Data Communications and Networking Fourth Edition

Forouzan

Chapter 1

Introduction

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Impact of Data Communications and Networking

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

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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

لـ الاختلاف

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Concepts of Data Communication

- ➢Components
- ➢Data Representation
- ➢Direction of Data Flow

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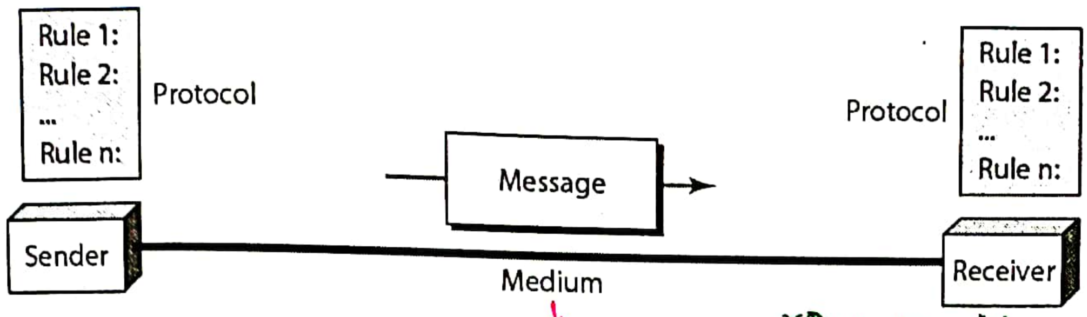
Figure 1.1 *Five components of data communication*

*مرات ممكن نضحي بال loss Packets بحساب ال delay تكون منيحة.



ناخذ فيها ال Bandwidth

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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

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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 link
  - Example: telephone network

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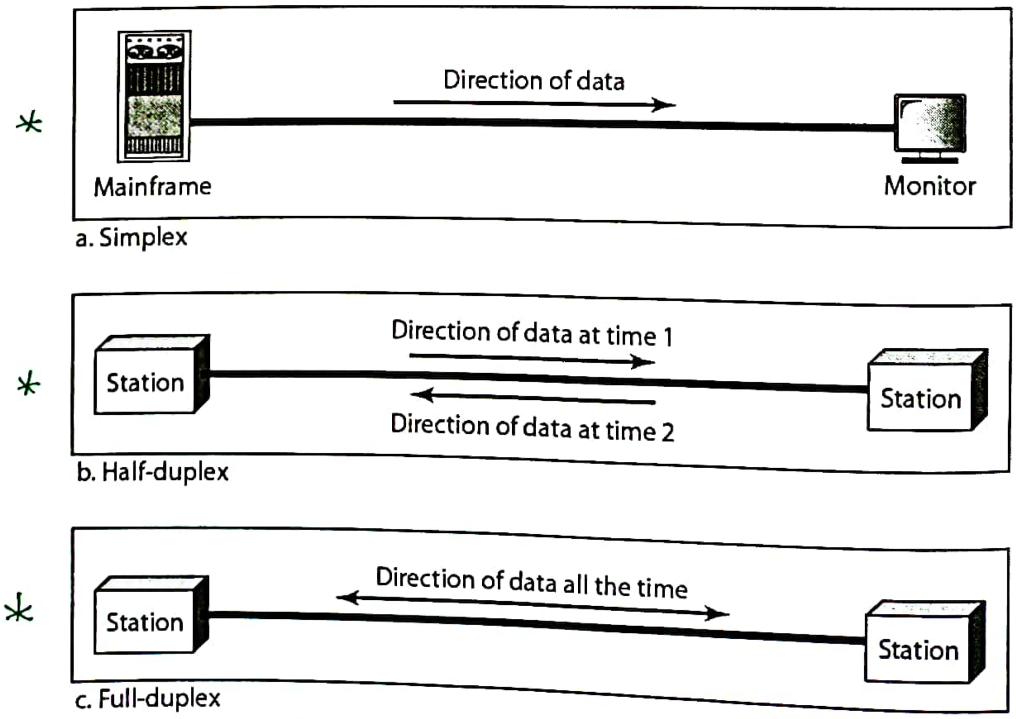
Figure 1.2 *Data flow (simplex, half-duplex, and full-duplex)*



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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

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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 جغرافياً الـ PAN وأكبر وحدة WAN

Wide Area Network

Personal Area Network

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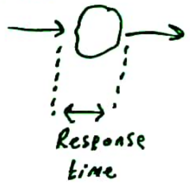
* Propagation delay

$T_d = \dfrac{\text{distance}}{\text{Speed} (\approx \text{Speed of light})}$

Network Criteria

- Performance
  - Bit rate vs. packet rate vs. throughput
  - Transit time (or propagation delay) ← الزمن اللازم عشان ابعت الداتا من Sender → Receiver
  - Response time (round trip delay) → الزمن ما لحظة دخولك ال System ل لحظة خروجك من ال System
- Reliability
  - Bit error rate vs. packet error rate
  - Error detection and correction
  - Link failure recovery time
  - Robustness to disasters → المتانة
    خطة لل backup او خطة بديلة عشان ما تروح كل ال Network
- Security
  - Protect against unauthorized access
  - Protect against alteration
  - Validity of sender



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Types of Network Connections

- Point-to-point link
  - * Dedicated link between two nodes
  - * Full link capacity → بتاخد full لما يكون بين 2nodes لما اطلع برا لا.
- Multi-point link
  - * Shared link among many nodes
  - * Spatially shared: simultaneous access
  - * Temporally shared: timely shared

→ كلما زاد عدد ال Users زيد الضغط بيزيد عال link وبالتالي ممكن تصير مشاكل.

