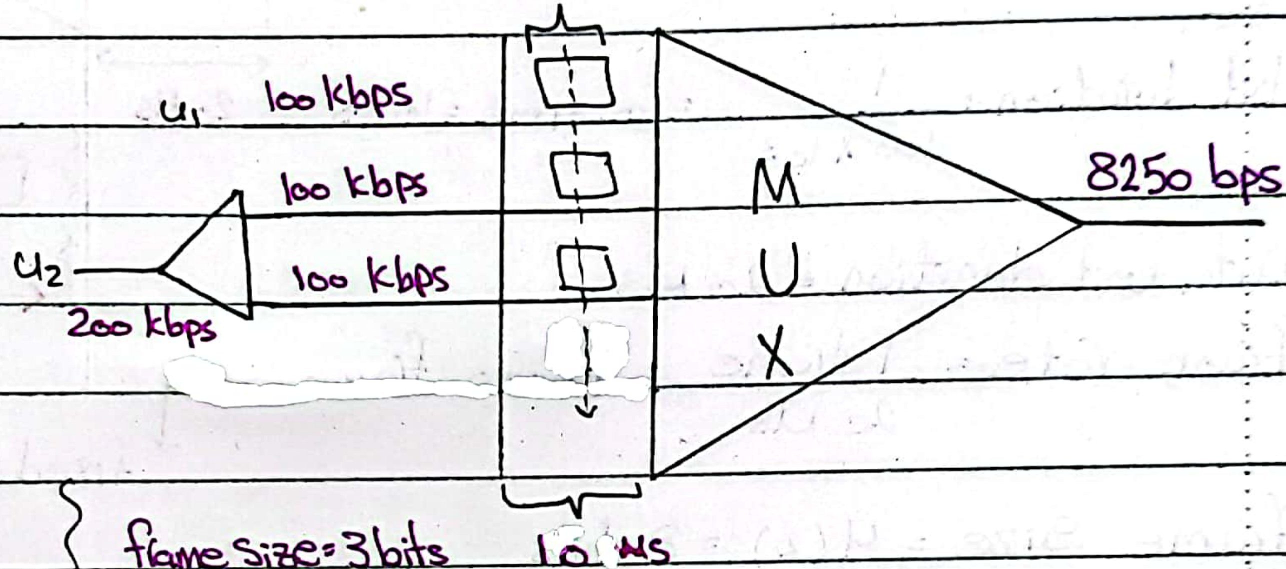
* Example 6.11, slide 6.45 8

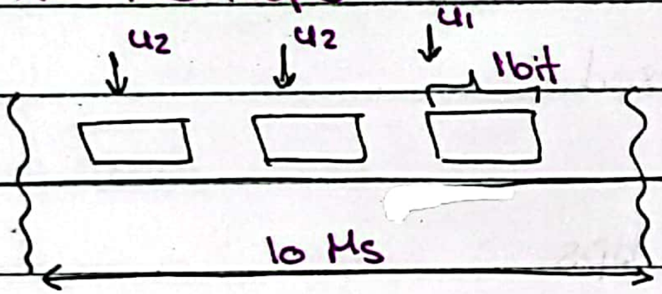
* $u_1 : X$

$+$ $u_2 : 2X$

$3X$

$\therefore X = 100$ kbps





1 - Data unit duration at Input $= \dfrac{1}{100 \times 10^3} = 0.01$ ms $= 10\ \mu s$

2 - frame rate = 100,000 frame/s

3 - Data rate of the output link = frame rate * frame size = 100,000 * 3

= 300,000 bps

13/12/2022

* أربعة أسئلة تخص تجربة الريموت الكنترول مع الراديو :

① مبني على أي Frequency بيندم ؟

AM Range : 530 kHz to 1700 kHz → (infrared) بعيد عن ال

② ال Signal اللي أنا بعتها اللي طلعت من ال remote قريبة ترددها ؟

38 KHz → لا تقع في range ال AM فكيف استقبلتها ؟

③ ال range اللي بيسمع فيه الانسان ؟

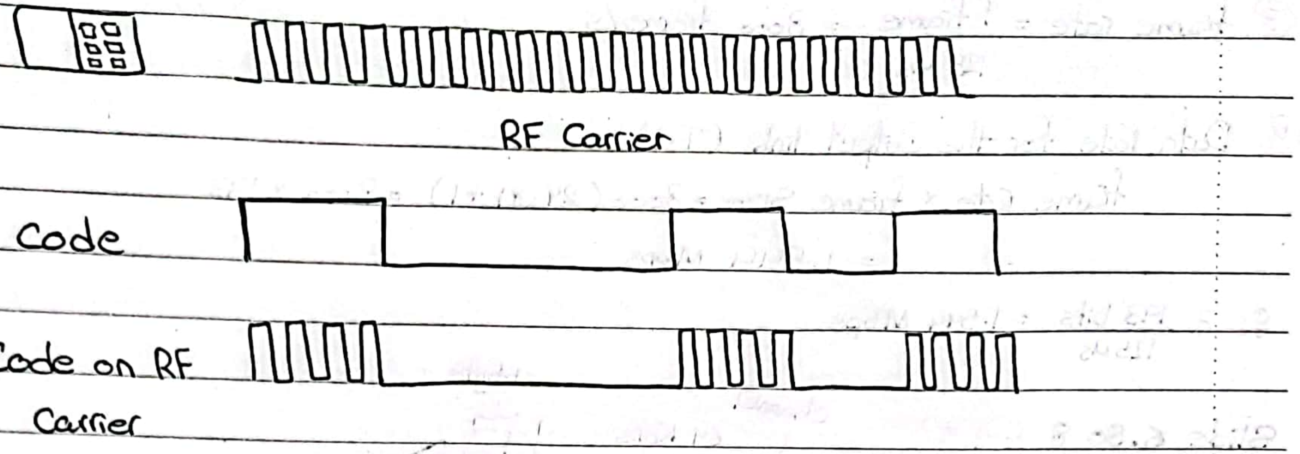
20 Hz → 20 KHz → ؟ 38 KHz وهي Signal فكيف سمعت ال

④ Infrared range (IR Range) ?

→

Five Apple

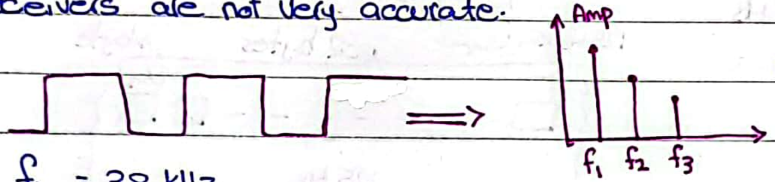
(إجابات الـ ٤ أسئلة)



* بعمل تردد معين داخل الـ Circuit من خلال الـ oscillator.

* IR Remote Control: داخلها resonator ← مسؤول عن التوليد.

- IR resonator 455 kHz
- Frequency divider: $\frac{455\ kHz}{12} = 37.9\ kHz \approx 38\ kHz$ → close enough, IR receivers are not very accurate.



* اللي استقبلناه هو الـ harmonic ←
f fondemantal frequency حيث الـ

- $f_1 = 38\ kHz$
- $f_2 = 2(38\ kHz) = 76\ kHz$
- $f_3 = 3(38\ kHz) = 114\ kHz$

⋮

- $f_{14} = 14(38\ kHz) = 532\ kHz$ ← "within AM range". → هنا اللي استقبلناه

* اللي سمعناه فعلاً هو الـ Information Signal (code) اللي محمّلة عالـ Carrier.

في نظام اوروبي وفي نظام امريكي.

Table 6.1 *DS and T line rates*

Service	Line	Rate (Mbps)	Voice Channels
DS-1	T-1	1.544	24
DS-2	T-2	6.312	96
DS-3	T-3	44.736	672
DS-4	T-4	274.176	4032

"مستخدم في الولايات المتحدة"

Figure 6.24 *T-1 line for multiplexing telephone lines*

.non periodic لا continuous *

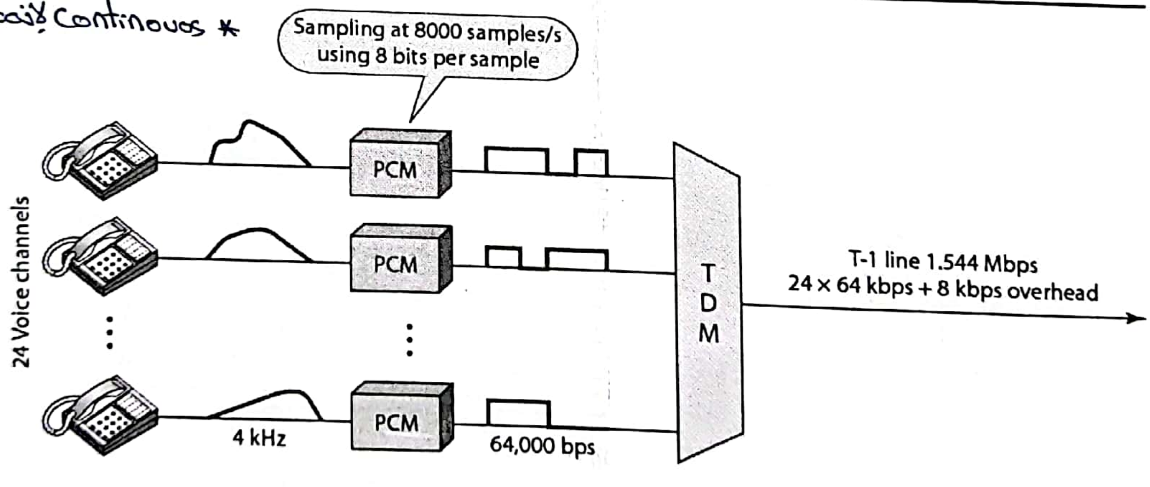
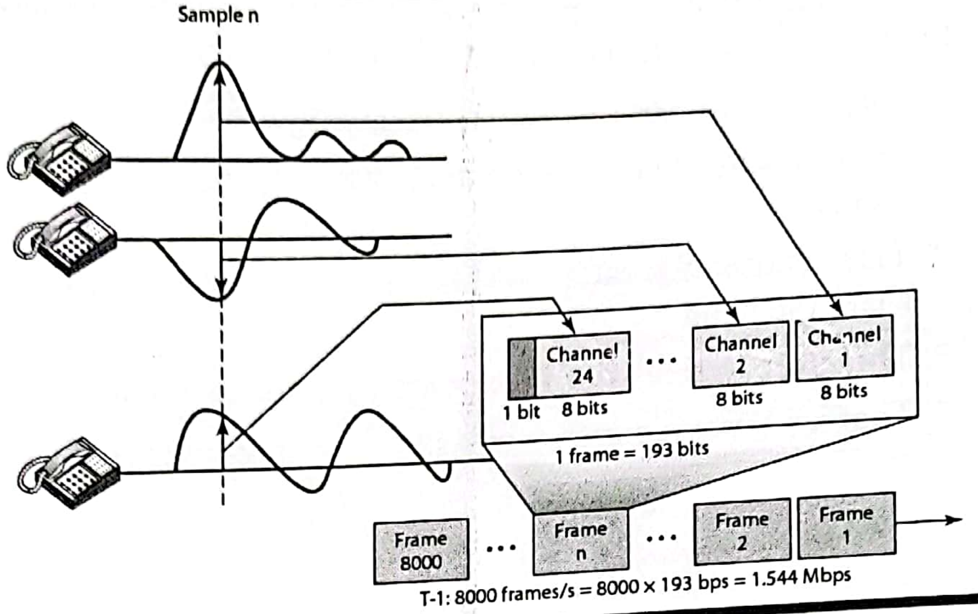


Figure 6.25 *T-1 frame structure*



6.49

"مستخدم بالأردن"

Table 6.2 *E line rates*

Line	*Rate (Mbps)*	*Voice Channels*
E-1	2.048	30
E-2	8.448	120
E-3	34.368	480
E-4	139.264	1920

فعليا هم 32 بس 2 منهم مستخدمين لأغراض ال Control

6.50

Slide 6.48 : T-1 :

① Input data rate per channel =

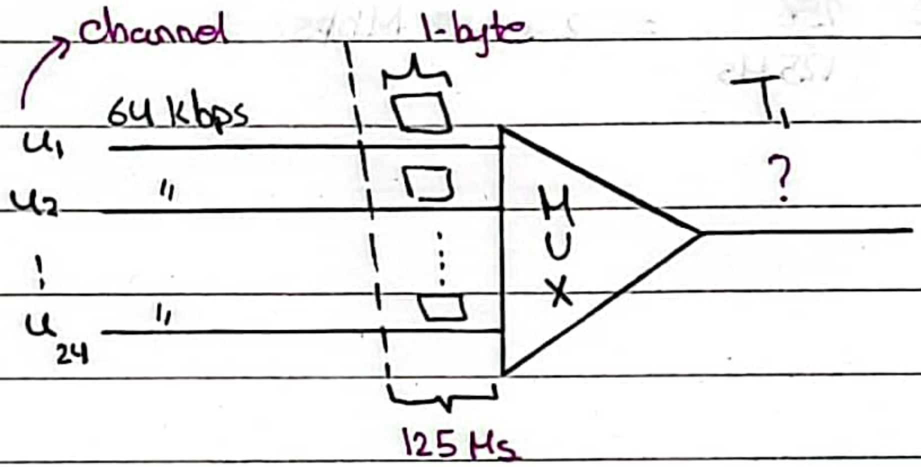
64 kbps

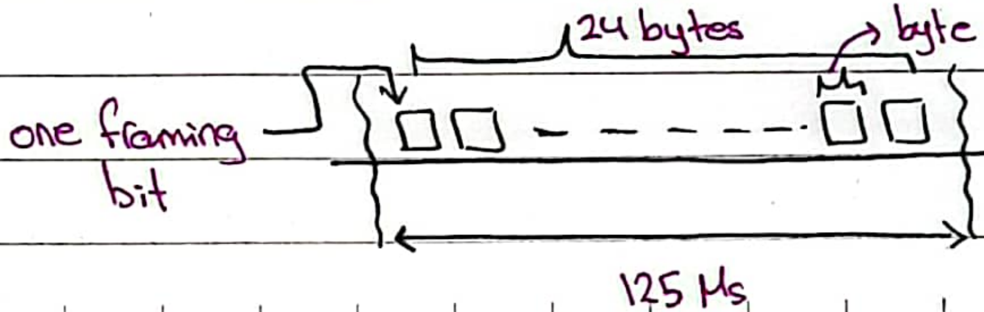
$4\,kHz * 2 * 8 = 64$ kbps

② Bit duration $= \dfrac{1}{64 \times 10^3} = 15.625\,\mu s$

→ Byte duration =

$8\,(15.625\,\mu s) = 125\,\mu s$





③ frame rate = $\frac{1 \text{ frame}}{125\ \mu s}$ = 8000 frame/s

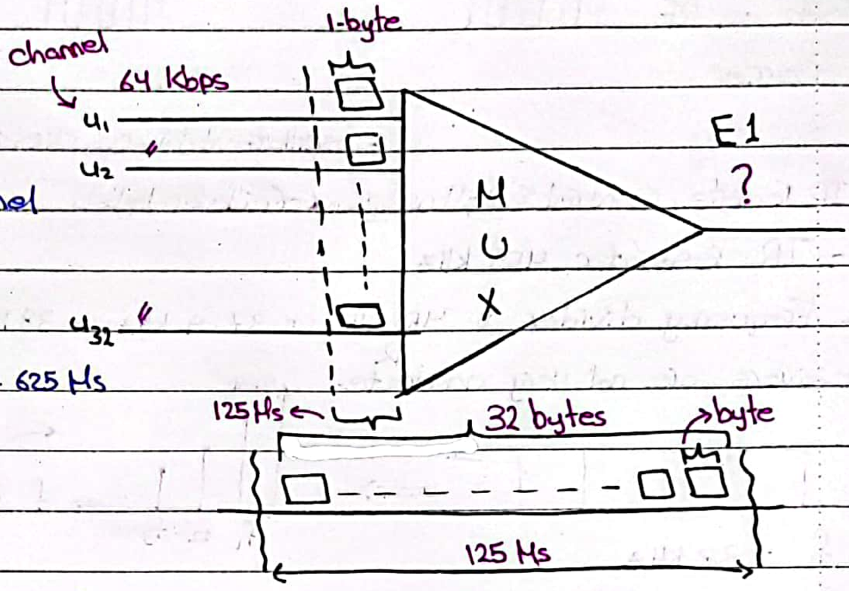
④ Data rate for the output link (T-1) =

frame rate x frame Size = 8000 (24 (8) + 1) = 8000 x 193

= 1.544 Mbps

OR : $\frac{193 \text{ bits}}{125\ \mu s}$ = 1.544 Mbps

Slide 6.50 : E-1 :



① Input data rate per channel

= 64 kbps

4 kHz * 2 * 8 = 64 kbps

② Bit duration = $\frac{1}{64 \times 10^3}$ = 15.625 µs

→ Byte duration =

8 (15.625 µs) = 125 µs

③ frame rate = $\frac{1 \text{ frame}}{125\ \mu s}$ = 8000 frame/s

④ Data rate for the output link (E-1) :

frame rate * frame Size = 8000 (32 (8) + 0) = 8000 X 256 = 2.048 Mbps

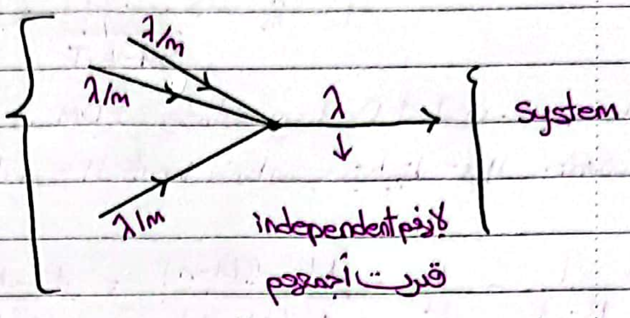
framing bit ولا

or : $\frac{256}{125\ \mu s}$ = 2.048 Mbps

* SM Vs FDM :

* m independent "Packet streams", each with an arrival rate of (λ/m) Pkts/s are transmitted over a communication link.