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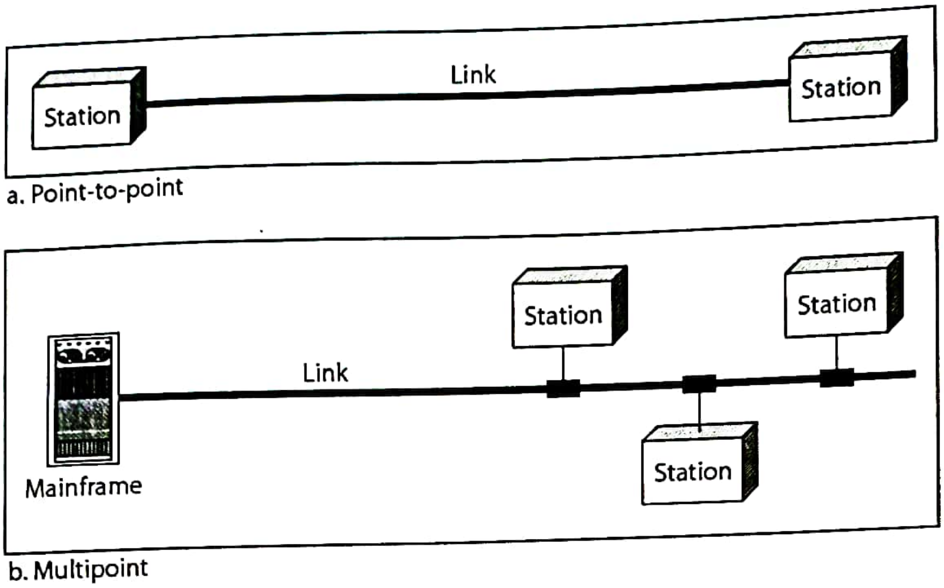
Figure 1.3 *Types of connections: point-to-point and multipoint*



اكثر من Station بينها
تحجز هاد ال link.
لما واحد يشتغل ما
تقدروا الباقين يبعتوا.

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Figure 1.4 *Categories of topology*

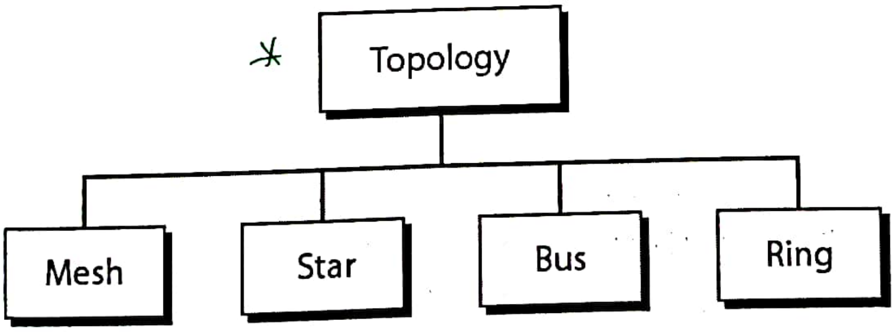

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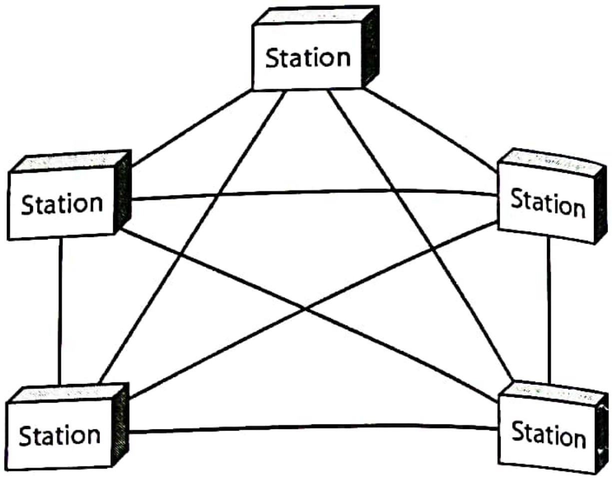
Figure 1.5 *A fully connected mesh topology (five devices)*

- 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 → كل وحدة عليها اكثر من link.



* كلو مشبوك مع كل ال stations
* مش دايما منيح لانه في كثير links

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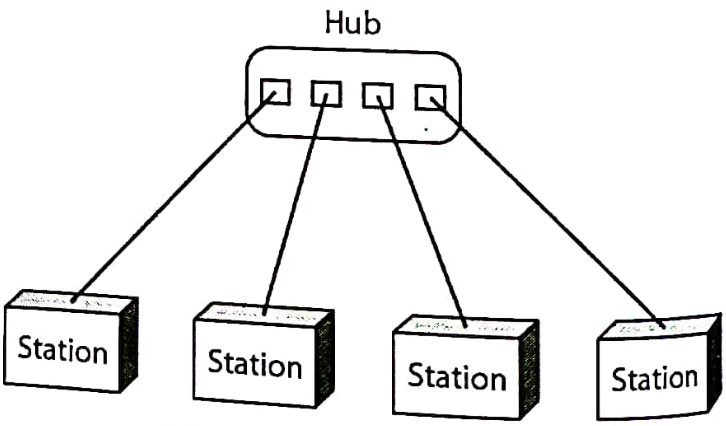
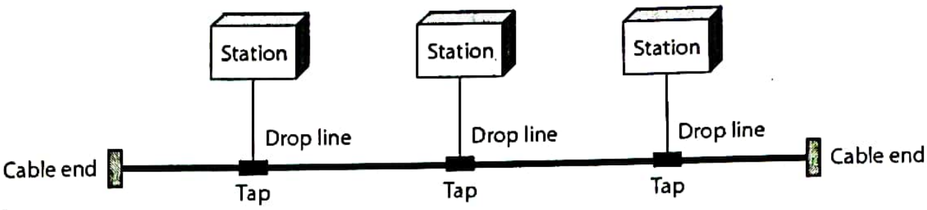
Figure 1.6 *A star topology connecting four stations*

- 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)



* ال Stations بيعتوا لبعض عن طريق ال Hub.
ال Security سيئة.

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.
هنا لا Star topology
(Hub)
(ال Slide الي قبل)

* ال bus محجوز للجميع.
* اذا صارت مشكلة بال bus كلهم بيتعطلوا وما بيقدروا يحددوا كثير وصعب اكتشف وين المشكلة.

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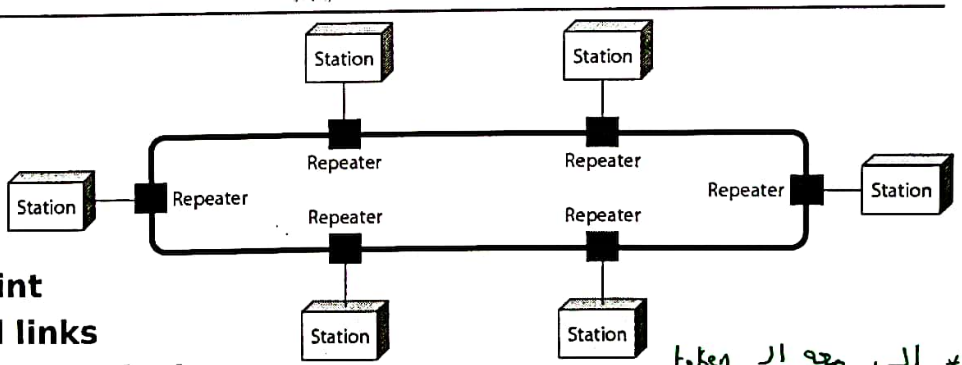
Figure 1.8 *A ring topology connecting six stations*



- 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

* الي معه ال token هو الي بيبعث.
* اي خلل بيصير كلهم بيتعطلوا.

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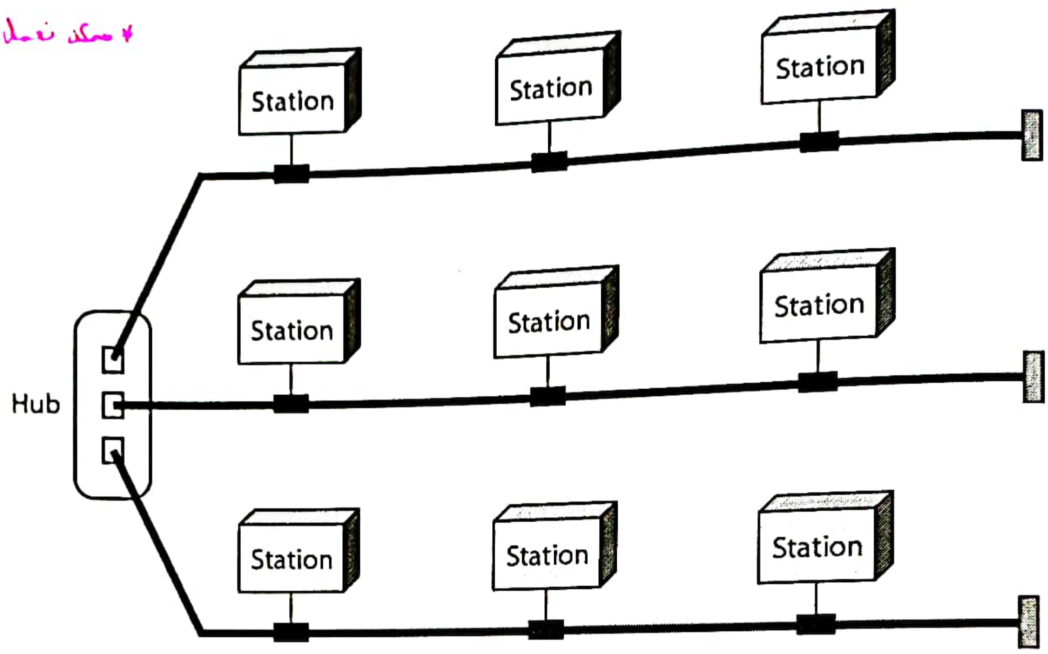
Figure 1.9 *A hybrid topology: a star backbone with three bus networks*

* ممكن نعمل مكس بين ال topology



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Categories of Networks

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

Km كبيرة

كمستوى المدينة

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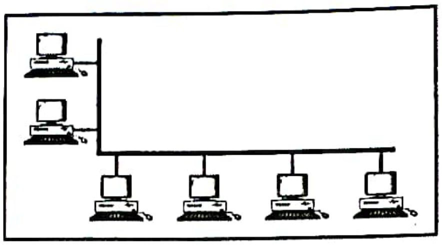
LAN



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

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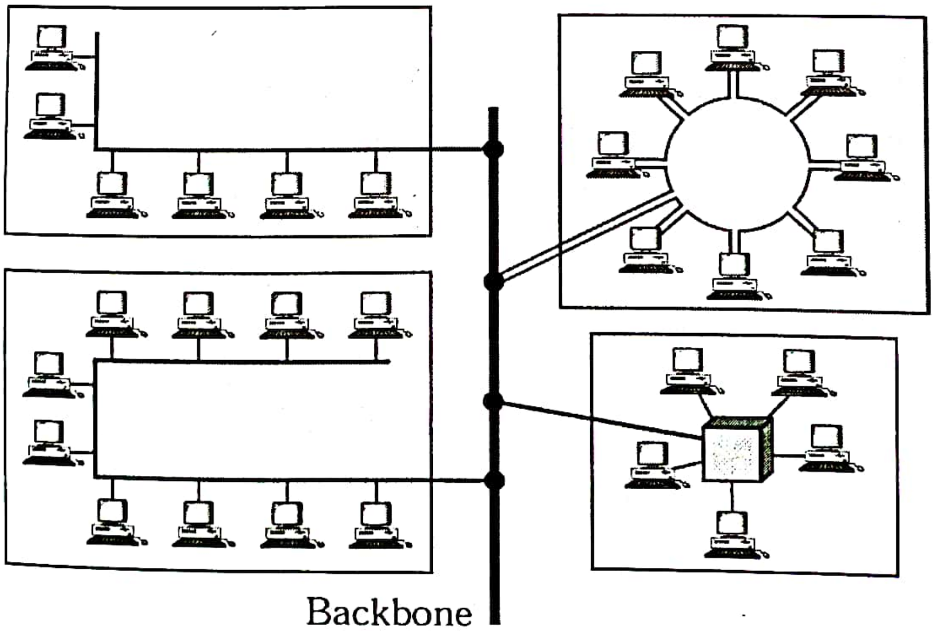
LAN (Continued)