* Transmission time for each packet on average is $\frac{1}{\mu}$ (average pkt rate is μ)



* Link Rate (R) constant bps
* pkt Size (P_c) bits
* $R_c = P_c\left(\frac{1}{R}\right)$

*ملاحظة : كيف الـ transmittion rate يكون random
مع إنه الـ links اللي بنشوفها بحياتنا fixed؟

15/12/2022

* SM Vs FDM : Continue

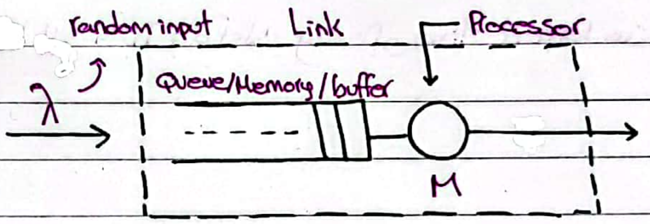
* λ, μ are average values

$R = 10$ Mbps or

100 Mbps

⋮

→ Constant Value (محددة مش random)



*E[·] : expected value or Mean or average

→ $E[T] \triangleq \bar{T}$ → average/mean

$T = Y \times \left(\frac{1}{R}\right)$

T → random

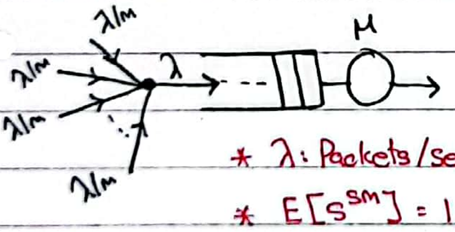
Y → Size of Packets (Random)

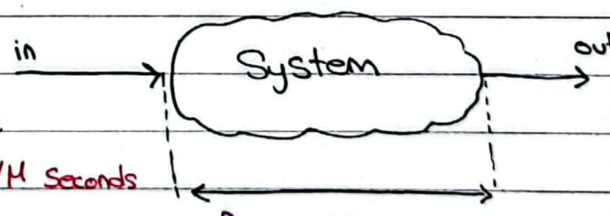
$\left(\frac{1}{R}\right)$ → constant (سرعة الـ link بتحدده)

→ time to transmitt one bit (bit duration)

* Y : Size of Packet in bits.

SM :



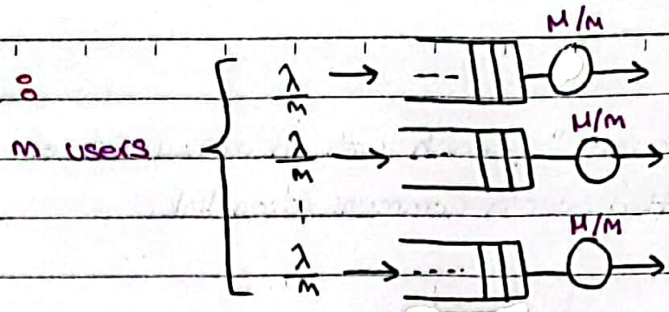


* λ: Packets/sec

* $E[S^{SM}] = 1/\mu$ Seconds

* $E[T^{SM}] = \frac{1}{\mu - \lambda}$ → response Time

FDM :



Response Time

$$E[T^{FDM}] = \frac{1}{\frac{\mu}{m} - \frac{\lambda}{m}}$$

$$= \frac{m}{\mu - \lambda} = m\,E[T^{SM}]$$

* اللي إله response time أعلى هو الـ FDM وكمان الـ Delay أعلى. بس فوائد الـ FDM إنه مافي randomness أو تداخل بين الـ inputs فمناسب بتطبيقات الـ audio والـ Video

$$* \quad \frac{1}{\mu-\lambda} - \frac{1}{\mu} = \frac{\mu - (\mu-\lambda)}{\mu(\mu-\lambda)} = \frac{\cancel{\mu} - \cancel{\mu} + \lambda}{\mu(\mu-\lambda)} = \frac{\lambda/\mu}{\mu-\lambda}$$

$$\rightarrow E[T^{SM}] = \frac{1}{\mu} + \frac{(\lambda/\mu)}{\mu-\lambda} \rightarrow E[T^{SM}] = E[S^{SM}] + E[T_a^{SM}]$$

$$= \frac{1}{\mu} + \frac{(\lambda/\mu)}{\mu-\lambda} = \frac{1}{\mu-\lambda}$$

أثبتنا إنه اللي طلعناه نفس اللي موجود بالسلايدة الإضافية اللي عالـ teams
(SM-vs-TDM-Example)

لـ برضه فيها الـ TDM إله أعلى Delay و response time من الـ SM

المشكلة فيه إنه لو ما عندك Data تظل الـ Time slot موجودة. اقتصاديًا ما في مشكلة بس من ناحية استغلالك للbandwidth في مشكلة، الحل لهذا الاشي بالـ Statistical. ← اللي ما عنده data ما بكون إله Time slot.

Synchronous TDM Application

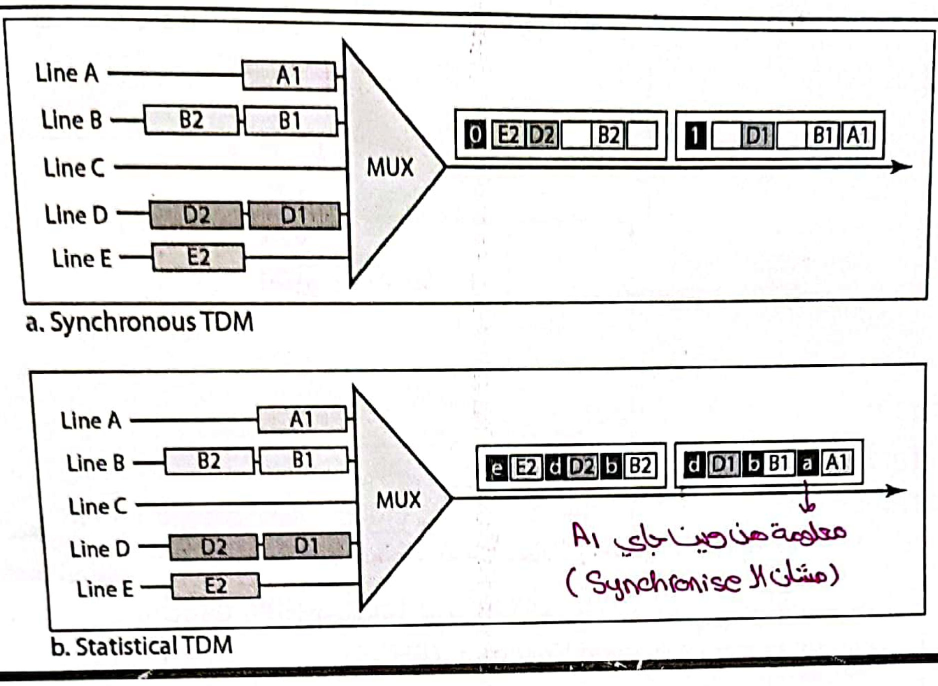
- Second-generation cellular phone technology called TDMA (Time-Division Multiple Access):
  - Digital version of the AMPS technology
  - FDM is still used to divide the bandwidth into 30 KHz bands
  - TDM is used for each FDM band so that 6 phone calls can share the band
  - Therefore, 6 time-slot frames are used for each band
  - That is, the capacity is increased by 6 times over AMPS

لازم نتأكد فيها إنه الـ Sender والـ reciever's فاهمين نفس الاشي لإنه بطل يطلع كل شي عالترتيب → "زيادة overhead"

Statistical Time-Division Multiplexing

- Synchronous TDM is inefficient when there are empty slots
- In statistical TDM:
  - The time-slots are dynamically allocated to improve bandwidth efficiency or utilization
  - The number of slots in each frame is less than the number of input lines
  - Addressing per slot is needed to specify the destination
  - The ratio of the data size to the address size must be reasonable
  - No frame synchronization bit would be needed
  - The bandwidth of the link is usually less than the sum of the bandwidth requirements of all input channels
  - The link bandwidth is usually sized based on the traffic intensity and statistics

Figure 6.26 *TDM slot comparison*



الـ Frame size بختلف بينهم

معلومة عن وين جاي A1 (مشان الـ Synchronise)

6.53

هون مش بنعمل Saving للـbandwidth هون بنزيد استخدامنا للـbandwidth وهذا في اللي متعارفين عليه

6-1 SPREAD SPECTRUM

In spread spectrum (SS), we combine signals from different sources to fit into a larger bandwidth, but our goals are to prevent eavesdropping and jamming. To achieve these goals, spread spectrum techniques add redundancy.

التنصت على الإشارة

ماتخلي المستقبل يوصله الـ Signal الأصلية لأنه يبعت تشويش

* بنزيد الـbandwidth مرات لأهداف مختلفة أهمها الـ Security ← بنضطر لذلك عشان مرات بنحتاج كثير للـSecurity مقابل زيادة الـ bandwidth.

→ التنصت
→ jamming

Topics discussed in this section:

* Frequency Hopping Spread Spectrum (FHSS)
* Direct Sequence Spread Spectrum (DSSS)

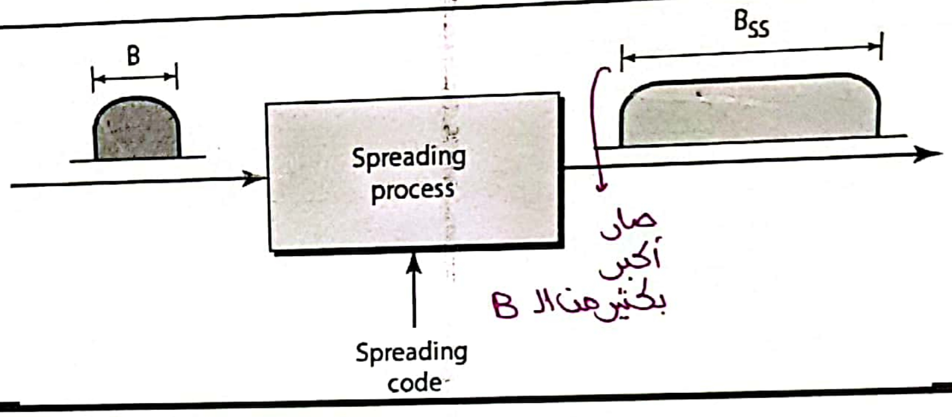
* المفروض أغير الـ frequency باستمرار لأمنع التنصت فلذلك لازم يكون عنا bandwidth أعلى.

* الـ Carrier هو النقطة الحساسة اللي بنشتغل عليها بهذا الموضوع.

6.54

* كل اللي بنعمله إنه نخلي العملية أصعب لأي مخرب بوصله يوصل الـ data ويخربها مع ذلك بظل في احتمالية إنه يوصلها.

Figure 6.27 *Spread spectrum*



صار أكبر بكثير من ال B

- ■ Principles of spread spectrum:
  1. The bandwidth allocated is much larger than needed allowing redundancy هي اللي بتستدعي انه نكبر ال bandwidth.
  2. The spreading process occurs after (i.e.; independent of) the signal creation (له مستقل دائما عن ال (الأصلية signal))
- ■ The two most well-known spreading techniques used are:
  1. Frequency Hopping Spread Spectrum (FHSS)
  2. Direct Sequence Spread Spectrum (DSSS)

6.55

Figure 6.28 *Frequency hopping spread spectrum (FHSS)*

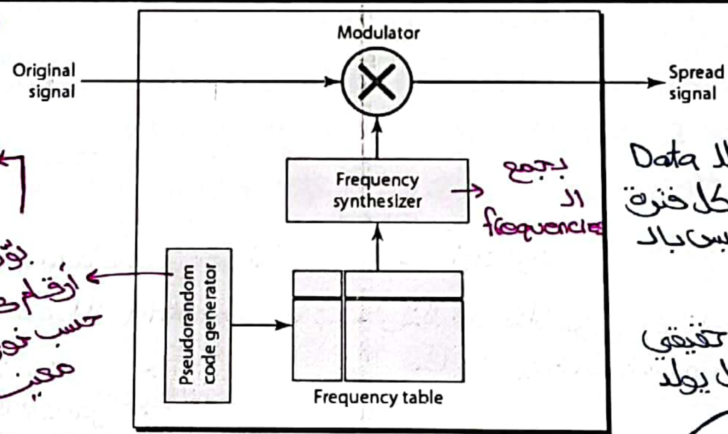
ميزتها لو كطيتها نفس ال int بتطلعلك دايما نفس التسلسل للأعداد العشوائية اللي طلعوا من أول مرة

لو بعطيها بذرة (seed) بتكون int بولد منها أرقام عشوائية.

Ex: seed(int)

5 7 2 1 عشوائية بين 0 و 10

بولد أرقام عشوائية حسب خوارزمية معينة



بجمع ال frequencies

* لو عرف المتنصت ال seed رح يقدر يوصل لل Data فالأفضل نغير ال seed كل فترة وفترة (بال real time صعبة بس بال theory ممكنة)

سمي Pseudo لأنه غير حقيقي (لو كانت نفس البذرة رح يولد نفس التسلسل موزي الحقيقي كل مرة تشكل)

- ■ M different carrier frequencies to modulate the signal
- ■ A pseudorandom code generator, also called pseudorandom noise (PN) creates a k-bit pattern for every hopping period T_h
- ■ A frequency synthesizer creates a carrier signal with the corresponding frequency according to a lookup table
- ■ Advantages: privacy and antijamming

6.56

18/12/2022

* من فوائد الModulation إنك بتقدر تعمل Multiplexing (تنقل الCarrier لأكثر من frequency)
، أيضاً إضافة الPower للCarrier ، وكمان طول الموجة بتناسب طردياً مع الdimension
Antenna بغض النظر شو نوعها فكل ما نزيد الfreq. بقل طول الموجة. فلو بدي استقبل إشارة ترددها عالي
جداً فحجم الAntenna بكون صغير، أما لو بدي ابعث اشارة بدون ما أعمل Modulation على الbase
band freq بتوقع إنه الfreq يكون كثير قليل فطول الموجة كثير كبير فلذلك فائدة الModulation
هو تصغير حجم الAntenna . ← لو بدي أولد different seq. of random num. بغير الSeed

نفس Tx لإنه نفس التسلسل والبذرة (5 أرقام بين 0 و 1)

* Slide 6.56 :

* $f_i \equiv f_c$

* من جهة الSender وجهة الreciever لازم يكونوا نفس الاشي.

*Pseudorandom :

Tx	Rx
* Seed (□, 5)	* Seed (□, 5)
0.2	0.2
0.4	0.4
0.1	0.1
0.3	0.3
0.0	0.0

Figure 6.29 *Frequency selection in FHSS*

* بنولد مجموعة أعداد عشوائية وبتصير تمشي عليهم وحدة وحدة ، لكل وحدة في فترة زمنية معينة (T_h) لحد ما نخلصهم كلهم وبنرجع نعيد من أول وجديد وهكذا.

* كل فترة تغير ال Pattern بكون أفضل وأمان أكثر
(Seed)

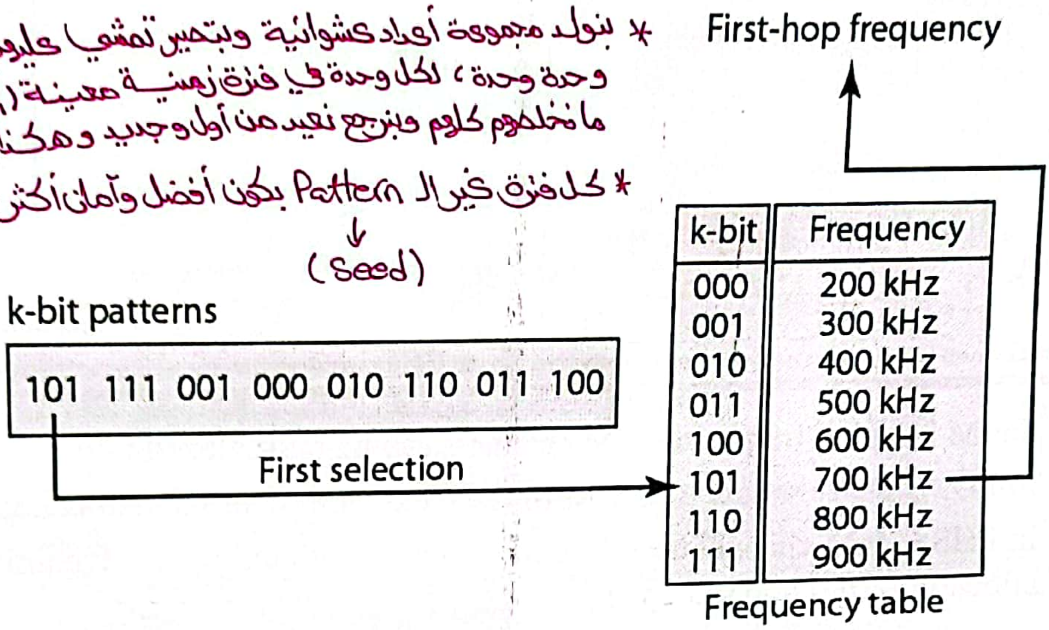
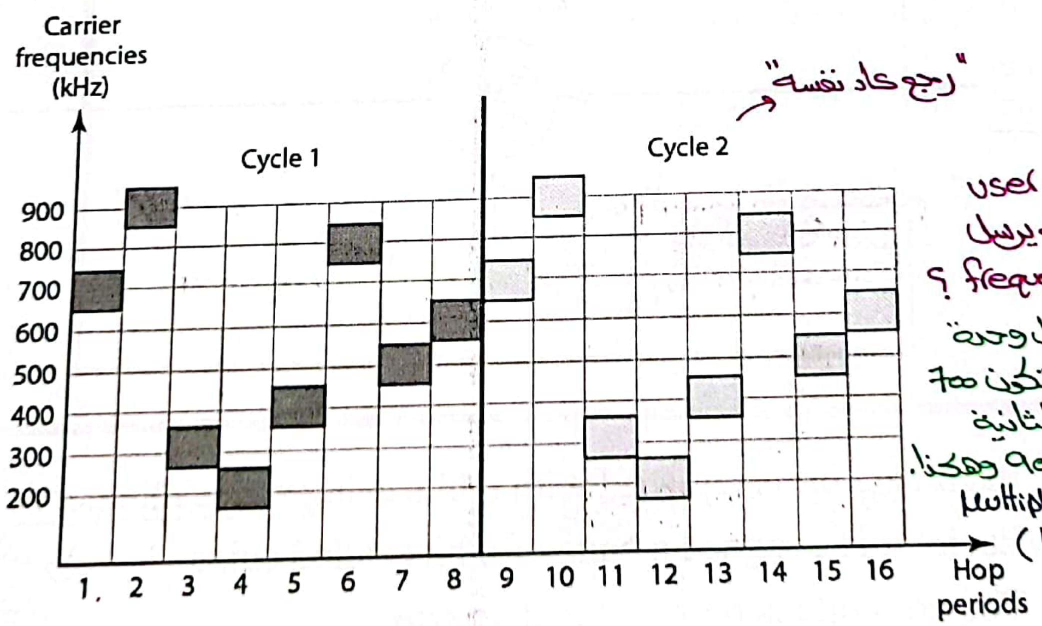


Figure 6.30 *FHSS cycles*

Bandwidth = 100 kHz

"رجع كاد نفسه"



* هلا بينفع أدخل user ثاني وأخليه يرسل بنفس ال frequencies ؟
بنقدر بس بأول وحدة مثلاً مابزبط تكون 700 الثانية و بالثانية مابزبط تكون 900 وهكذا.
(هنا اسمه multiplexing مع ال hopping)

هون بال hop period الوحدة بتقدر تستخدم 8 users.