b. Multiple-building LAN

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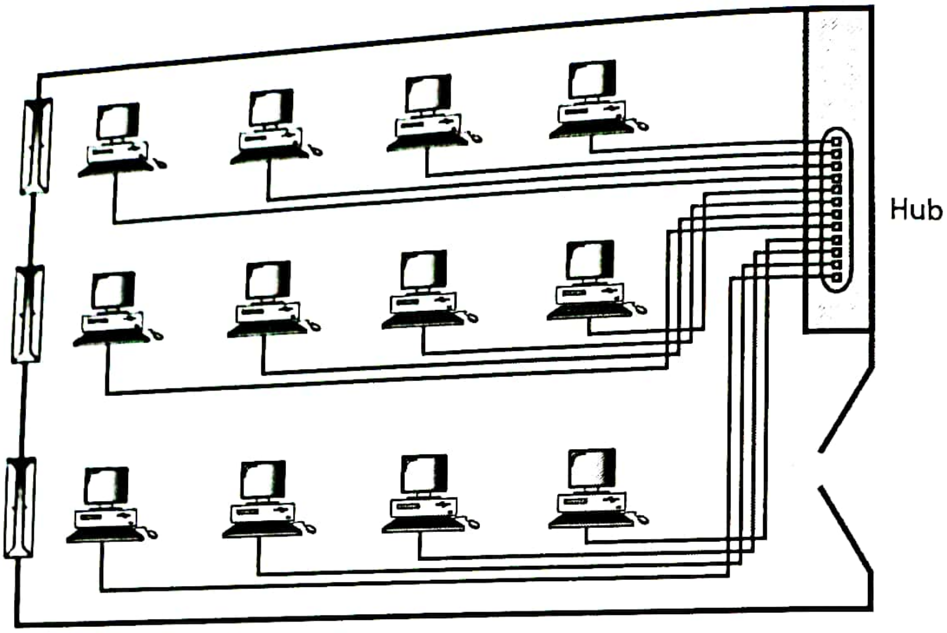
Figure 1.10 *An isolated LAN connecting 12 computers to a hub in a closet*



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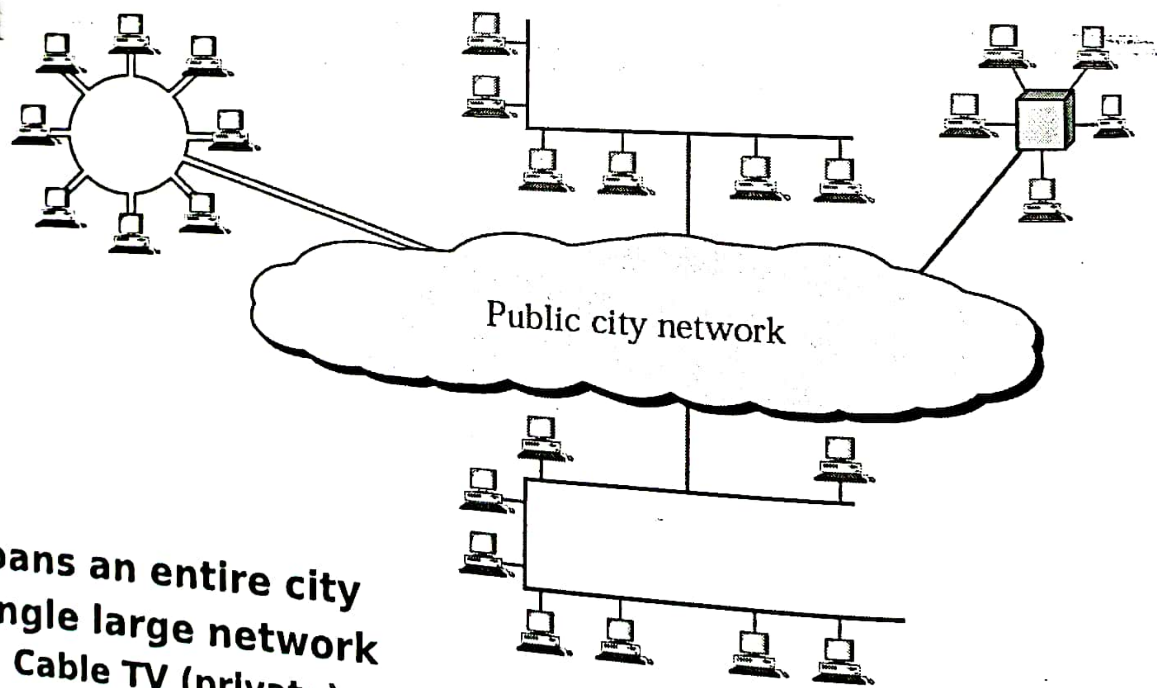
MAN



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

Comm:

2/3

Network Criteria:

* Propagation delay

$$T_d = \frac{\text{distance}}{\text{Speed } (\approx \text{Speed of light})}$$

* Transmission delay. (الزمن اللازم عشان ارسل One packet)
* Processing delay.
* Queueing delay

مجموعهم كلهم One-way delay

One-Way delay

$= T_d + T_t + T_p + \boxed{T_q}$

الكراشي ممكن يأثر

$T_t = \frac{P}{R}$

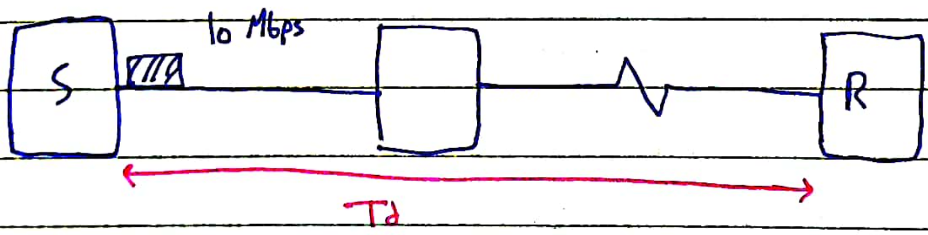
* ما بنقدر نعرف مين العامل المحدد عندي بختلف حسب شو مستخدم عندي مثلا Fiber ...