*ما بصير في Collision بال FHSS

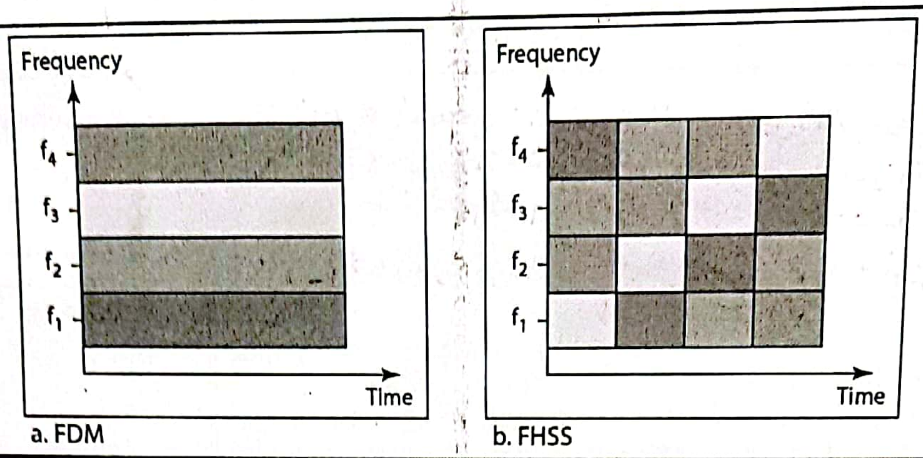


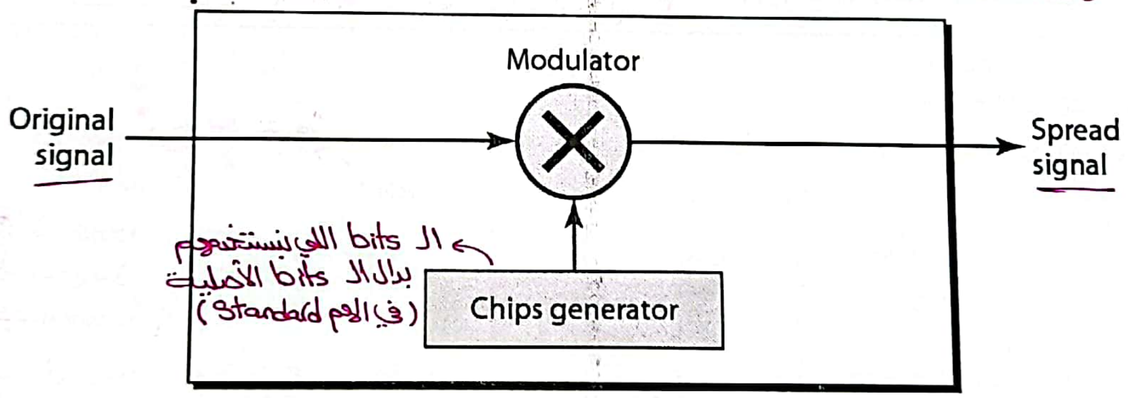
Figure 6.31 ***Bandwidth sharing***

- * For M hopping frequencies, M channels can be multiplexed into 1
- * In FDM, each channel uses 1/M of the bandwidth with fixed allocation
- * In FHSS, each channel uses 1/M of the bandwidth but with a dynamic allocation from hop to hop

* ال wifi بستخدم الـ frequency hopping *

Figure 6.32 ***Direct Sequence Spread Spectrum (DSSS)***

بنشيل ال bit الواحدة ← لمجموعة من ال bits عشان يصير في تمويه.



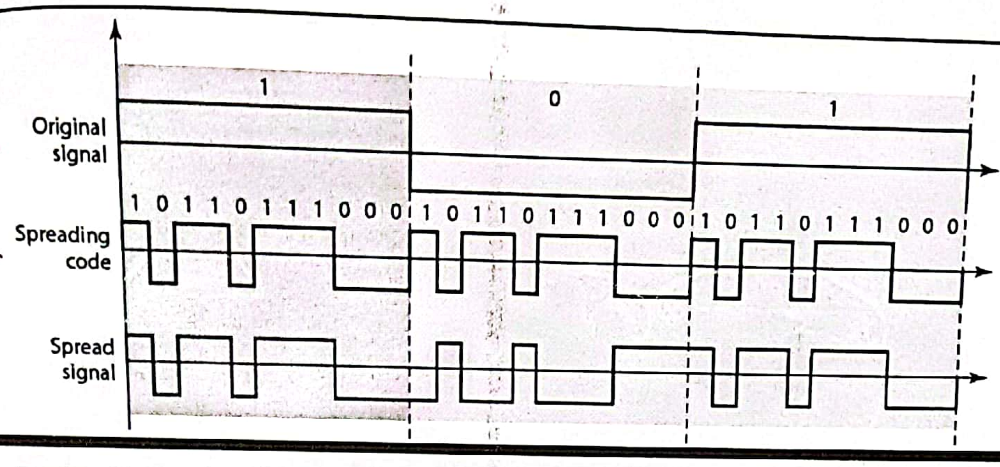
← ال bits اللي بنستخدمها بدال ال bits الأصلية (في الـ standard)

- Each data bit is replaced with n bits using a spreading code
- Each bit is assigned a code of n bits called chips
- The chip rate is n times the data rate

جدول للتبديل بنلتزم فيه

بزيد ال data rate وهذا اشي مش كثير محبب بس بالمقابل بنزيد الحماية لل data.

Figure 6.33 *DSSS example*



- Barker Sequence with n=11 is used in wireless LAN
- Assume that the original signal and the chips use polar NRZ encoding
- The rate of the spreading signal is 11 times that of the original signal. Therefore, the required bandwidth of the spreading signal is 11 times that of the original
- Advantages: privacy and immunity to interference
- Can we share the bandwidth in DSSS?
  - * Depends on the orthognality of the spreading code → يعتمد على نوع الـ code المستخدم.
  - * If different channel is assigned a different code from a set of orthogonal codes, then yes.

Data Communications and Networking Fourth Edition

Forouzan

Chapter 10

Error Detection and Correction

أسهل من ال Correction أكيد

بالإضافة لمعرفة حدوث الخطأ
بحاجة نعرف وين حدث هذا الخطأ ونصلحه
وهل حدث خطأ واحد ولا أكثر من خطأ ومين منهم اللي إله أكثر احتمال لتكرار حدوثه.

Note

Data can be corrupted during transmission.

مثل ال noise وال attenuation.
(بتصير بال wireless أكثر).

Some applications require that errors be detected and corrected.

وبعضها لو صار error ممكن ما يأثر مثل لو راح Pixel وحدة من الصورة أو الفيديو ما بأثر عكس لو ضاع اشي من File فبكون بأثر ال error.

10-1 INTRODUCTION

Let us first discuss some issues related, directly or indirectly, to error detection and correction.

* بنعتمد على ال data rate اللي بنشتغل عليها وعلى ال noise period of the.

* الشائع في ال data comm انه تضرب مجموعة bits مش وحدة، (burst loss)

Topics discussed in this section:

* Types of Errors
* Redundancy
* Detection Versus Correction
* Forward Error Correction Versus Retransmission
* Coding
* Modular Arithmetic

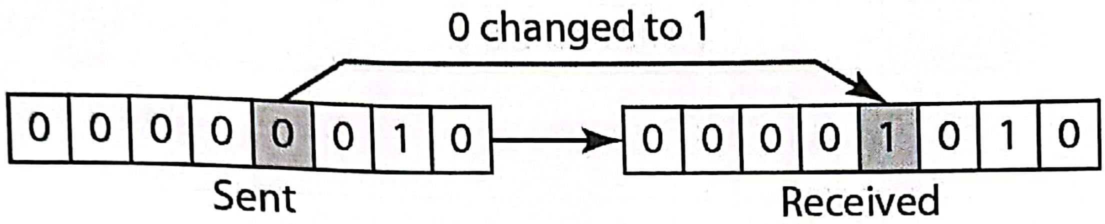
* Automated Repeat Request :
اعمل error detection واذا طلع عندك خطأ في ال Packet اطلب إعادة ارسالها.

تغيرت حالة bit واحدة فقط من 0 إلى 1 أو العكس → في أسباب كثير للتغير مثل ال noise أو ال attenuation

Note

In a single-bit error, only 1 bit in the data unit has changed.

Figure 10.1 *Single-bit error*



10.5

سواء كانوا جمب بعض
أو بعاد عن بعض كل شيء بينهم يكون loss
حتى لو كان اللي بينهم استلمناه صح.

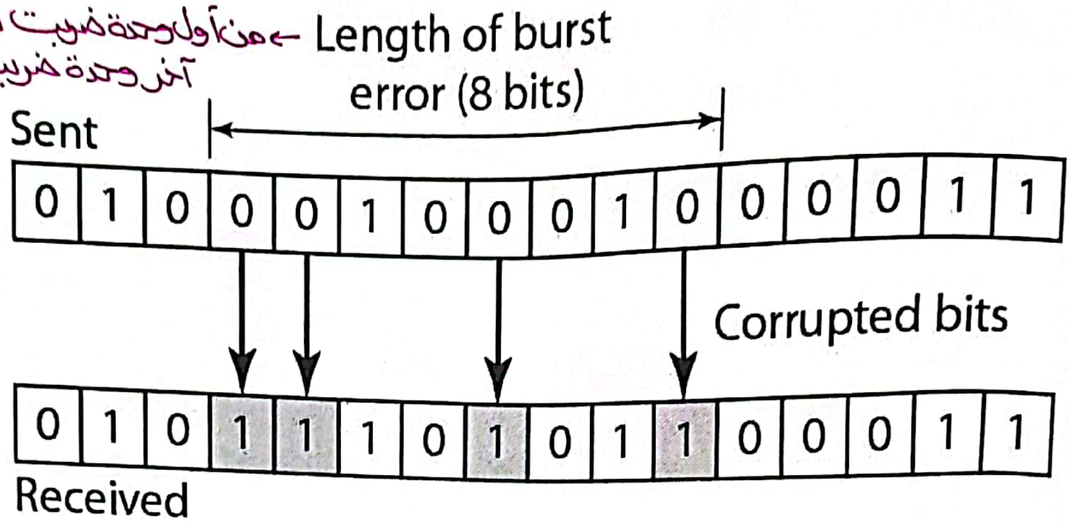
Note

A burst error means that 2 or more bits in the data unit have changed.

10.6

Figure 10.2 ***Burst error of length 8***

من أول وحدة ضربت لعند آخر وحدة ضربت



10.7

Note

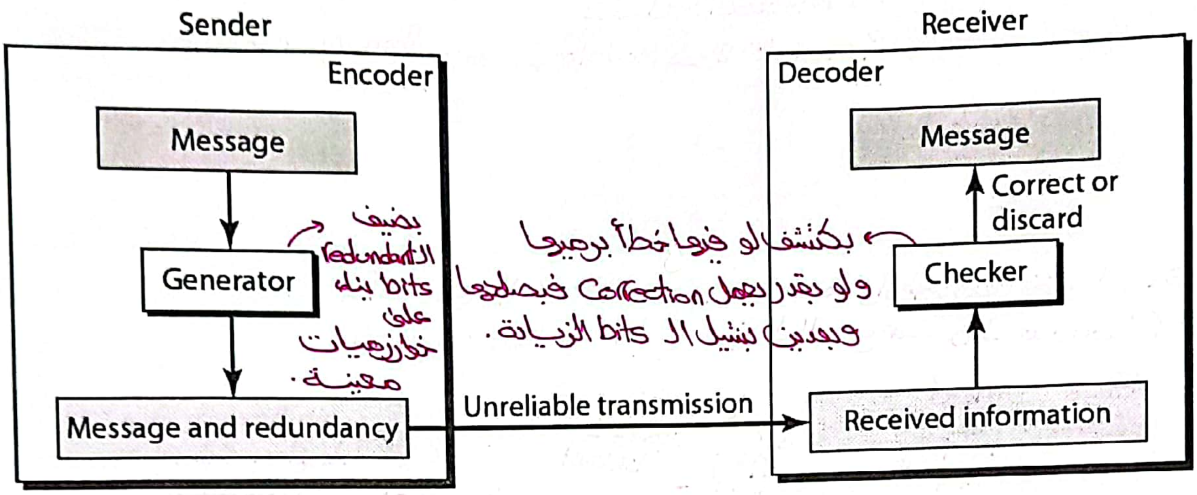
* قاعدة لاكتشاف الخطأ. مثل الـ Parity bit

To detect or correct errors, we need to send extra (redundant) bits with data.

"لازم يكون موجود"

"عشان نقدر نعمل detection و correction"

Figure 10.3 *The structure of encoder and decoder*



10.9 * في تطبيقات بتعتمد فقط عالـ detect لحتى ما ترفع الـ Packet للـ layers اللي فوق، بتعمل بعين detect مباشرة بترميه لا تهتم إذا سيعاد إرسال الـ Frame ولا لأ عشان توفر وقت وبعضهم لا بحاول دايما يصلح الـ errors ويعمل Correction ويحفظ الأخطاء اللي صارت.

Note

In this book, we concentrate on block codes; we leave convolution codes to advanced texts.

* نوعين من الـ Codes ، احنا حناخذ الأول *

10.10

* بل ما انشتغل بكل الأعداد الصحيحة
بنشتغل subsets منها.
* أصغر Modulo هو Modulo-2. → binary modulo.

Note

In modulo-N arithmetic, we use only the integers in the range 0 to N −1, inclusive.

* أهم فكرة بال modulo-2 هي فكرة الجمع والطرح وهي نفسها ← فإعملية الجمع والطرح بال binary : (بنهمل ال carry وال borrow) ال mod

EX : Mod 2

أهملناها

$$\begin{array}{r} 101 \\ +\;011 \\ \hline 110 \end{array} \qquad \begin{array}{r} 101 \\ -\;011 \\ \hline 110 \end{array} \Longrightarrow \underline{\underline{XOR}}$$

كإنو فاضلت 1 ولا كإنه استلفنا

سهل ووفير ومكلف إنه نعمله Implementation عال hardware.

10.11

عملية الجمع والطرح نفس النتيجة و نفس ناتج ال XOR
(المتشابهين = 0 والمختلفين = 1) (لو عدد ال bit فردي ناتج ال XOR = 1 ولو زوجي = 0)

Figure 10.4 *XORing of two single bits or two words*

$0 \oplus 0 = 0 \qquad 1 \oplus 1 = 0$

a. Two bits are the same, the result is 0.

$0 \oplus 1 = 1 \qquad 1 \oplus 0 = 1$

b. Two bits are different, the result is 1.

$$\begin{array}{cccccc} & 1 & 0 & 1 & 1 & 0 \\ \oplus & 1 & 1 & 1 & 0 & 0 \\ \hline & 0 & 1 & 0 & 1 & 0 \end{array}$$

c. Result of XORing two patterns

10-2 BLOCK CODING

In block coding, we divide our message into blocks, each of k bits, called datawords. We add r redundant bits to each block to make the length n = k + r. The resulting n-bit blocks are called codewords.

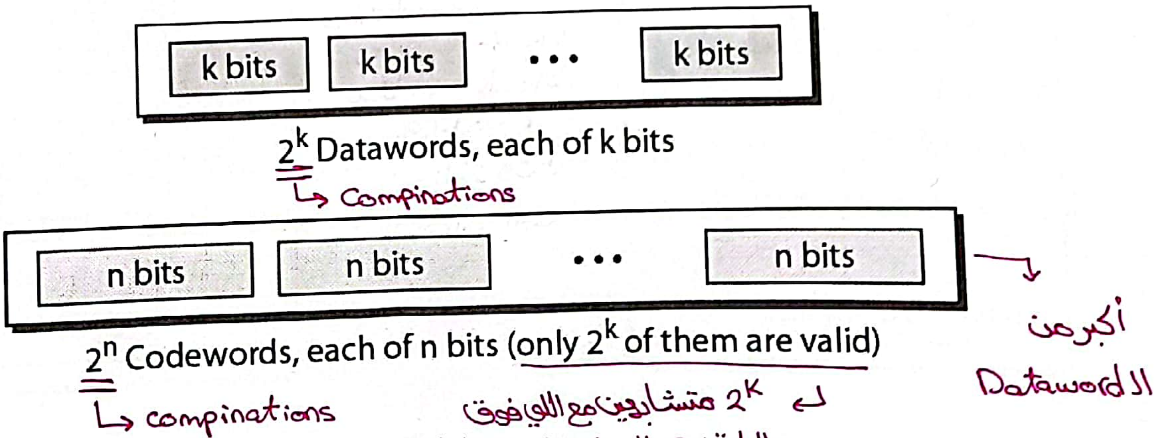
redundant

*في فرق بين الcodeword وبين ال dataword.