$\frac{10 \text{ Mb}}{1 \text{ S}}$

$\frac{1}{10 \times 10^6} = 0.1 \ \mu s/\text{bit}$

bit duration

(time needed to transmit one bit)



الزمن اللازم عشان ابعت packets 1 بدي Trans. الـ * 3

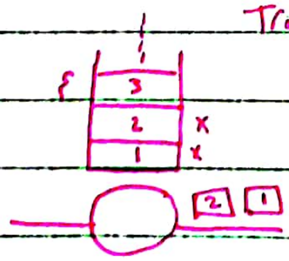
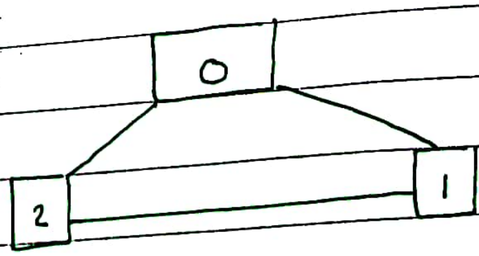


Figure 1.5



$i : 0, 1, 2$

L : total # of links

L_i : # of links on the i^{th} node.

N : total # of nodes

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

$$L = \sum_{i=0}^{N-1} L_i = \sum_{i=0}^{N-1} N-(i+1)$$

$$= \sum_{k=1}^{N} (N-k)$$

$$= \sum_{k=1}^{N} N - \sum_{k=1}^{N} k$$

$$= \underbrace{N^2 - \frac{N(N+1)}{2}}_{1+2+\cdots+N} \rightarrow = \frac{N(N-1)}{2}$$

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

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

$$2S_N = (1+N)+(2+N-1)+(3+N-2)+\cdots$$

$$2S_N = N(N+1)$$

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Chapter 3

Data and Signals

" Physical Layer "

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Note

*

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

ممكن تكون على شكل ضوء ممكن تكون
اشارة ممكن باد Copper wire .

* عشان نبعت الداتا لازم
نحولها لـ electromagnetic
Signals .

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3-1 ANALOG AND DIGITAL

لـ منفصلة. عدد معيناً ومحدود.

***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.*

اذا كان عندي N وبدي احولها لـ Cont.
بخلي N تروح لل ∞ .

Topics discussed in this section:

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

3.3

*الصوت بتحول لـ electromagnetic Signal عن طريق إنه صوتك بتحول لـ Vibration بال Voltage وبتمثل كـ 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.

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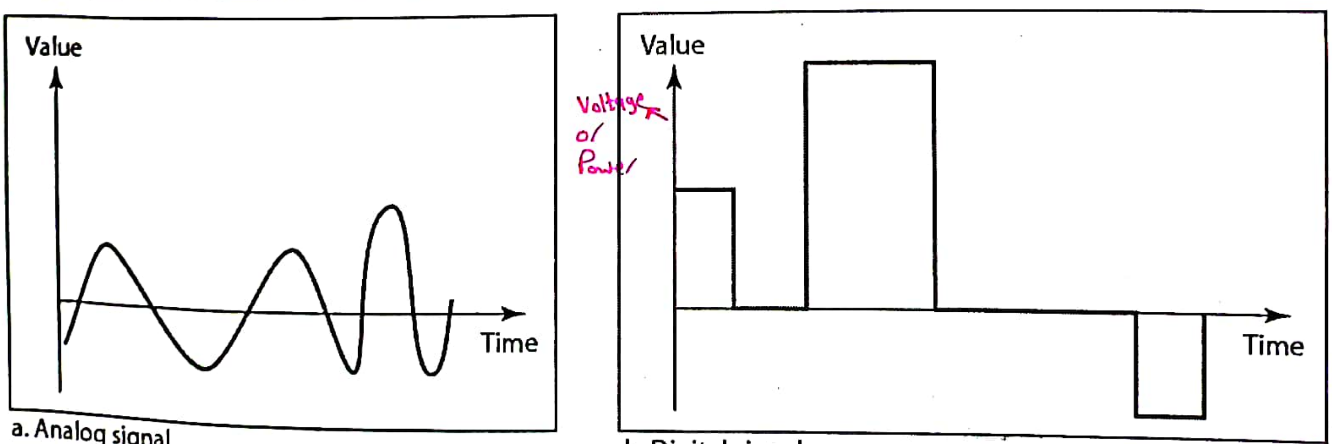
Note

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.

عدد لانهائي من النقاط.

3.5

Figure 3.1 *Comparison of analog and digital signals*



a. Analog signal

b. Digital signal

- Continuous
- Infinite number of possible values

- Discrete
- Finite number of possible values

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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
  - Completes a pattern within a measurable certain time frame called a period
  - Repeats the pattern over subsequent identical periods called cycles

كل Cycle بضل أعيدها ال Pattern. * عدد محدد من ال Freq.

- Aperiodic signal → ما بتعيد حالها
  - Changes with no patterns or cycles over time
  - Both analog and digital signals can be either periodic or aperiodic

ممكن يكون عندي analog, digital بتعيد حالها بس عدد النقاط بكل وحدة منها ∞. * عدد لا نهائي من ال Freq.

Note

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

لأنه ال Bandwidth تاعها محدود.