Topics discussed in this section:

Error Detection
Error Correction
Hamming Distance
Minimum Hamming Distance

Figure 10.5 ***Datawords and codewords in block coding***

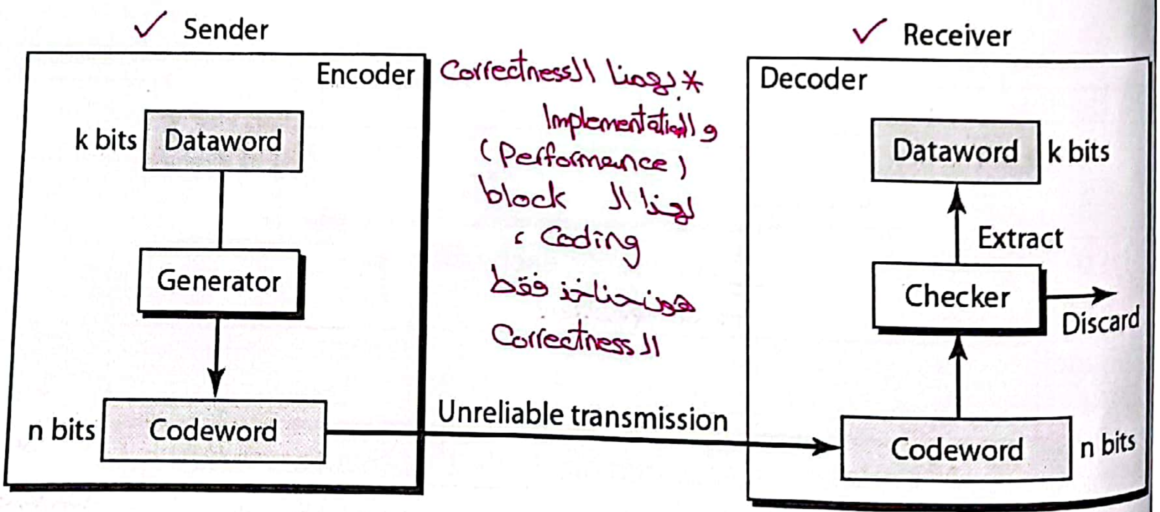


والباقي هي الزيادة اللي ضفناها. ← الزيادة مو عشوائية بناءً عقواعد معينة
(الزيادة بالعادة بتكون قليلة)

Example 10.1

* *The 4B/5B block coding discussed in Chapter 4 is a good example of this type of coding. In this coding scheme, k = 4 and n = 5. As we saw, we have $2^k = 16$ datawords and $2^n = 32$ codewords. We saw that 16 out of 32 codewords are used for message transfer and the rest are either used for other purposes or unused.*

* **Figure 10.6** *Process of error detection in block coding*



* يوضح الCorrectness
و الImplementation
(Performance)
لجزء ال block
Coding ،
هناخد فقط
الCorrectness

Example 10.2

2^2 datawords 2^3 code words

Let us assume that k = 2 and n = 3. Table 10.1 shows the list of datawords and codewords. Later, we will see how to derive a codeword from a dataword.

بتوافق مع ال Table.

Assume the sender encodes the dataword 01 as 011 and sends it to the receiver. Consider the following cases:

1. *The receiver receives 011. It is a valid codeword. The receiver extracts the dataword 01 from it.*

صحيح 100%

Example 10.2 (continued)

2. *The codeword is corrupted during transmission, and 111 is received. This is not a valid codeword and is discarded.*

" عمل detect بشكل صحيح "

3. *The codeword is corrupted during transmission, and 000 is received. This is a valid codeword. The receiver incorrectly extracts the dataword 00. Two corrupted bits have made the error undetectable.*

بل ما استقبل 01 استقبل 00 لإنك خربطت ب 2 bits
فما قدر يعمل detection .

Table 10.1 *A code for error detection (Example 10.2)*

Datawords	*Codewords*
00	000
01	011
10	101
11	110

* اذا استلمت اشي غير هدول مارح يعتبرها ك data

* Example:

* لو استلمت 011 معناها ال Dataword كانت 01

* لو استلمت 110 معناها ال data كانت 11 لكن لنفرض تغيرت أول bit وصار 111 فهيك حيشوف إنه هذا مش موجود بالجدول فبعمله discard فهيك عمل error detection بشكل صحيح.

اختلاف 2 bits على الأقل ممكن يخليك تستقبل bit ثانية غير عن اللي كانت لازم تستقبل (يعني مثلاً لو كان لازم تستقبل 01 فصار غلط واستقبلت 00)

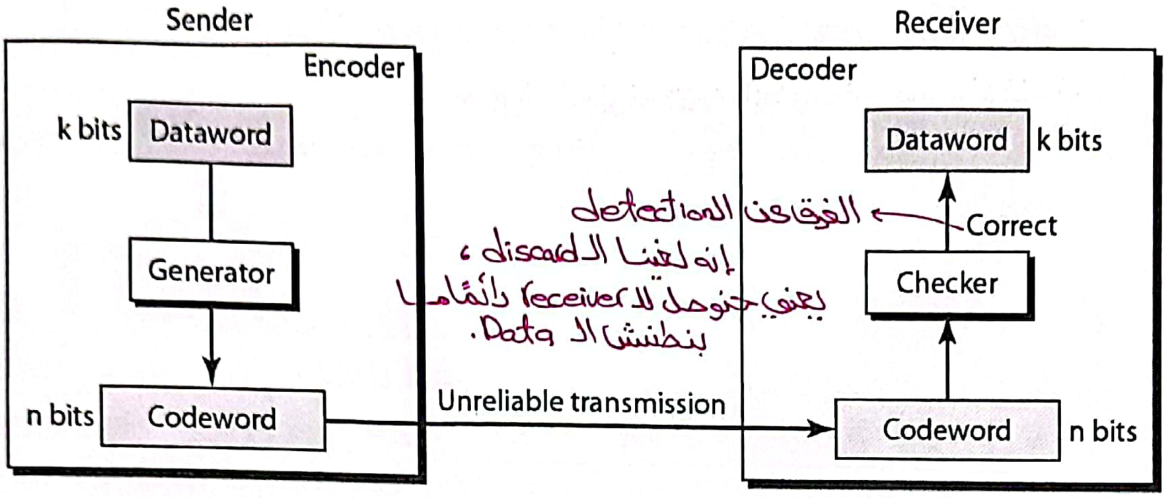
10.19

Note

An error-detecting code can detect only the types of errors for which it is designed; other types of errors may remain undetected.

10.20

Figure 10.7 *Structure of encoder and decoder in error correction*



10.21

Example 10.3

Let us add more redundant bits to Example 10.2 to see if the receiver can correct an error without knowing what was actually sent. We add 3 redundant bits to the 2-bit dataword to make 5-bit codewords. Table 10.2 shows the datawords and codewords. Assume the dataword is 01. The sender creates the codeword 01011. The codeword is corrupted during transmission, and 01001 is received. First, the receiver finds that the received codeword is not in the table. This means an error has occurred. The receiver, assuming that there is only 1 bit corrupted, uses the following strategy to guess the correct dataword.

bit وحدة
غيرها
مشكلة

10.22

bit compare (كل وحدة مع الثانية)

Example 10.3 (continued)

1. ***Comparing the received codeword with the first codeword in the table (01001 versus 00000), the receiver decides that the first codeword is not the one that was sent because there are two different bits.***

← مع إنها مفروض تكون bit 1 وحدة حسب افتراضنا.

2. ***By the same reasoning, the original codeword cannot be the third or fourth one in the table.***

3. ***The original codeword must be the second one in the table because this is the only one that differs from the received codeword by 1 bit. The receiver replaces 01001 with 01011 and consults the table to find the dataword 01.***

10.23

* لو كان الخطأ مش في bit واحده فكل الحل والافتراض بكون غير صحيح.

Table 10.2 *A code for error correction (Example 10.3)*

Dataword	Codeword
00	00000
01	01011
10	10101
11	11110

* 01001
00000
01001 = 2

* 01001
01011
00010 = 1 ✓

* 01001
10101
11100 = 3

* 01001
11110
10111 = 4

10.24

* بالـ Computer بحتاج الـ xor لأعمل شغلة المقارنة، الناتج بمثل الاختلافات.
(عدد الـ ones بمثل عدد الاختلافات).

Note

* **The Hamming distance between two words is the number of differences between corresponding bits.**

والاختلاف بصير لما يكون صاير error بال data اللي وصلت مقارنة باللي انبعثت.

EX: * $d(C_1, C_2) = 0 \longrightarrow$ This mean that $C_1 = C_2$ (no error)

10110 ↲ ↳ 10110

* $d(C_1, C_2) = 1 \longrightarrow$ This mean that $C_1 \neq C_2$ (with error)

11110 ↲ ↳ 10110

10.25

Example 10.4

Let us find the Hamming distance between two pairs of words.

✓*1. The Hamming distance d(000, 011) is 2 because*

$\boxed{000 \oplus 011 \text{ is } 011 \text{ (two 1s)}}$

✓*2. The Hamming distance d(10101, 11110) is 3 because*

$\boxed{10101 \oplus 11110 \text{ is } 01011 \text{ (three 1s)}}$

10.26

Note

أقل hamming بين set of words

* **The minimum Hamming distance is the smallest Hamming distance between all possible pairs in a set of words.**

* مهمة عشان لو الـ hamming distance عند الـ reciever كانت أقل من الـ minimum اللي طلعناه بعرف إنها مش موجودة بالـ datawords اللي عندي فبنشيلها .

10.27

* *Example 10.5*

Find the minimum Hamming distance of the coding scheme in Table 10.1.

Solution

We first find all Hamming distances.

$d(000, 011) = 2$	$d(000, 101) = 2$	$d(000, 110) = 2$	$d(011, 101) = 2$
$d(011, 110) = 2$	$d(101, 110) = 2$		

The d_{min} in this case is 2.

10.28

20 / 12 / 2022

Chapter 10 :

Slide 10.25 :

* ال Hamming distance هو مجموع الاختلافات بين ال bits الناتج من ال XOR.

Slide 10.27 :

Ex : ① 1011 Calculate the Minimum Hamming distance?

② 1001

③ 0111

Sol. { 1011, 1001, 0111 } (all possible compinations لازم نحسب ال)

1 : $\begin{array}{r} 1011 \\ 1001 \\ \hline 0010 \end{array} \oplus$ $\hookrightarrow d = 1$

2 : $\begin{array}{r} 1001 \\ 0111 \\ \hline 1110 \end{array} \oplus$ $\hookrightarrow d = 3$

3 : $\begin{array}{r} 1011 \\ 0111 \\ \hline 1100 \end{array}$ $\hookrightarrow d = 2$

$\longrightarrow d_{min} = 1 \triangleq d(1011, 1001) = 1$

*Example 10.6

Find the minimum Hamming distance of the coding scheme in Table 10.2.

Solution

We first find all the Hamming distances.

$d(00000, 01011) = 3$	$d(00000, 10101) = 3$	$d(00000, 11110) = 4$
$d(01011, 10101) = 4$	$d(01011, 11110) = 3$	$d(10101, 11110) = 3$

The d_{min} in this case is 3.

Note

لأنها لو كانت بس s فممكن الـ Codeword الخاطئة تمثل
عابرة صحيحة وهاي مشكلة . (حنشوف بالمثال الجاي)

***To guarantee the detection of up to s errors in all cases, the minimum Hamming distance in a block code must be $d_{min} = s + 1$.**

example : 1 bit error ⇒ $s=1 \rightarrow d_{min} = 1+1 = 2$

Example 10.7

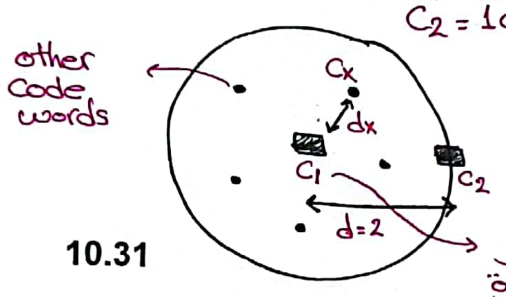
The minimum Hamming distance for our first code scheme (Table 10.1) is 2. This code guarantees detection of only a single error. For example, if the third codeword (101) is sent and one error occurs, the received codeword does not match any valid codeword. If two errors occur, however, the received codeword may match a valid codeword and the errors are not detected.

هون بقدر أجزم إنها مش ضمن الـ table لأنه كل الـ table أقل شي كان $d=2$

$d_{min}=2$

$C_1 = 011 \rightarrow C_x = 001 \rightarrow d(C_x, C_1) = 1 \rightarrow d_x < d_{min}$

$C_2 = 101 \qquad C_y = 101$



لو صار خطأ بـ 2 bits ممكن أضل داخل محيط الدائرة وتنحسب الـ code word على إنها صح وهذا لا يعتبر error detection.

$\rightarrow d(C_y, C_1) = 2$

مركز الدائرة ← (إحدى الـ code word اللي بالجدول)

10.31

Example 10.8

في خطأ بـ 2 bits بس ما التقط فالـ error ركون داخل الـ system فعشان هيك الـ min خلوه يساوي S+1

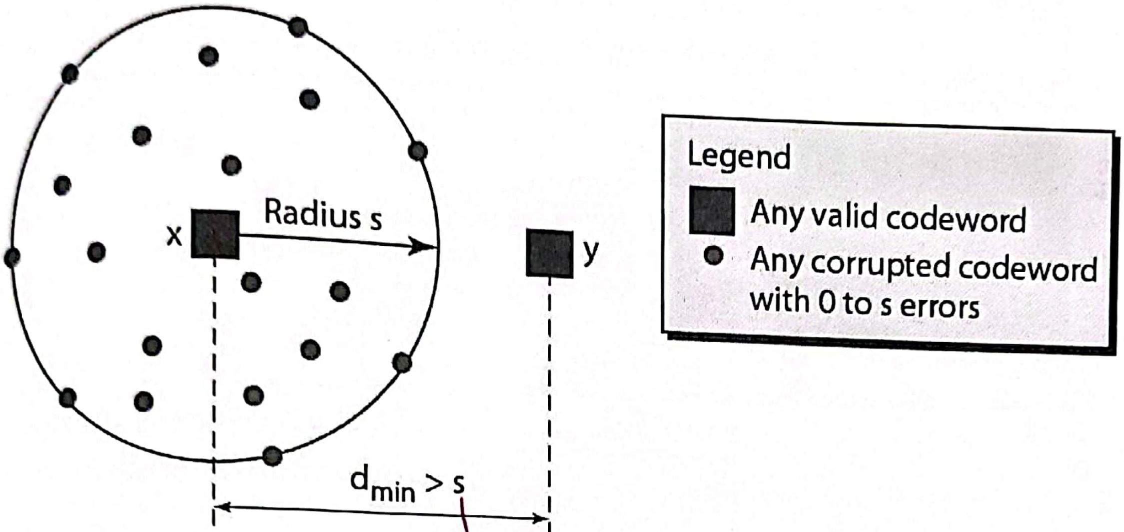
Our second block code scheme (Table 10.2) has $d_{min} = 3$. This code can detect up to two errors. Again, we see that when any of the valid codewords is sent, two errors create a codeword which is not in the table of valid codewords. The receiver cannot be fooled.

$\rightarrow d = S+1$

However, some combinations of three errors change a valid codeword to another valid codeword. The receiver accepts the received codeword and the errors are undetected.

10.32

Figure 10.8 *Geometric concept for finding d_{min} in error detection*



أي نقطة تقع على ال Circle.

10.33

* for Error Detection : $d_{min} = s+1$ and $(d_{min} > s)$

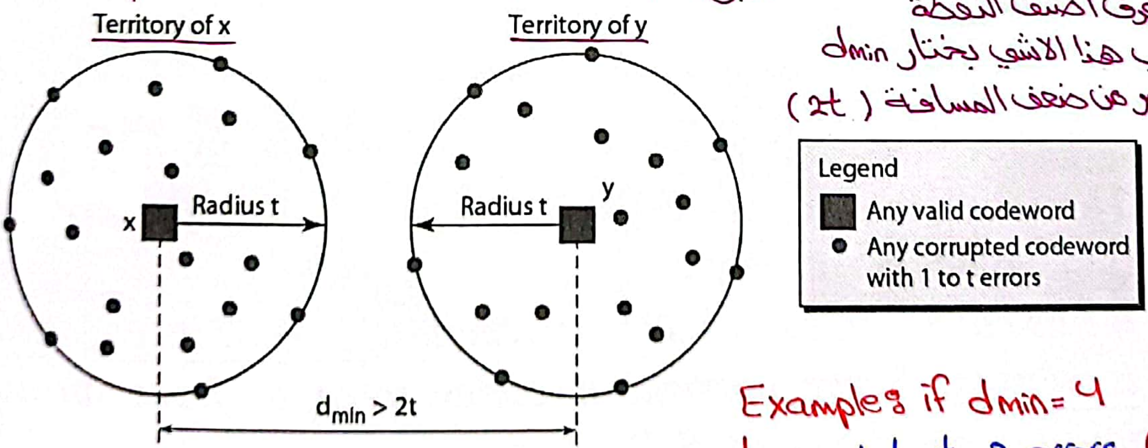
* for Error Correction : $d_{min} = 2t+1$ and $(d_{min} > 2t)$

Figure 10.9 *Geometric concept for finding d_{min} in error correction* مش detection

أي نقطة تقع ضمنها بعتبرها x

أي نقطة تقع ضمنها بعتبرها y

* إذا تقاطعوا الدائرتين مع بعض حبطل أعرف أصنف النقطة فلأتجنب هذا الاشي بختار d_{min} تكون أكبر من ضعف المسافة $(2t)$



Examples: if $d_{min} = 4$

↳ can detect 3 errors $(s+1)$

↳ can correct 1 error $(2t+1)$

10.34

Note

To guarantee correction of up to *t* errors in all cases, the minimum Hamming distance in a block code must be $d_{min} = 2t + 1$.

Example 10.9

A code scheme has a Hamming distance $d_{min} = 4$. What is the error detection and correction capability of this scheme?

Solution

This code guarantees the detection of up to three errors ($s = 3$), but it can correct up to one error. In other words, if this code is used for error correction, part of its capability is wasted. Error correction codes need to have an odd minimum distance (3, 5, 7, . . .).

جزء من الـ bit راح wasted

"حنتطلع كوجهة النظر الهندسية مش الرياضية"

10-3 LINEAR BLOCK CODES

* *Almost all block codes used today belong to a subset called **linear block codes**. A linear block code is a code in which the exclusive OR (addition modulo-2) of two valid codewords creates another valid codeword.*

↳ for example: * 01011 ⊕ 10101 = 11110 ← طلعلي وحدة Valid موجودة بجدول ال Codewords (Table 10.2)

* 10101 ⊕ 00000 = 10101 ← Valid ومتاحة بالجدول

Topics discussed in this section:

Minimum Distance for Linear Block Codes

Some Linear Block Codes

* فإذاً أول شرط من شروط ال linear code إنو إذا أخذت 2 codewords من الجدول وعملت بينهم XOR حيطلعولك codeword ثانية موجودة بالجدول برضه

10.37

* In general: | Dataword (k bits) | 1 bit (additional bits) | → n bits code word ($k+1$)

* $n = k + 1$ ← لا مش بس زيادة bit وحدة، ممكن تكون الزيادة أكثر من 1.

↳ d_{min} = Single bit error ($s=1$) → $d_{min} = s+1 = 2$

* **Note**

In a linear block code, the exclusive OR (XOR) of any two valid codewords creates another valid codeword.

10.38

Example 10.10

Let us see if the two codes we defined in Table 10.1 and Table 10.2 belong to the class of linear block codes.

1. ***The scheme in Table 10.1 is a linear block code because the result of XORing any codeword with any other codeword is a valid codeword. For example, the XORing of the second and third codewords creates the fourth one.*** → Examples: 101 ⊕ 110 = 011 → valid in table

2. ***The scheme in Table 10.2 is also a linear block code. We can create all four codewords by XORing two other codewords.***

*العمليات الحسابية عال linear أسهل عشان هيك مابنروح عال Non-linear.

Example 10.11

In our first code (Table 10.1), the numbers of 1s in the nonzero codewords are 2, 2, and 2. So the minimum Hamming distance is $d_{min} = 2$. In our second code (Table 10.2), the numbers of 1s in the nonzero codewords are 3, 3, and 4. So in this code we have $d_{min} = 3$.

* النقطة الموجودة في ال linear إني قدرت أحصر ال Set لإنه بعرف إنه كل 2 Valid codewords بيعمل Xor بتطلعلي Valid codeword بال table.

فكرتها إنك تخلي عدد الـ 1's في الـ word
دائمًا يا even يا odd ، احنا حنتعامل مع الـ even

Note

* **A simple parity-check code is a single-bit error-detecting code in which $n = k + 1$ with $d_{min} = 2$.**

Data word

$\overbrace{001}^{k=3}$ | 1

total # of bits = 3+1 = 4

بضيفها بحيث الـ total number of 1's تكون even .

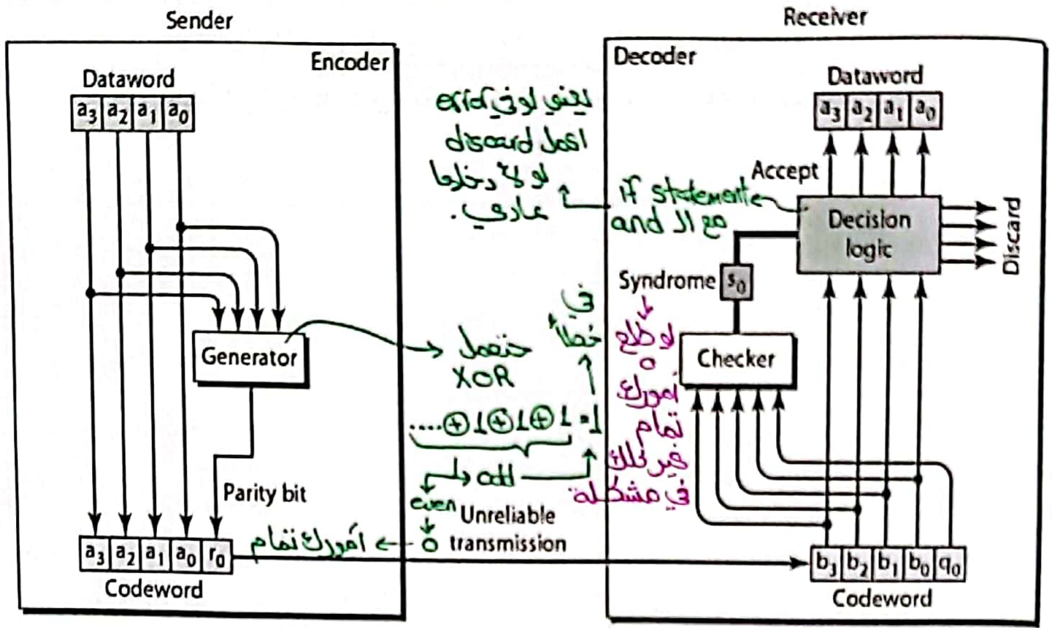
* حتلاحظ بـ Table 10.1 إنه آخر bit كنا نضيفها هي Parity-check .
* Table 10.2 مش Parity check ، لأنه أضفنا أكثر من 1 bit *

Table 10.3 *Simple parity-check code C(5, 4)* min d = 2 → ثابتة زي قاعدة

Datawords	Codewords	Datawords	Codewords
0000	00000	1000	10001
0001	00011	1001	10010
0010	00101	1010	10100
0011	00110	1011	10111
0100	01001	1100	11000
0101	01010	1101	11011
0110	01100	1110	11101
0111	01111	1111	11110

بعمل detect فقط لل error .

Figure 10.10 *Encoder and decoder for simple parity-check code*



* ما حيلقط الأخطاء ال even حيلقط بس ال odd .

10.43

Example 10.12

مثال بشرحلك الآلية لدي الرسمة

Let us look at some transmission scenarios. Assume the sender sends the dataword 1011. The codeword created from this dataword is 10111, which is sent to the receiver. We examine five cases:

وصلت زي ما هي طبيعية

1. *No error occurs; the received codeword is 10111. The syndrome is 0. The dataword 1011 is created.* برفعوا من غير ال Parity
2. *One single-bit error changes a_1. The received codeword is 10011. The syndrome is 1. No dataword is created.*
3. *One single-bit error changes r_0. The received codeword is 10110. The syndrome is 1. No dataword is created.* (ال Parity bit)

10.44

لو رغم إنو ال data فعليًا ما فيها خطأ

(هاي من الأضرار الجانبية لموضوع ال Coding مع ال Parity)

Example 10.12 (continued)

Parity

4. ***An error changes r_0 and a second error changes a_3. The received codeword is 00110. The syndrome is 0. The dataword 0011 is created at the receiver. Note that here the dataword is wrongly created due to the syndrome value.*** → Parity مع إنه عني خطئين واحد في ال data وواحد بال
5. ***Three bits—a_3, a_2, and a_1—are changed by errors. The received codeword is 01011. The syndrome is 1. The dataword is not created. This shows that the simple parity check, guaranteed to detect one single error, can also find any odd number of errors.***

* هاي الطريقة Simple فيها مشاكل.

* حنشوف غير ال linear codes اشي اسمه ال cyclic codes
اللي بتعمل detect لل error مع Probability.

* من أهم ال detect error الموجود بال wifi (IEEE 802.11) بيعمل
detects لل error مع Probability
(CRC Protocol)

Note

*** A simple parity-check code can detect an odd number of errors. ***

Note

*All Hamming codes discussed in this book have $d_{min} = 3$.

Can detect up to 2 errors ←
↳ $s = 2$

* The relationship between m and n in these codes is $n = 2m - 1$.

10.47

طريقة أفضل للـ error detection لإنه بـ Parity check وحدة كنا نقدر نحمل بس للـ odd error الـ even error مانقدر

Figure 10.11 *Two-dimensional parity-check code*

((بحسب الـ Parity على الـ rows والـ Columns))

* الـ data بترتيبها بـ rows ، الأصل بنعرفها Sequential بس احنا بنفسك كل عدد معين منهم وبنحطهم تحت بعض

rows = 7, Column = 4

							Row parities
1	1	0	0	1	1	1	1
1	0	1	1	1	0	1	1
0	1	1	1	0	0	1	0
0	1	0	1	0	0	1	1
0	1	0	1	0	1	0	1

Column parities

a. Design of row and column parities

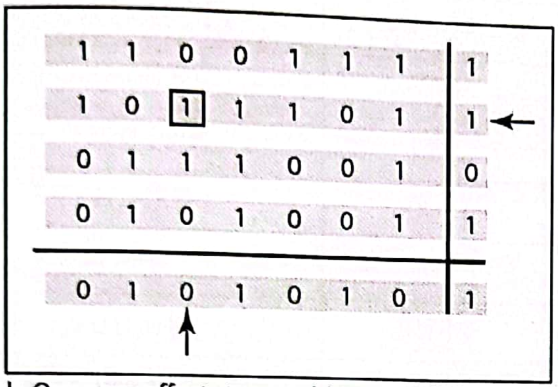
إضافتهم عشان نخلي عدد الـ 1's even

* لو ضربت bit 2 بتأثر ع 2 parity *

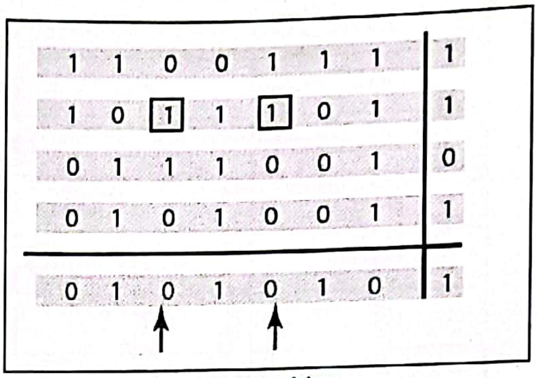
10.48

بس زيادة overhead / "أفضل من ال one"

Figure 10.11 *Two-dimensional parity-check code*



b. One error affects two parities

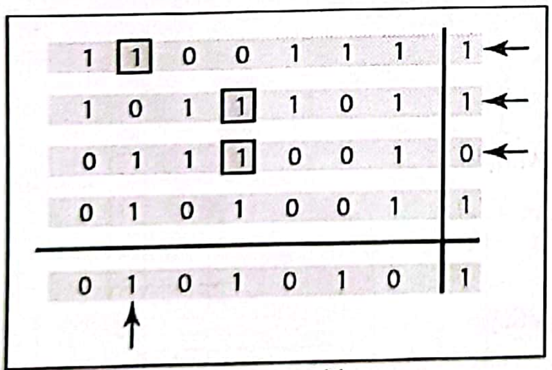


c. Two errors affect two parities

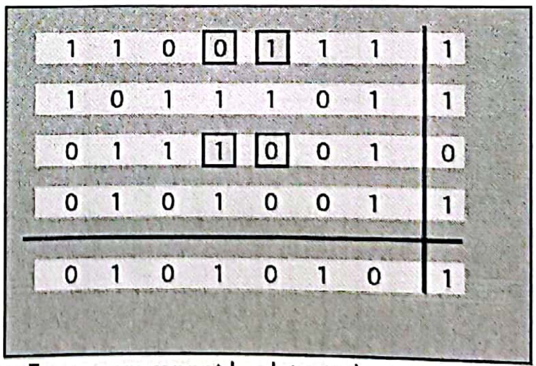
* لو كانت كل ال data موجودة بـ row واحد ما كنت حقدر اعمل detect زي هون *

10.49

Figure 10.11 *Two-dimensional parity-check code*



d. Three errors affect four parities



e. Four errors cannot be detected

أخطر الأنواع ، يعني لما ممكن يجيني error ما أقدر أعملها detect بس بضل أفضل من ال Single Parity.

10.50

طريقة تانية غير الـ Parity .

اختيارهم بناءً على طريقة معينة

3 bits
مش بس 1 .

Table 10.4 *Hamming code C(7, 4)*

Datawords	Codewords	Datawords	Codewords
0000	0000000	1000	1000110
0001	0001101	1001	1001011
0010	0010111	1010	1010001
0011	0011010	1011	1011100
0100	0100011	1100	1100101
0101	0101110	1101	1101000
0110	0110100	1110	1110010
0111	0111001	1111	1111111

* error detection and correction :
يعني بعرف إنه في bit ضربت وكمان بعرف وين ضربت
عشان أقدر أعدلها .

Figure 10.12 *The structure of the encoder and decoder for a Hamming code*

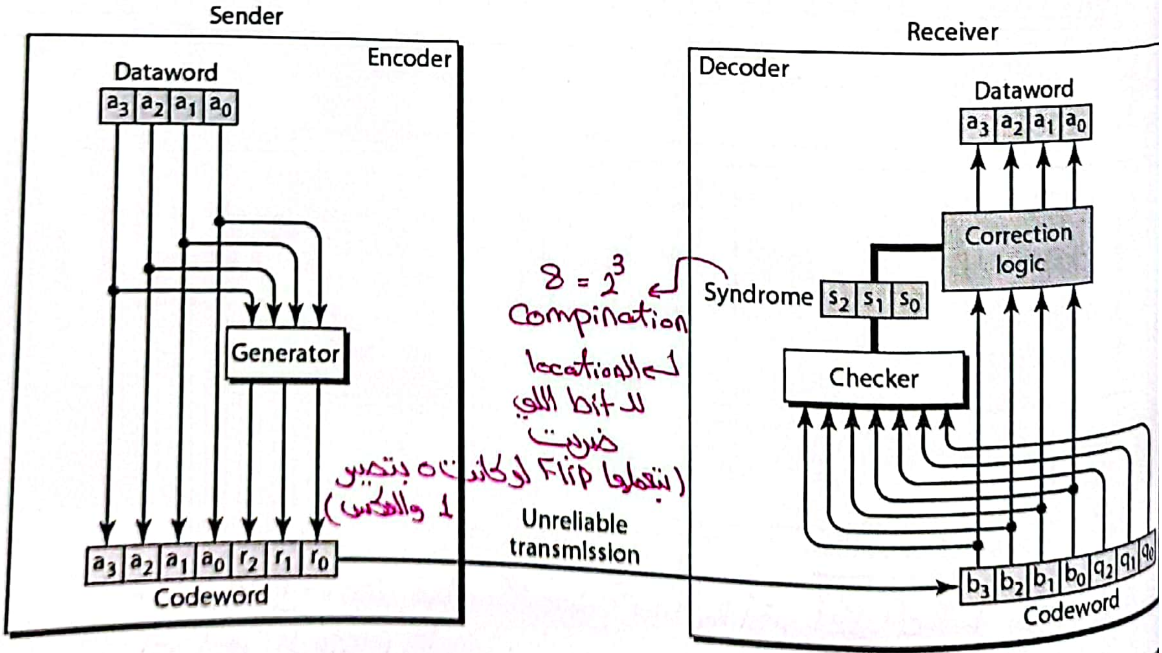


Table 10.5 *Logical decision made by the correction logic analyzer*

Syndrome	000	001	010	011	100	101	110	111
Error	None	q_0	q_1	b_2	q_2	b_0	b_3	b_1

(بستنتج موقع ال error وبعمله Flip لل bit)

Example 10.13

Let us trace the path of three datawords from the sender to the destination:

1. ***The dataword 0100 becomes the codeword 0100011. The codeword 0100011 is received. The syndrome is 000, the final dataword is 0100.***
2. ***The dataword 0111 becomes the codeword 0111001. The syndrome is 011. After flipping b_2 (changing the 1 to 0), the final dataword is 0111.***
3. ***The dataword 1101 becomes the codeword 1101000. The syndrome is 101. After flipping b_0, we get 0000, the wrong dataword. This shows that our code cannot correct two errors.***

$k \geq 7 \rightarrow (n-m) \geq 7 \rightarrow (2^m - 1 - m) \geq 7$
$\rightarrow$ we try s $\rightarrow m=3 \rightarrow$ gives $4 \rightarrow 4 \geq 7$ (wrong)
$\hookrightarrow m=4 \rightarrow$ gives $11 \rightarrow 11 \geq 7$ (right)

Example 10.14

We need a dataword of at least 7 bits (dataword). Calculate values of k and n that satisfy this requirement.

Solution

عدد ال bits

We need to make $k = n - m$ greater than or equal to 7, or $2m - 1 - m \geq 7$. (2^m)

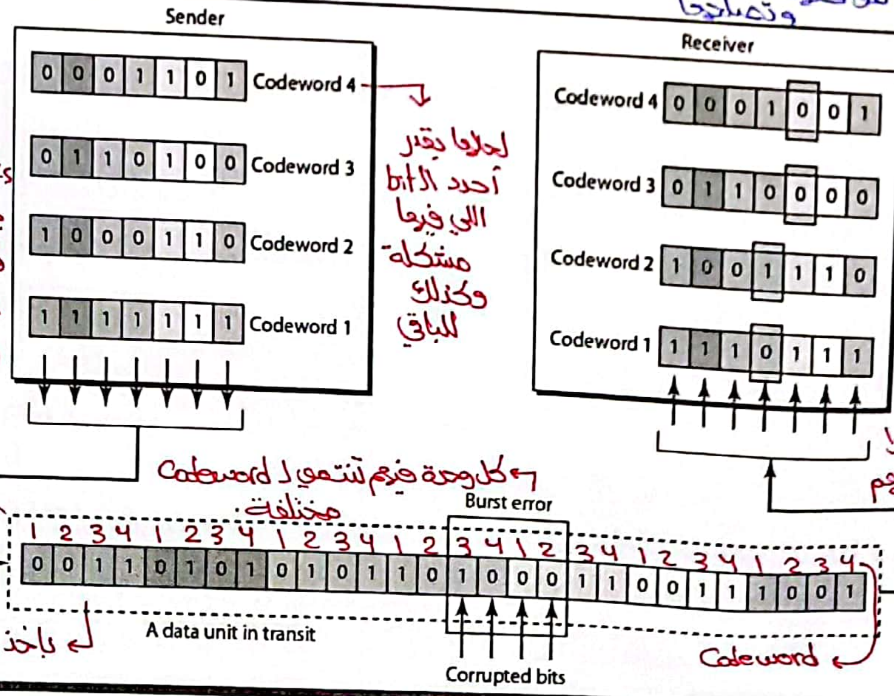
1. If we set $m = 3$, the result is $n = 23 - 1$ and $k = 7 - 3$, or 4, which is not acceptable. → أكبر من ال range بطلع
2. If we set $m = 4$, then $n = 24 - 1 = 15$ and $k = 15 - 4 = 11$, which satisfies the condition. So the code is

$$\boxed{C(15, 11)}$$

Figure 10.13 *Burst error correction using Hamming code*

لو مجموعة من ال bits ضربت كيف ممكن تكتشف مواقعها وتصلحها

* حسب ال Protocols في ال Computer and Communication networks دائما بنبعث ال Frame او Packet كامل فكيف بدي اكتشف لو صار مجموعة من ال bits فيها في يوم خطأ.