لنقل 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.*

Multiple Sin Wave

Topics discussed in this section:

Sine Wave
Wavelength
Time and Frequency Domain
Composite Signals
Bandwidth

3.8

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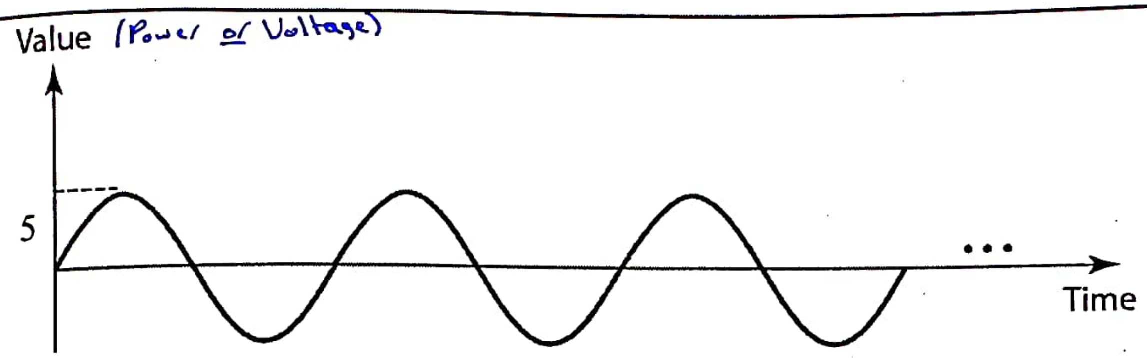
Figure 3.2 *A sine wave*



$$\underset{\text{Instantaneous Amplitude}}{s(t)} = \underset{\text{Peak Amplitude}}{A} \sin\left(2\pi \underset{\substack{\text{Frequency}\\ \frac{1}{t}}}{f}\,\underset{\text{Sec}}{t} + \underset{\text{Phase}}{\varphi}\right)$$

Instantaneous Amplitude

Peak Amplitude

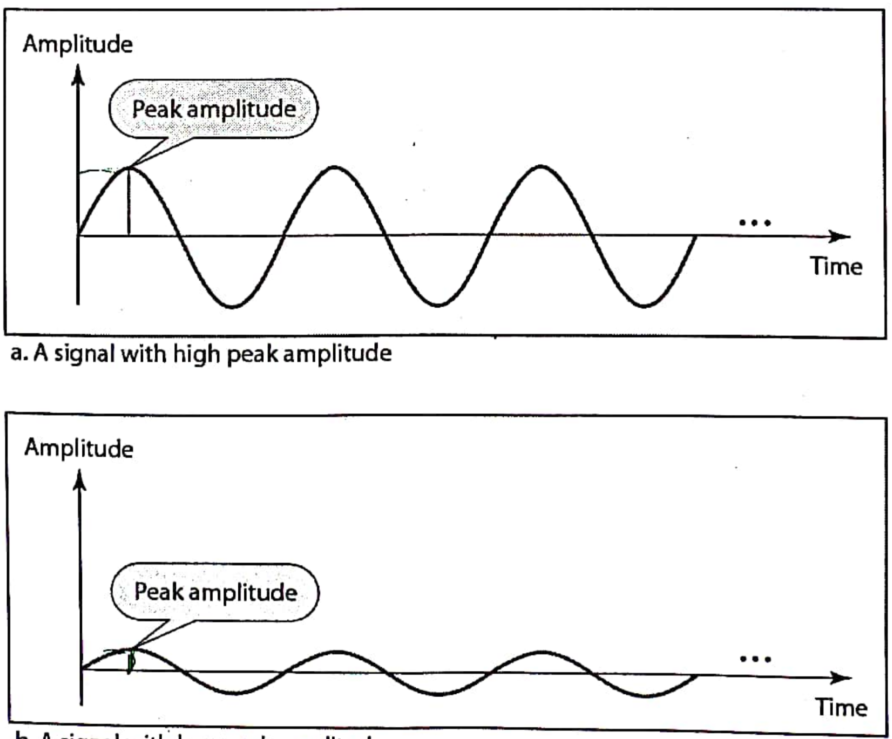
Frequency علاقة عكسية مع الـPeriod

Phase

Characteristics of the sine wave

3.9

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

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Note

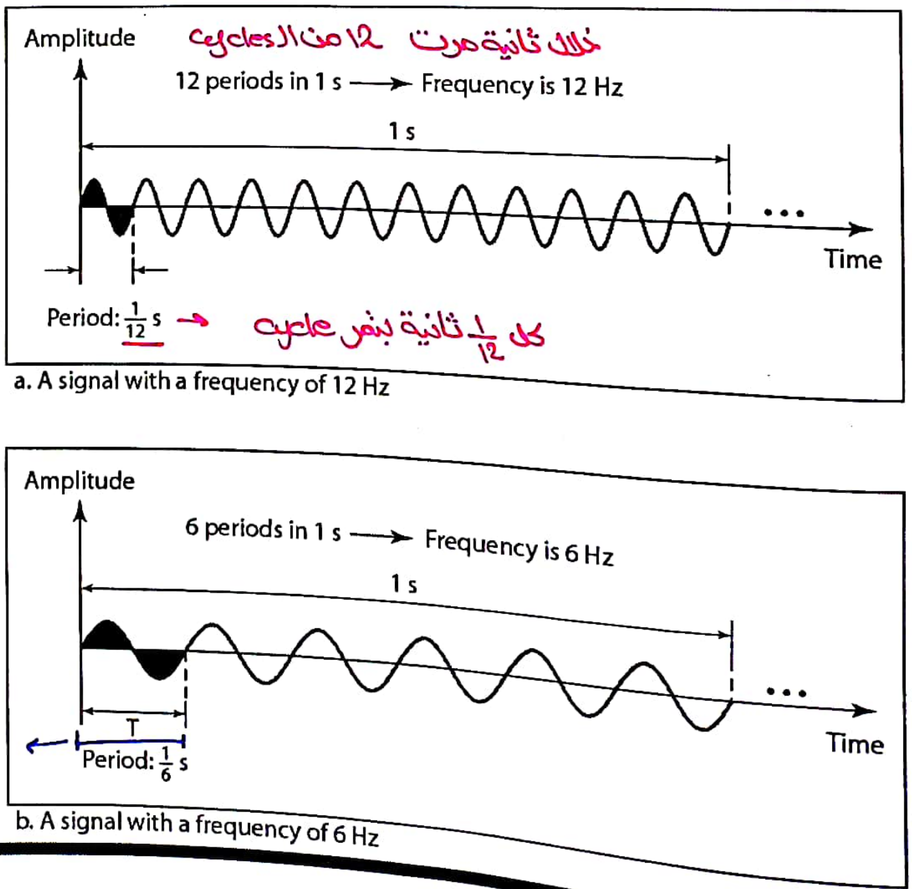
Frequency and period are the inverse (or the reciprocals) of each other.

$$f = \frac{1}{T} \quad \text{and} \quad T = \frac{1}{f}$$

Hz   Second

3.11

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



a. A signal with a frequency of 12 Hz

b. A signal with a frequency of 6 Hz

Cycle
وحدة .