لحالها بتقدر أحدد ال bit اللي فيها مشكلة وكذلك للباقي

نعرف مكان كل error كود فبتصلحوم لشبه ال Time Division Multiplexing

كل وحدة فيهم تنتمي ل Codeword مختلفة

حط أكثر من Codeword داخل هذا ال Packet

باخذ كل row ورا الثاني

Codeword

* بتقدر تحلها كمان لو عندك 2 Burst error

* المشاكل ممكن تصير لو عندك 5 Burst error فممكن يطلعلك 2 errors بال Codeword الوحدة فممكن ينفذ هنا ال error.

27/12/2022

Slide 10.53 : " كيف طلعنا هذا الجدول " → XOR

Receiver

$r_0 = a_2 + a_1 \oplus a_0$ modulo-2

$r_1 = a_3 + a_2 + a_1$ modulo-2

$r_2 = a_1 + a_0 + a_3$ modulo-2

Sender

$S_0 = b_2 + b_1 + b_0 + q_0$ Modulo-2

$S_1 = b_3 + b_2 + b_1 + q_1$ modulo-2

$S_2 = b_1 + b_0 + b_3 + q_2$ modulo-2

* لنفرض b_0 صار فيها error مين حيتأثر من المعادلات ؟ S_0 و S_2

* b_1 〃 〃 〃 〃 〃 〃 〃 〃 ؟ S_0 و S_1 و S_2

* b_2 〃 〃 〃 〃 〃 〃 〃 〃 ؟ S_1 و S_2

* ممكن اللي يضرب ال q مش ال b فهيك ما حنشوف ال error بس تذكر انه احنا بنشتغل s 1 bit error فلو صار عندك 2 bit error هاي الطريقة ما بتقدر تتعامل معه.

Five Apple

* Slide 10.54 : Example : لنشرح أول Senario

XOR

Modulo-2

$S_0 = b_2 + b_1 + b_0 + q_0$ Modulo-2

$S_1 = b_3 + b_2 + b_1 + q_1$ //

$S_2 = b_1 + b_0 + b_3 + q_2$ //

Data

a_3	a_2	a_1	a_0	r_2	r_1	r_0	Sender
0	1	0	0	0	1	1	

$r_0 = a_2 + a_1 + a_0$ modulo-2

$r_1 = a_3 + a_2 + a_1$ //

$r_2 = a_1 + a_0 + a_3$ //

b_3	b_2	b_1	b_0	q_2	q_1	q_0	reciever

Codeword

e.g : $r_0 = 1 + 0 + 0 = 1$

$r_1 = 1$

$r_2 = 0$

* ثاني سيناريو :

Dataword

a_3	a_2	a_1	a_0	r_2	r_1	r_0
0	1	1	1	0	0	1

$r_0 = 1$

$r_1 = 0$

$r_2 = 0$

b_3	b_2	b_1	b_0	q_2	q_1	q_0

codeword

10-4 CYCLIC CODES

Cyclic codes are special linear block codes with one extra property. In a cyclic code, if a codeword is cyclically shifted (rotated), the result is another codeword.

نفس ال linear مع شغلة اضافية اللي هي ال (rotated)
Slide 6.37

* حنشوف ال bits بتلف بشكل دائري
كأنها cycle . *

لو عملت Shift لـ Codeword بتطلع Valid Codeword ثانية :

general :

$A = a_6\ a_5\ a_4\ a_3\ a_2\ a_1\ a_0$

lift shift

$a_5\ a_4\ a_3\ a_2\ a_1\ a_0\ a_6$

$B = b_6\ b_5\ b_4\ b_3\ b_2\ b_1\ b_0$

another valid codeword

Topics discussed in this section:

CRC ← Cyclic Redundancy Check (ال bits الزيادة)
Hardware Implementation
Polynomials
Cyclic Code Analysis
Advantages of Cyclic Codes
Other Cyclic Codes

مستخدم بال wifi حاليا (IEEE 802.11)

10.57

* حنشوف هون إنه في errors ممكن تمرق وما بنعملها detect بس باحتمالية قليلة جدا.

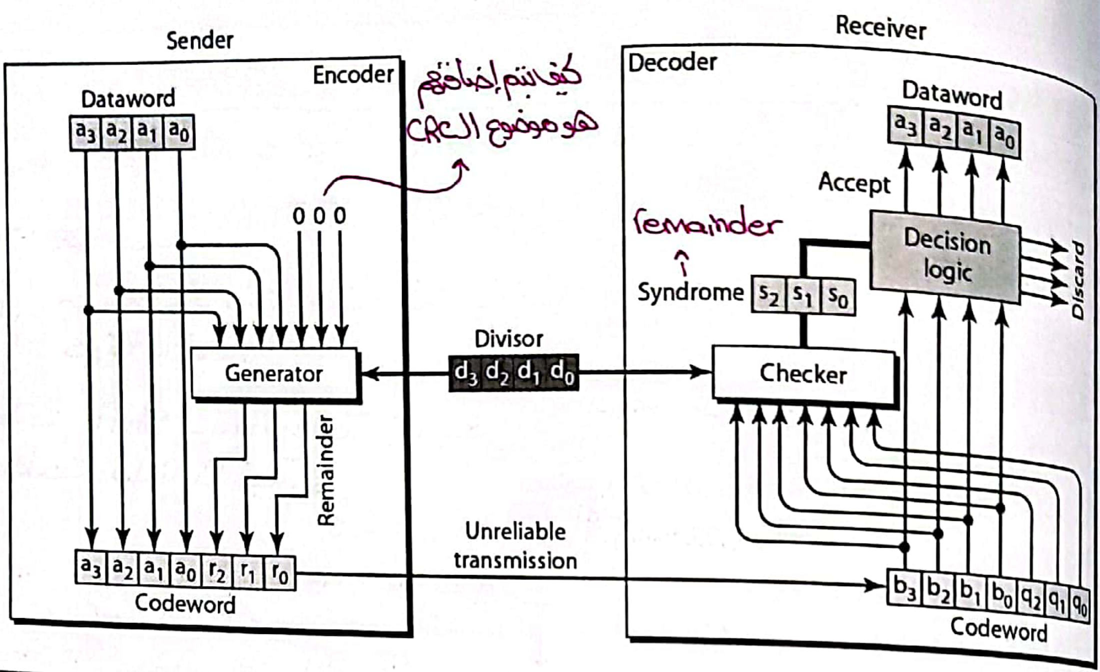
Table 10.6 *A CRC code with C(7, 4)*

→ Data word
↳ code word

Dataword	*Codeword*	*Dataword*	*Codeword*
0000	0000000	1000	1000101
0001	0001011	1001	1001110
0010	0010110	1010	1010011
0011	0011101	1011	1011000
0100	0100111	1100	1100010
0101	0101100	1101	1101001
0110	0110001	1110	1110100
0111	0111010	1111	1111111

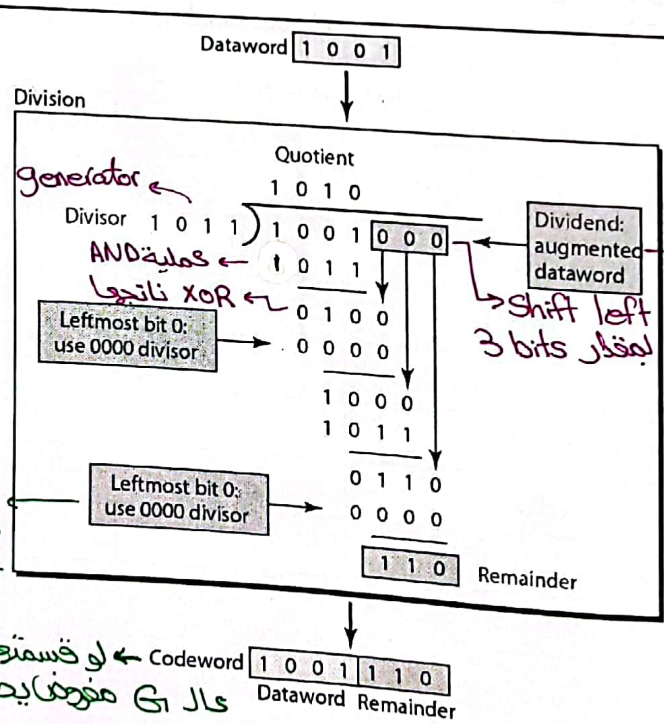
10.58

Figure 10.14 *CRC encoder and decoder*



10.59 *بالعادة بنعمل للـ data word إزاحة بمقدار معين (Shift left) لنترك مساحة
للـ remainder bits بدل ما نعمل عملية XOR . (طريقة إضافة الـ remainder bits)

Figure 10.15 *Division in CRC encoder*



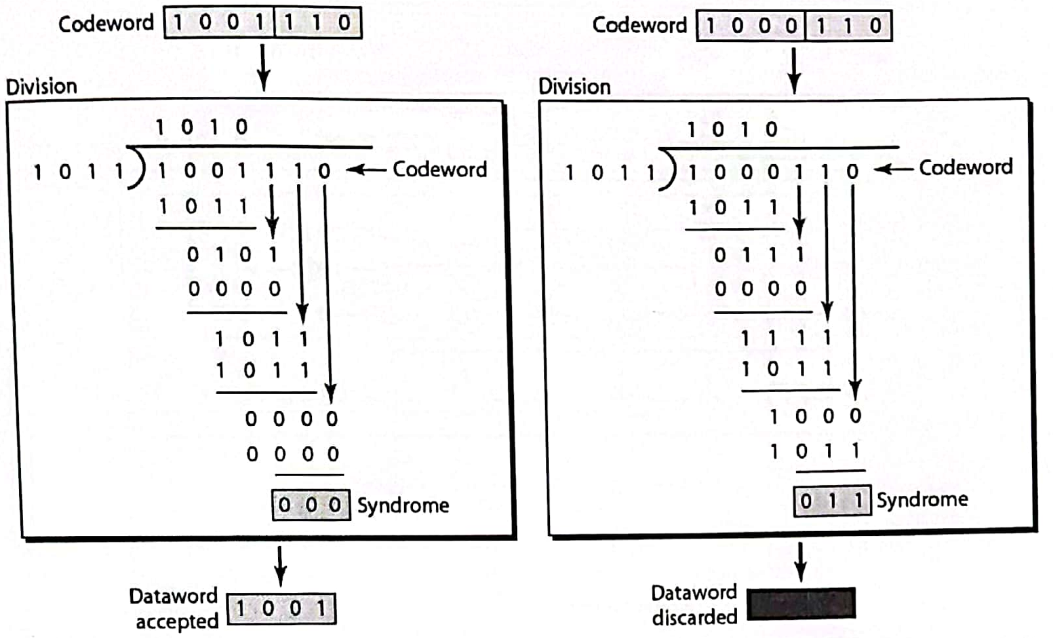
* تبدأ عملية القسمة
من الـ Most sig. bit
(لو كانت 1 بنحط 1
ولو 0 بنحط 0)
وبعد كل عملية بطنش
الـ MSb ونزحف لليمين.

فعليا عملية القسمة بلشت
عند T=4 قبلها كنا نعمل
Shift و XOR

لو قسمتها ← Codeword
على الـ G مفروض يطلع الناتج = صفر ما في أخطاء

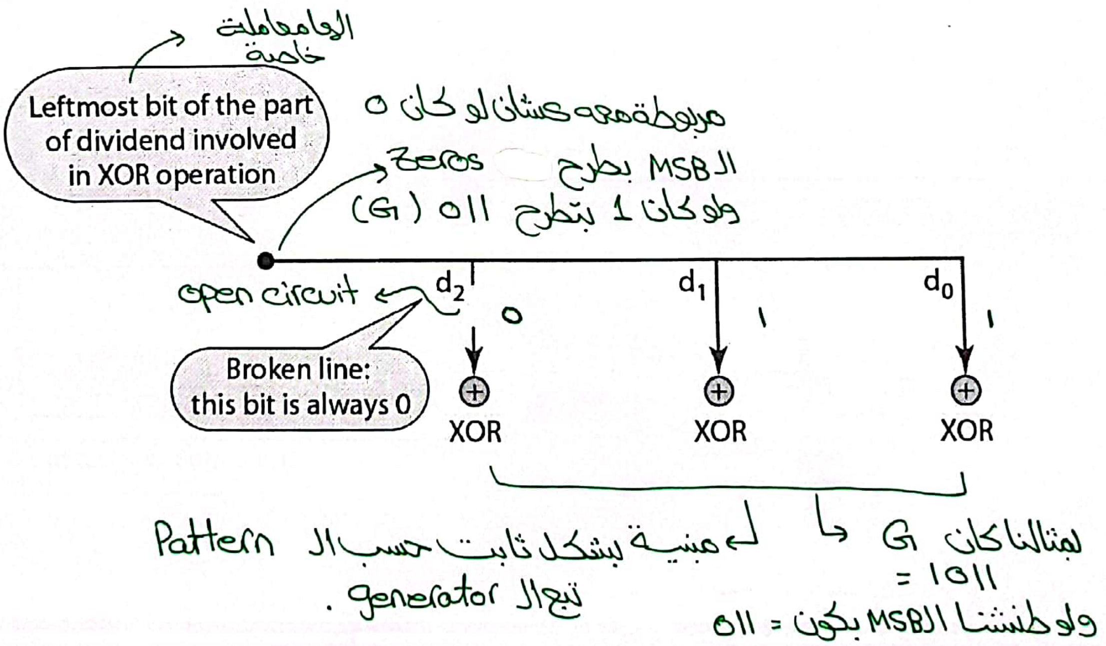
بالـ bits **10.60**

Figure 10.16 *Division in the CRC decoder for two cases*



10.61

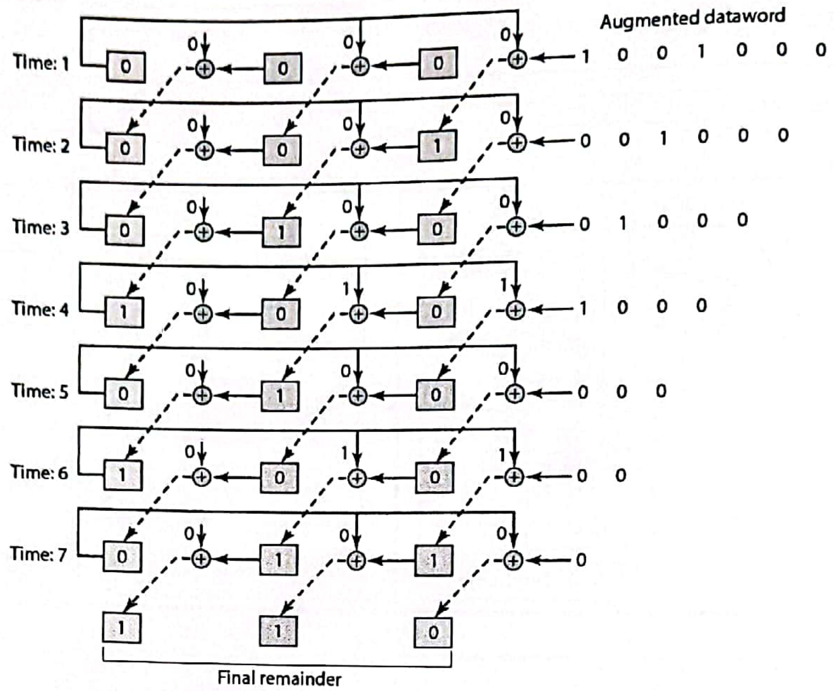
Figure 10.17 *Hardwired design of the divisor in CRC*



10.62

*كل ما كان ال generator فيه 0's أكثر كل ما كانت تكلفته أقل، وانت بتختار حسب نوع ال error الي بدك تعمله detect، ولازم نفس ال generator المستخدم عند ال Sender يكون مستخدم عند ال reciever

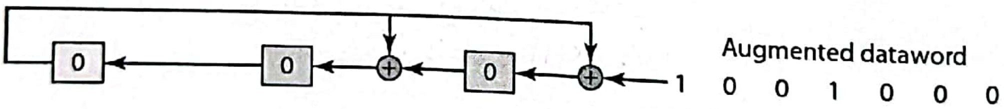
Figure 10.18 *Simulation of division in CRC encoder*