3.12

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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

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:

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

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:.

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

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).

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

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

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Note

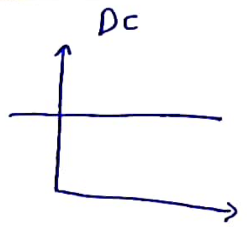
Frequency is the rate of change with respect to time.

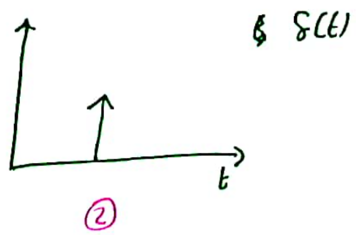
كم cycle بمر كذلك every Time .

Change in a short span of time means high frequency.

Change over a long span of time means low frequency.

3.17





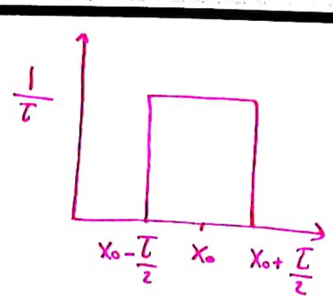
Note

If a signal does not change at all, its frequency is zero (DC Signal). → معطى خط ثابت

If a signal changes instantaneously, its frequency is infinite (only in theory). → لعطي (على الفهم) → نظريًا موجودة

$\tau \to 0$

لما تكون هيك الرسمة يتحول زي رسمة ②



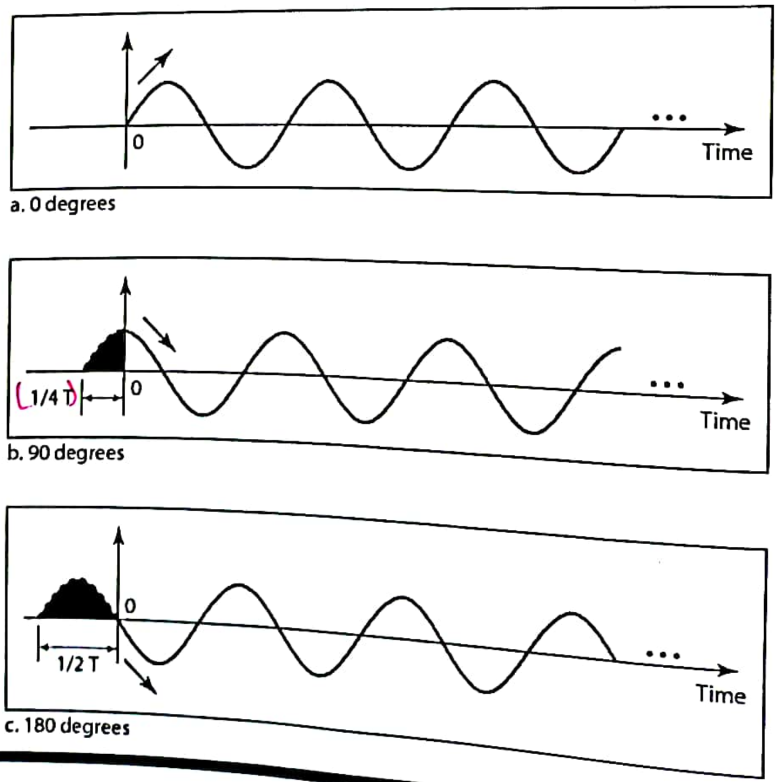
3.18

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 "