10.63

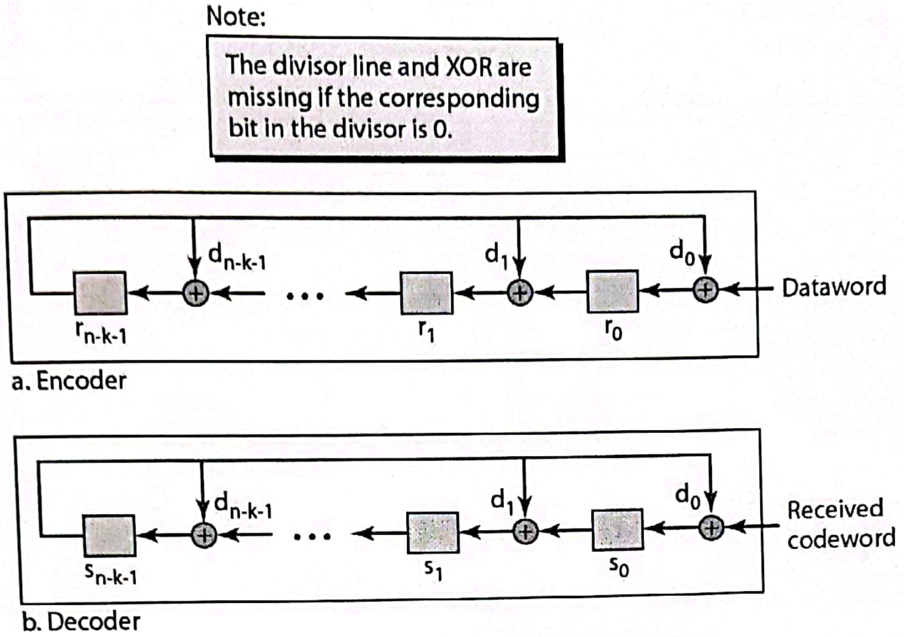
* مو ضروري احتفظ بجميع ال registers السابقة بوضعي بس آخر وحدة فبختصر عدد ال register

Figure 10.19 *The CRC encoder design using shift registers*



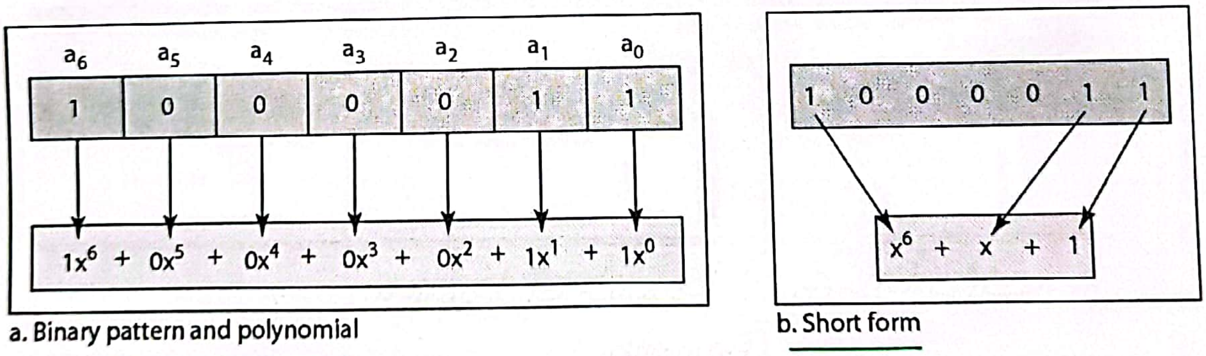
10.64

Figure 10.20 *General design of encoder and decoder of a CRC code*



10.65

Figure 10.21 *A polynomial to represent a binary word*



10.66

29/12/2022

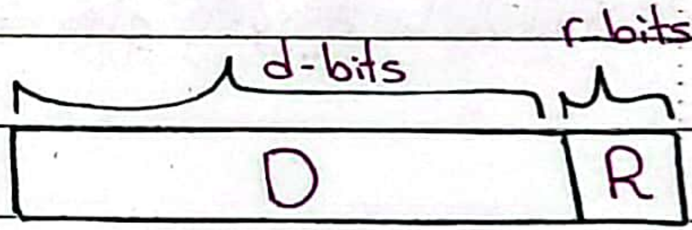
*Slide 10.59

Cyclic Redundancy Check (CRC)

→ d - bit piece of data (size of data word)

→ r+1 bit pattern agreed upon between sender and reciever, known as "generator" (G).

→ The most significant bit (MSB) in the generator is always "1".



$2^r D + R$

→ $2^r D$ XOR R

* Note (تذكير) :

$$\begin{array}{r|l} & 3 \rightarrow q \\ n \leftarrow 2 & 7 \rightarrow d \\ & \underline{6} \\ & 1 \rightarrow r \end{array}$$

$\Rightarrow 7 = 2(3) + 1$

$\boxed{* \; d = nq + r}$ "In general"

→ In binary :

D بدي أضيف عليها r (الها Shift بمقدار معين من ال bits)

$$\begin{array}{r|l} & N \\ G & D \cdot 2^r \;] \rightarrow \text{كأني عملت Shift بمقدار } r \\ & \vdots \\ & R \end{array}$$

→ $D \cdot 2^r = NG \text{ XOR } R$ → $D \cdot 2^r \text{ XOR } R = NG$ → $D \cdot 2^r + R = NG$

لـ نفسها الجمع أو الطرح ب Modulo-2

لو عملت هيك بطلعلك ال Codeword

* فالخطوات كالتالي : حنعمل ال dataword إزاحة (Shift left) بمقدار r من ال bits بعد ذلك بدك تقسمها على ال divisor (generator "G") وبتطلع من هاي القسمة ال remainder (R) بعد ذلك بتضيف ال remainder إلى ال $D \cdot 2^r$ فهيك بتعمل ال Codeword تبعتك وبتبعتها للطرف الثاني ، الطرف الثاني رح يوصله ال Codeword ويقسمها على ال generator ، إذا طلع معاه الجواب صفر معناه مافي errors ولو طلع مش صفر معناه في error . احتمال بسيط يكون فيه بس ما قدرنا نحدده ←

* المعلومات المشتركة بين الاثنين (Sender, reciever) هي ال generator .

* عدد ال bits في ال generator أكثر بمقدار 1 من ال remainder لذلك بس نعمل عملية القسمة بتكون ناقصة بمقدار bit واحدة من قيمة ال divisor .

* عمليات الجمع والطرح عبارة عن XOR ما فيها carry فحتكون العمليات بسيطة Shift و XOR . (عشان هيك مفضلة) .

* Slide 10.66 : 1 0 1 1 0 (binary)

→ convert it to Polynomial (أسهل)

لو ما كان 1 ممكن ما نعرف الـ degree تبعنا لو اللي بعدها كله Zero

$1x^4 + 0x^3 + 1x^2 + 1x^1 + 0x^0$ → Position of the bit

→ $x^4 + x^2 + x$ (بيمثل عدد ال bits ناقص 1) زوجي

degree of the polynomial (أعلى أس)

* بنسهم موجب زي بعض فـ 0 =

* Addition (Using Modulo-2) :

$$\begin{array}{r} x^2 + 1 \\ \oplus \; x^3 + x^2 \\ \hline x^3 + 1 \end{array}$$

* لو كان $3x^2$ فوق بعض بكون الناتج x^2 لإنه عددهم فردي .

* divide polynomial : $\frac{X^5}{X^2} = X^3$

* $X^2(X+1)$: $X^3 + X^2$ (عملية Shift كأنها)

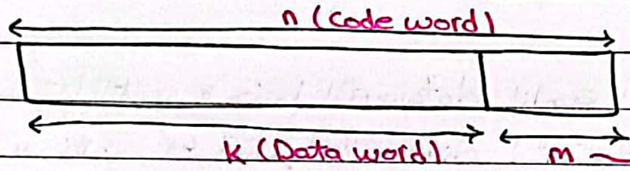
* فيديو عن ال hamming distance وحل المثال 10.14 , Slide 10.55 *

Hamming Codes :

* error - correcting codes.
* originally designed with d-min = 3.
* thus,

1- d-min = S+1 → S=2 → able to detects 2 errors

2- d-min = 2t+1 → t=1 → able to correct 1 error



m → additional bits added to data word

(k, n, m : sizes in bits)

1: k = n - m → m = n - k

2: $n = 2^m - 1$

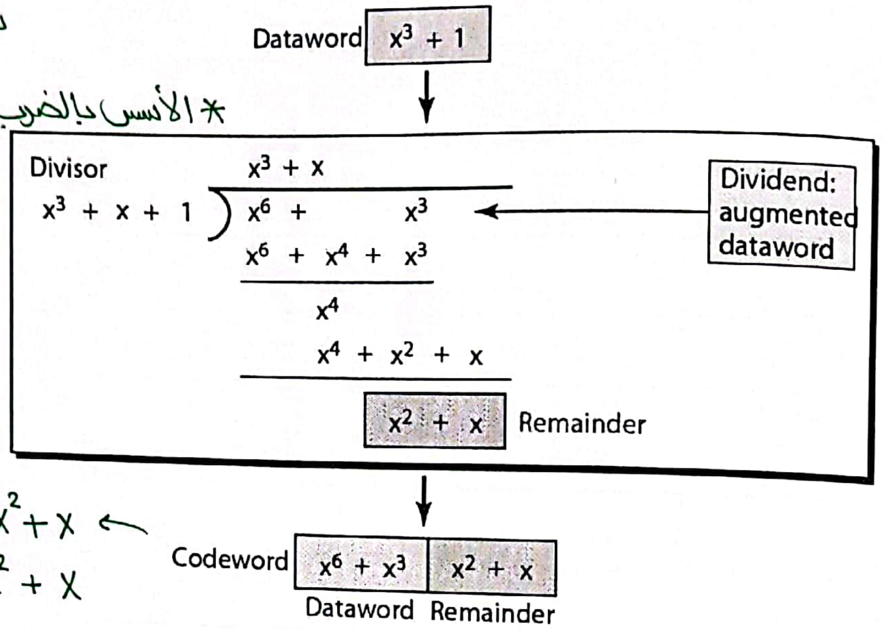
↳ من خصائص ال hamming codes

→ C(n, k) is used to represent a coding scheme.

Figure 10.22 *CRC division using polynomials*

الComputer مابقدر يعمل هيك بعملها
بال binary زي قبل.

* الأس بالضرب تجمع .



$X^3(X^3+1)+X^2+X \leftarrow$

$= X^6+X^3+X^2+X$

10.67

Note

مايتم اختياره عشوائي

بنختاره بذكاء ليلقط عدد معين من ال error ← بناء تحليلات رياضية معينة

The divisor in a cyclic code is normally called the generator polynomial or simply the generator.

* بدل ما نسميها G حنسميها G(x) لأنها صارت بال Polynomial .

10.68

Note

In a cyclic code,
If $s(x) \neq 0$, one or more bits is corrupted.
If $s(x) = 0$, either

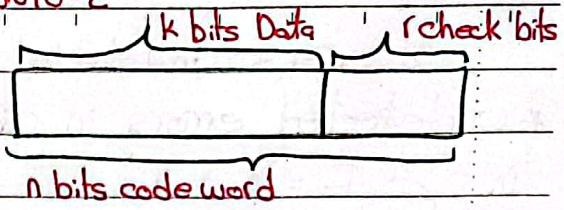
a. No bit is corrupted. or
b. Some bits are corrupted, but the decoder failed to detect them.

Note

In a cyclic code, those $e(x)$ errors that are divisible by $g(x)$ are not caught.

3/1/2023 all work with modulo-2

* Algebric – analysis of CRC code



	Polynomial	Degree	# bits
Data word (Y)	$Y(x)$	$k-1$	k
CRC check bits	—	—	r
Generating polynomial	$G(x)$	r	$r+1$
Remainder	$R(x)$	$r-1$	r
Quotient	$Q(x)$	—	—
Dividend ($x^r Y(x)$)	$D(x)$	$k+r-1$	$k+r$
Transmitted code word (T)	$T(x)$	$k+r-1$	$k+r$
Recieved code word (T`)	$T`(x)$	$k+r-1$	$k+r$
Error word	$E(x)$	$k+r-1$	$k+r$

* $G(x) \overline{\big)\, D(x)}$ → quotient $Q(x)$, remainder $R(x)$

$D(x) = Q(x)\,G(x) + R(x)$

$\underbrace{D(x) + R(x)}_{T(x)} = Q(x)\,G(x) \rightarrow T(x) = Q(x)\,G(x)$

ال transmitted word من مضاعفات ال generator

فالمفروض لما تقسم G(x) يطلع ال Remainder = 0 لو مافي أخطاء.

= T(x) is completely divisible by G(x)

* $\underbrace{------+x^r}_{D(x)} + \underbrace{x^{r-1} + -----}_{R(x)}$ على اعتبار إنه : ($D(x) = x^r\, Y(x)$) ل shift ل Y(x)

* الأصل إنه ال Error word (اللي حتأشر عال bits المختلفة) تكون قيمتها 0 لو مافي أخطاء،
أما لو فيها 1 فبتأشر عال bit اللي صار فيها خطأ.

* كيفية اختيار G(x) بحيث مايكون عندي undetected errors :

* undetected errors in CRC :

in $E = T + T'$ → Modulo-2

Polynomial → $E(x) = T(x) + T'(x)$

or $T'(x) = T(x) \oplus E(x)$

لـ نفس الاشي + أو - لأنه Modulo-2

→ At the recieving end,

$$\frac{T'(x)}{G(x)} = \frac{T(x)}{G(x)} + \frac{E(x)}{G(x)}$$

$\therefore$ Remainder $\left[\frac{T'(x)}{G(x)}\right]$ = remainder $\left[\frac{T(x)}{G(x)} + \frac{E(x)}{G(x)}\right]$

$=0$ (لأنه الـ transmit code word صحيحة ما في باقي)

ممكن يكون 0 فممكن يكون فيه في undetected error

لو ما كان 0 فيكون مكتشف الخطأ أصلاً.

→ Remainder $\left[\frac{E(x)}{G(x)}\right]$

$\therefore$ Transmission errors remain undetected when remainder of division of E(x) by G(x) is zero.

Example : T = 1 0 0 0 1 0 0 1 → sender ⇒ in polynomial $T(x) = x^7 + x^3 + 1$

reciever ← T' = 1 1 0 0 0 1 0 1 $T'(x) = x^7 + x^6 + x^2 + 1$

E = 0 1 0 0 1 1 0 0 $E(x) = x^6 + x^3 + x^2$

* أنواع الـ undetected errors *

① Single bit errors :

$E(x) = x^i$

	i	

Position of error bit

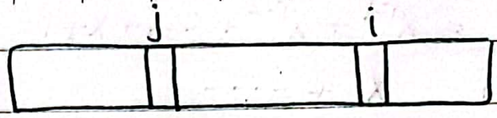
$$\frac{E(x)}{G(x)} = \frac{x^i}{\underbrace{\square + \square + \cdots}}$$

↳ at least two terms (أكثر من x^i يكون عدد الـ terms)

$\therefore$ G(x) has at least two terms, G(x) will never divide a single-term E(x)

* هيك بتضمن إنك عامل detect لكل الـ small errors الممكنة *

② Double bit errors:



$E(x) = x^j + x^i = x^i(x^{j-i}+1) \rightarrow$ "طريقة كتابة للتحويل"

$= \frac{E(x)}{G(x)} = \frac{x^i(x^{j-i}+1)}{G(x)}$

If G(x) contains at least three terms it can be shown that except for very large impractical (غير عملي) value of (j-i) factors $x^i(x^{j-i}+1)$ cannot contain all of the factors of G(x) unless (j-i) is large

لا

$\hookrightarrow \frac{x^i(x^{j-i}+1)}{\square+\square+\square+\cdots}$

∴ all double-bit errors are detected if G(x) contains three terms.

③ odd number of errors: $E(x) = \square+\square+\cdots\square \rightarrow$ odd number of terms

for odd number of errors, E(x) will contain any odd number of terms, e.g. $E(x) = x^7 + x^6 + x^2$. It can be easily shown that any such polynomial (with 3 terms) cannot be divided by x+1 without remainder (i.e. x+1 is not a factor of such polynomial)

$F(x) = (x+1)P(x)$

$F(1) = (1+1)P(1) = 0 \rightarrow$ number of terms odd 1

∴ x+1 cannot be a factor of E(x).

معناها بحط ال generator فيه (x+1) وهيك بعمل detect لكل ال error الموجودة.

* E(x)/G(x), E(x) cannot be divided by (x+1)

5/1/2023

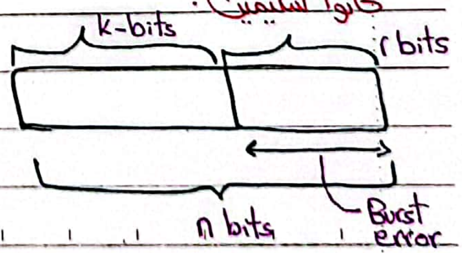
④ Burst error of length ≤ r

Ⓐ Assume that burst error of length "r" affects the check bit field.

لو ال i bit و j bit صار فيهم مشكلة، كل ال bits اللي بينهم بنعتبر إنه فيهم مشاكل حتى لو كانوا سليمين.

$\rightarrow E(x) = x^{r-1} + x^{r-2} + \cdots + x + \underline{\underline{1}}$ ← r-bits affected

لأنه في عندك bit خاربة في أقصى اليمين.



→ what is the degree of G(x)?

r+1 of bits → degree of G(x) is r

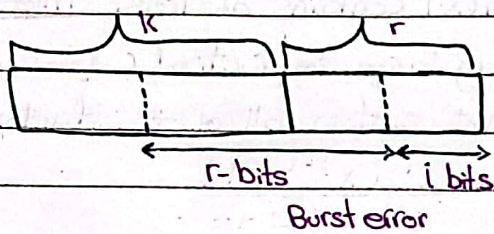
لـ أقلب لـ من r+1

$$\frac{E(x)}{G(x)} = \frac{x^{r-1} + x^{r-2} + \cdots + x + 1}{x^{r} + \cdots}$$

↳ will never divide $E(x)$.

Ⓑ Assume that burst error occurs any where else in the Codeword.

$E(x) = x^{i}(x^{r-1} + x^{r-2} + \cdots + x + 1)$



$\therefore \dfrac{E(x)}{G(x)} = \dfrac{x^{i}(x^{r-1} + x^{r-2} + \cdots + x + 1)}{}$

No x or x^{i} as factor of $G(x)$

∴ all burst errors of length r irrespective (بغض النظر) of their location will be detected.

⑤ Burst error of length $r+1$:

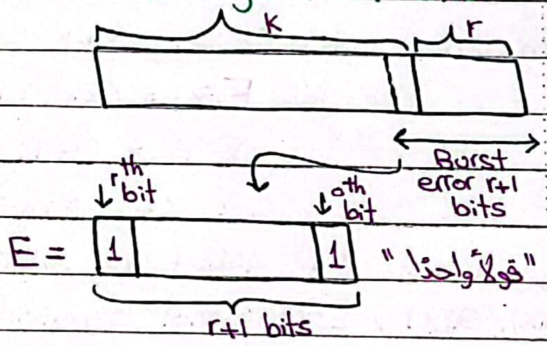
Ⓐ Assume burst error of length $r+1$ affects the right most bits.

$$\frac{E(x)}{G(x)} = x^{r} + \underbrace{\cdots}_{r-1 \text{ of terms in between}} + 1$$

If $G(x) = E(x)$

↳ (matches term by term)

* only when $G(x) = E(x)$, $G(x)$ will divide $E(x)$.

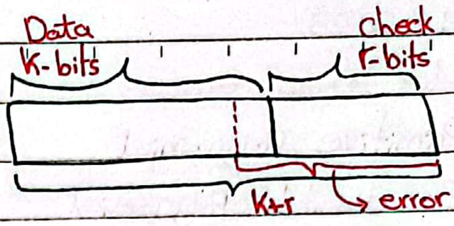


* Only one out of 2^{r-1} combinations will not be detected.

Probability of not detecting $(r+1)$-bit long burst error is $\dfrac{1}{2^{r-1}}$

Ⓑ The result in Ⓐ is true for all locations.

⑥ Burst error of length > r+1



Ⓐ In this when the burst error Covers the whole code word,

E(X) has a degree of K+r-1

$E(x) = \sum_{i=0}^{K+r-1} a_i x^i$, $(a_0, a_1, a_2, \text{-----}, a_{K+r-1})$

→ There are 2^{K+r} Possible combination.

→ Possible error combination $2^{K+r} - 1$

"كله 0 بكون مستثنى لأنه لا يعتبر error."

* $\frac{E(X)}{G(X)}$, Now, for errors to remain undetected. G(X) Should be one of the factors of E(X), we can write:

* remember that G(X) has a degree of "r" we can write :

$$E(X) = G(X) \sum_{i=0}^{K-1} a_i X^i$$

G(X) مضاعفات

The number of Combination $(a_0, a_1, \text{----}, a_{K-1})$ $\underline{\underline{2^K}}$

$\underline{\underline{2^K - 1}}$ error Polynomials. ("number of error combinations)

8/1/2023

* بناء على اللي فوق * → Probability of undetected error :

$$\frac{2^K - 1}{2^{K+r} - 1} \approx \frac{2^K}{2^{K+r}} = \frac{1}{2^r} \longrightarrow \text{for large K}$$

→ Probability of detected error : $1 - \frac{1}{2^r}$

* Summary *

→ Single error : 100 %

→ Double error : 100 %

→ odd numbers of errors : 100 %

→ Burst error of length < r+1 : 100 %

→ Burst error = r+1 : $1 - \left(\frac{1}{2}\right)^{r-1}$

→ Burst error > r+1 : $1 - \left(\frac{1}{2}\right)^{r}$

(Probability of detected errors)

* لهيك الـ CRC منيح لأنه سهل ال implementation ولأنه بعمل detect جيد ال error.

Note

* **If the generator has more than one term and the coefficient of x^0 is 1, all single errors can be caught.**

مش صحيح مطلقاً 100%، المهم يكون at least two terms.

Example 10.15

Which of the following g(x) values guarantees that a single-bit error is caught? For each case, what is the error that cannot be caught?

a. $x+1$ ***b.*** x^3 ***c.*** 1

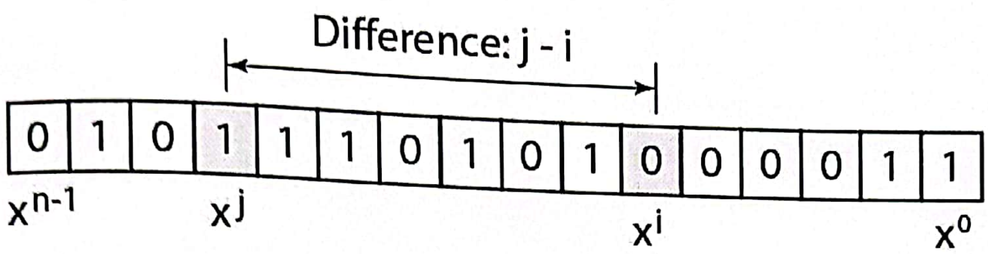
Solution

a. No x^i can be divisible by $x + 1$. Any single-bit error can be caught.

b. If i is equal to or greater than 3, x^i is divisible by g(x). All single-bit errors in positions 1 to 3 are caught.

c. All values of i make x^i divisible by g(x). No single-bit error can be caught. This g(x) is useless.

Figure 10.23 *Representation of two isolated single-bit errors using polynomials*



10.73

Note

If a generator cannot divide $x^t + 1$ (t between 0 and n – 1), then all isolated double errors can be detected.

10.74

Example 10.16

Find the status of the following generators related to two isolated, single-bit errors.

a. $x + 1$ ***b.*** $x^4 + 1$ ***c.*** $x^7 + x^6 + 1$ ***d.*** $x^{15} + x^{14} + 1$

Solution

a. This is a very poor choice for a generator. Any two errors next to each other cannot be detected.

b. This generator cannot detect two errors that are four positions apart.

c. This is a good choice for this purpose.

d. This polynomial cannot divide $x^t + 1$ ***if t is less than 32,768. A codeword with two isolated errors up to 32,768 bits apart can be detected by this generator.***

احنا حليناه أدق وبالزبط بالدفتر

Note

A generator that contains a factor of *x* + 1 can detect all odd-numbered errors.

Note

- All burst errors with $L \leq r$ will be detected.
- All burst errors with $L = r + 1$ will be detected with probability $1 - (1/2)^{r-1}$.
- All burst errors with $L > r + 1$ will be detected with probability $1 - (1/2)^{r}$.

10.77

Example 10.17

Find the suitability of the following generators in relation to burst errors of different lengths.

a. $x^6 + 1$ *b.* $x^{18} + x^7 + x + 1$ *c.* $x^{32} + x^{23} + x^7 + 1$

Solution

a. This generator can detect all burst errors with a length less than or equal to 6 bits; 3 out of 100 burst errors with length 7 will slip by; 16 out of 1000 burst errors of length 8 or more will slip by.

10.78

Example 10.17 (continued)

b. This generator can detect all burst errors with a length less than or equal to 18 bits; 8 out of 1 million burst errors with length 19 will slip by; 4 out of 1 million burst errors of length 20 or more will slip by.

c. This generator can detect all burst errors with a length less than or equal to 32 bits; 5 out of 10 billion burst errors with length 33 will slip by; 3 out of 10 billion burst errors of length 34 or more will slip by.

Note

A good polynomial generator needs to have the following characteristics:

1. **It should have at least two terms.**
2. **The coefficient of the term x^0 should be 1.**
3. **It should not divide $x^t + 1$, for t between 2 and $n - 1$.**
4. **It should have the factor $x + 1$.**

10/1/2023

Slide 10.72 :

→ Primitive Polynomial

→ Irreducible Polynomial

① Definition : a Polynomial that has no factors other than 1 and itself is called irreducible polynomial. (in analogy with prime number)
(زي العدد الأولي تقريباً)

② Definition : An irreducible polynomial $P(x)$ of degree N is Primitive if $P(X)$ is a factor of $X^M + 1$ for $M = 2^N - 1$ an NO SMALLER M

بيستفيد منها في المعلومة إنه إذا كانت N كبيرة بقدر أضمن أنه كل شيء أقل من هذا الـ N
مارح يقسم عليه وبالتالي كل هاي الـ errors رح تكون detected.

* $1 \le M \le 2^N - 2$ → $\dfrac{X^M + 1}{P(X)}$ Does not divide (Remainder)	* $M = 2^N - 1$ → $\dfrac{X^M + 1}{P(X)}$ Divide with no remainder

Example : $P(x) = X^{15} + X^{14} + 1$ ⟶ Irreducible and Primitive Polynomial (Slide 10.75)

↳ هلأ إذا بعوض 1 محل X بعطيني 0 ؟ X

→ Solution :

$1 \le M \le 32766 \leftarrow 1 \le M \le 2^{15} - 2$

$\dfrac{X^M + 1}{X^{15} + X^{14} + 1}$ with remainder

$M = 2^N - 1 = 2^{15} - 1 = 32767$

→ $\dfrac{X^M + 1}{X^{15} + X^{14} + 1}$ Divide without remainder

$E(X) = X^j + X^i = X^i \underbrace{(X^{j-i} + 1)}_{X+1}$

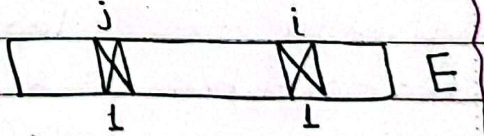


Table 10.7 *Standard polynomials* CRC مستخدمة بال

Name	*Polynomial*	*Application*
CRC-8	x^8+x^2+x+1	ATM header
CRC-10	$x^{10}+x^9+x^5+x^4+x^2+1$	ATM AAL
CRC-16	$x^{16}+x^{12}+x^5+1$	HDLC
CRC-32	$x^{32}+x^{26}+x^{23}+x^{22}+x^{16}+x^{12}+x^{11}+x^{10}+x^8+x^7+x^5+x^4+x^2+x+1$	LANs (wifi)

يستخدم بال error detection وممكن نستخدمه بال correction

* في تشابه بينه وبين ال TCP.

* في طريقتين لتعديل الأخطاء بال Computer Networks وال data communication:

الأولى اذا اكتشفت الخطأ ووين ال bit موجودة بتعمل Correct لل bit (forward error correction).

الثانية اذا اكتشفت الخطأ بتبعث للمرسل طلب بإعادة الإرسال (Automated repeat request).

10-5 CHECKSUM

يتشابه مع ال CRC بالبساطة.

The last error detection method we discuss here is called the checksum. The checksum is used in the Internet by several protocols although not at the data link layer. However, we briefly discuss it here to complete our discussion on error checking

Topics discussed in this section:

Idea
One's Complement
Internet Checksum

Example 10.18

Suppose our data is a list of five 4-bit numbers that we want to send to a destination. In addition to sending these numbers, we send the sum of the numbers. For example, if the set of numbers is (7, 11, 12, 0, 6), we send (7, 11, 12, 0, 6, 36), where 36 is the sum of the original numbers. The receiver adds the five numbers and compares the result with the sum. If the two are the same, the receiver assumes no error, accepts the five numbers, and discards the sum. Otherwise, there is an error somewhere and the data are not accepted.

Example 10.19

We can make the job of the receiver easier if we send the negative (complement) of the sum, called the checksum. In this case, we send (7, 11, 12, 0, 6, −36). The receiver can add all the numbers received (including the checksum). If the result is 0, it assumes no error; otherwise, there is an error.

* Slide 10. 82 :

" يوضح مبدأ عمل ال CheckSum "

Example :

Original Data

قسمناها لمجموعات ← | 10011001 | 11100010 | 00100100 | 10000100 |

4 أقسام ← K = 4, M = 8 → في كل قسم

Sender

10011001
11100010 +

(1) 01111011
1 ← ال Carry نزلناه وجمعنا +

01111100
00100100 +

10100000
10000100 +

(1) 00100100
1 ← ال Carry نزلناه وجمعنا +

Sum: 00100101

CheckSum: 11011010
← 1's complement (كأنه أخذنا سالب ال Sum)

وهنا بنبعته مع ال Packet الرئيسية.

reciever

10011001
11100010 +

(1) 01111011
1 +

01111100
00100100 +

10100000
10000100 +

(1) 00100100
1 +

00100101
11011010 +

Sum: 11111111

Complement: 00000000

Conclusion: Accept Data.

ل ال reciever لو طلع معه اشي غير ال 0 بعرف إنه عنده error.

Example 10.20

How can we represent the number 21 in one's complement arithmetic using only four bits?

Solution

The number 21 in binary is 10101 (it needs five bits). We can wrap the leftmost bit and add it to the four rightmost bits. We have (0101 + 1) = 0110 or 6.

10.85

Example 10.21

How can we represent the number −6 in one's complement arithmetic using only four bits?

Solution

In one's complement arithmetic, the negative or complement of a number is found by inverting all bits. Positive 6 is 0110; negative 6 is 1001. If we consider only unsigned numbers, this is 9. In other words, the complement of 6 is 9. Another way to find the complement of a number in one's complement arithmetic is to subtract the number from $2^n - 1$ (16 − 1 in this case).

10.86

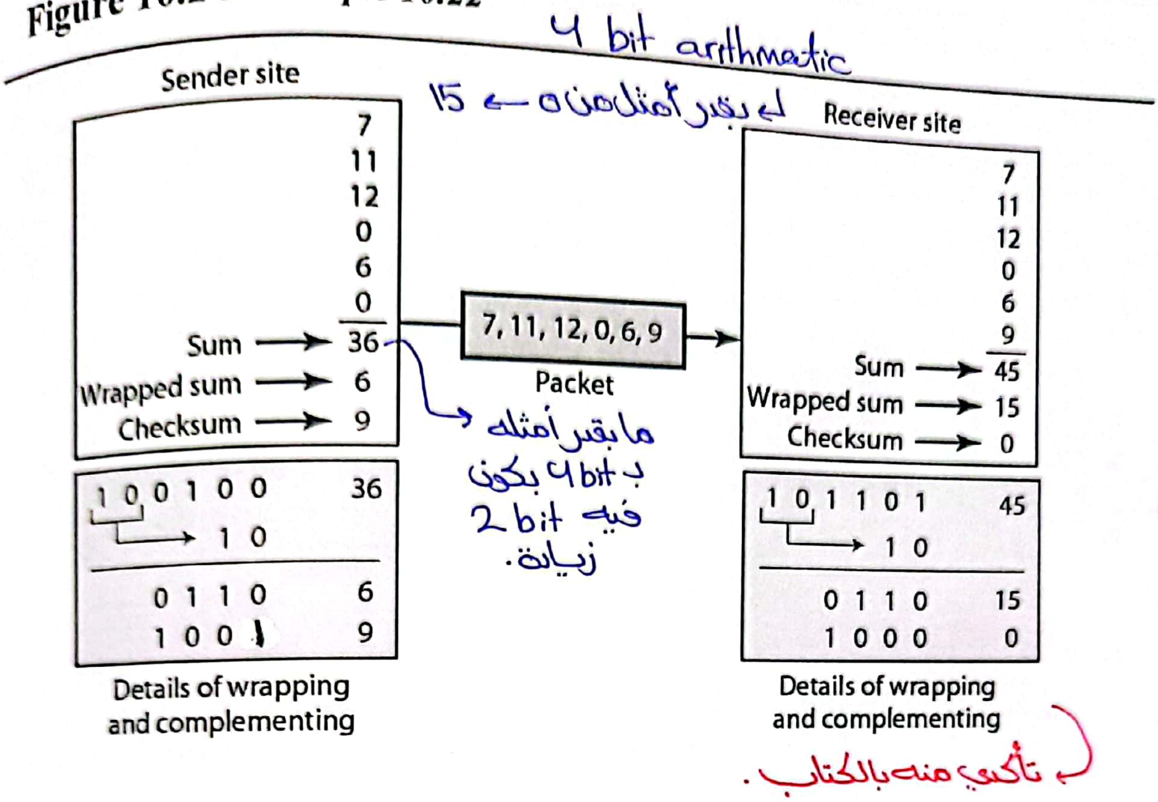
Example 10.22

Let us redo Exercise 10.19 using one's complement arithmetic. Figure 10.24 shows the process at the sender and at the receiver. The sender initializes the checksum to 0 and adds all data items and the checksum (the checksum is considered as one data item and is shown in color). The result is 36. However, 36 cannot be expressed in 4 bits. The extra two bits are wrapped and added with the sum to create the wrapped sum value 6. In the figure, we have shown the details in binary. The sum is then complemented, resulting in the checksum value 9 (15 − 6 = 9). The sender now sends six data items to the receiver including the checksum 9.

Example 10.22 (continued)

The receiver follows the same procedure as the sender. It adds all data items (including the checksum); the result is 45. The sum is wrapped and becomes 15. The wrapped sum is complemented and becomes 0. Since the value of the checksum is 0, this means that the data is not corrupted. The receiver drops the checksum and keeps the other data items. If the checksum is not zero, the entire packet is dropped.

Figure 10.24 *Example 10.22*



Note

Sender site:
1. The message is divided into 16-bit words.
2. The value of the checksum word is set to 0.
3. All words including the checksum are added using one's complement addition.
4. The sum is complemented and becomes the checksum.
5. The checksum is sent with the data.

Note

Receiver site:

1. The message (including checksum) is divided into 16-bit words.
2. All words are added using one's complement addition.

← جمع كل شي بما فيها الChecksum

3. The sum is complemented and becomes the new checksum.
4. If the value of checksum is 0, the message is accepted; otherwise, it is rejected.

10.91

Example 10.23

Let us calculate the checksum for a text of 8 characters ("Forouzan"). The text needs to be divided into 2-byte (16-bit) words. We use ASCII (see Appendix A) to change each byte to a 2-digit hexadecimal number. For example, F is represented as 0x46 and o is represented as 0x6F. Figure 10.25 shows how the checksum is calculated at the sender and receiver sites. In part a of the figure, the value of partial sum for the first column is 0x36. We keep the rightmost digit (6) and insert the leftmost digit (3) as the carry in the second column. The process is repeated for each column. Note that if there is any corruption, the checksum recalculated by the receiver is not all 0s. We leave this an exercise.

10.92

Figure 10.25 *Example 10.23*

1	0	1	3	Carries
4	6	6	F	(Fo)
7	2	6	F	(ro)
7	5	7	A	(uz)
6	1	6	E	(an)
0	0	0	0	Checksum (initial)
8	F	C	6	Sum (partial)
			1	
8	F	C	7	Sum
7	0	3	8	Checksum (to send)

a. Checksum at the sender site

1	0	1	3	Carries
4	6	6	F	(Fo)
7	2	6	7	(ro)
7	5	7	A	(uz)
6	1	6	E	(an)
7	0	3	8	Checksum (received)
F	F	F	E	Sum (partial)
			1	
8	F	C	7	Sum
0	0	0	0	Checksum (new)

a. Checksum at the receiver site

" تأكدي منها من الكتاب "