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

$S(f)$

$\theta = \omega t$

$= 2\pi f t$

$= \frac{2\pi}{T} t$

$t = \frac{T}{2\pi} \theta$

ازاحة لليسار او يمين

$\sin(\omega t - \varphi)$

$\sin(\omega t + \varphi)$

$\underline{\omega} = 2\pi \underline{f}$ Hz

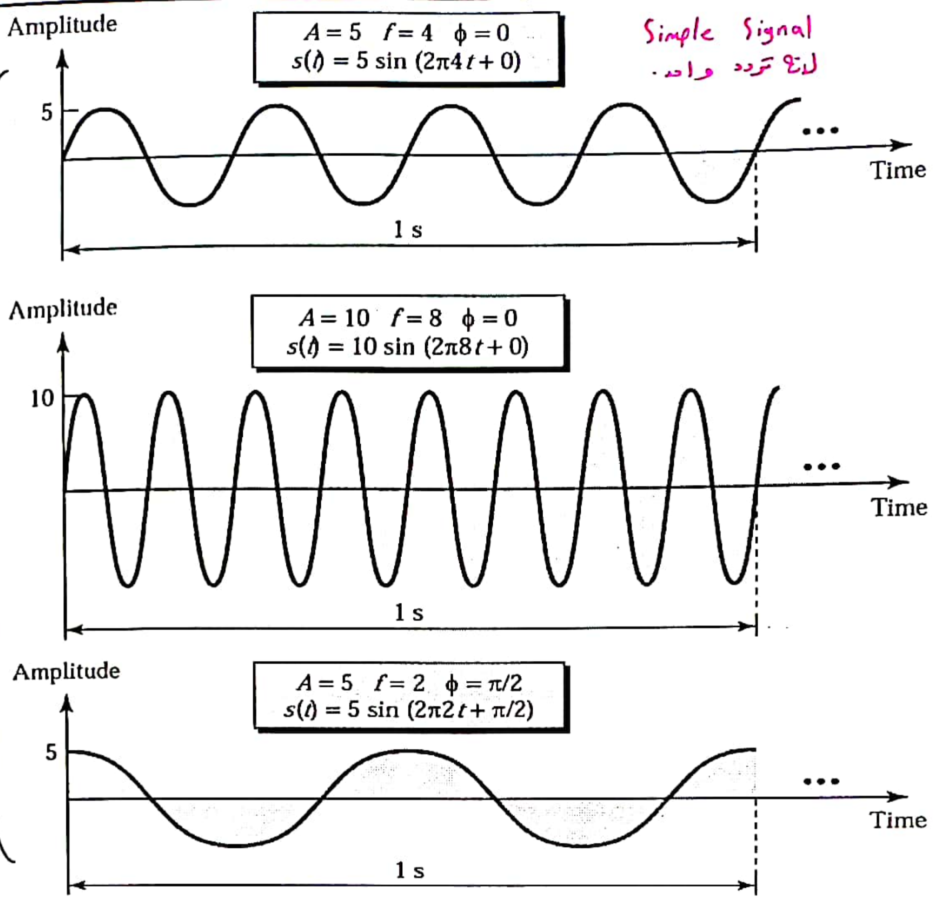
$\underline{rad}$

اللفة الكاملة $2\pi f$

3.20

Simple Signal
لإنو تردد واحد.

Time - Domai n Plots



Example 3.6

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

كل $2\pi \leftarrow 360^\circ$

عشان احول من ° لـ rad

بضرب ب $\frac{2\pi}{360}$

Solution

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

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

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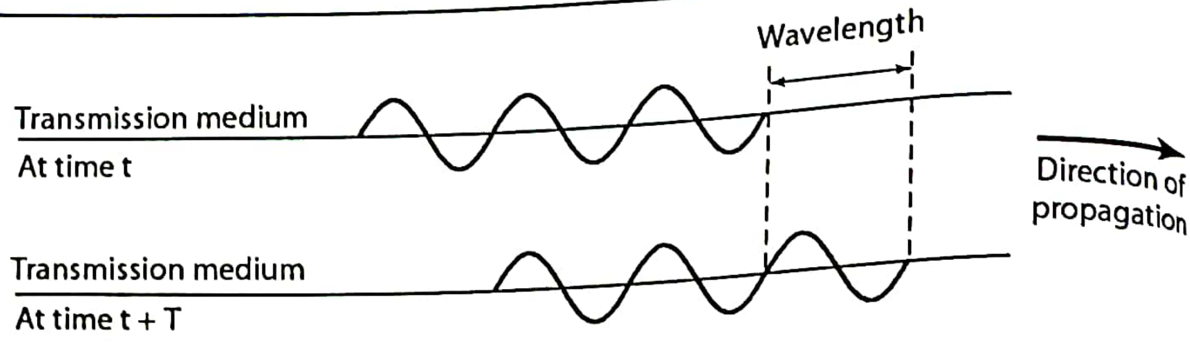
Figure 3.6 *Wavelength and period*



- 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

* السؤال تجود لو السرعة Constant اول.

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

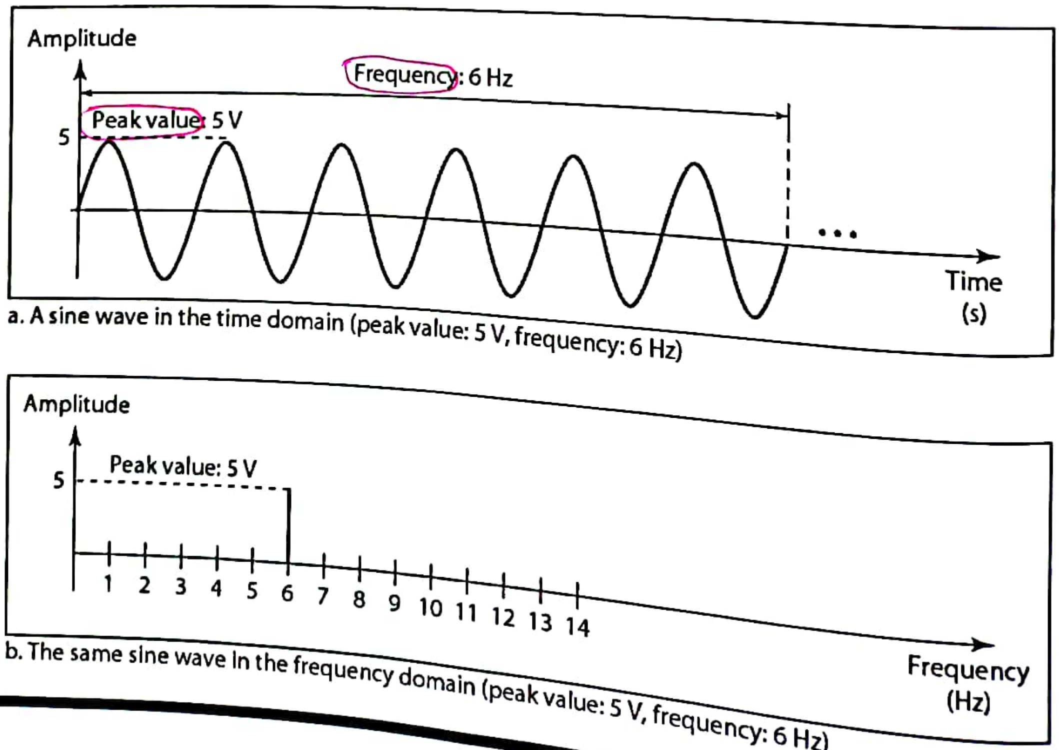
سرعة الضوء ← c ، الزمن ← T

عشان احسب ال Wavelength

3.23

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

Phase = 0
لانه مافي shift



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

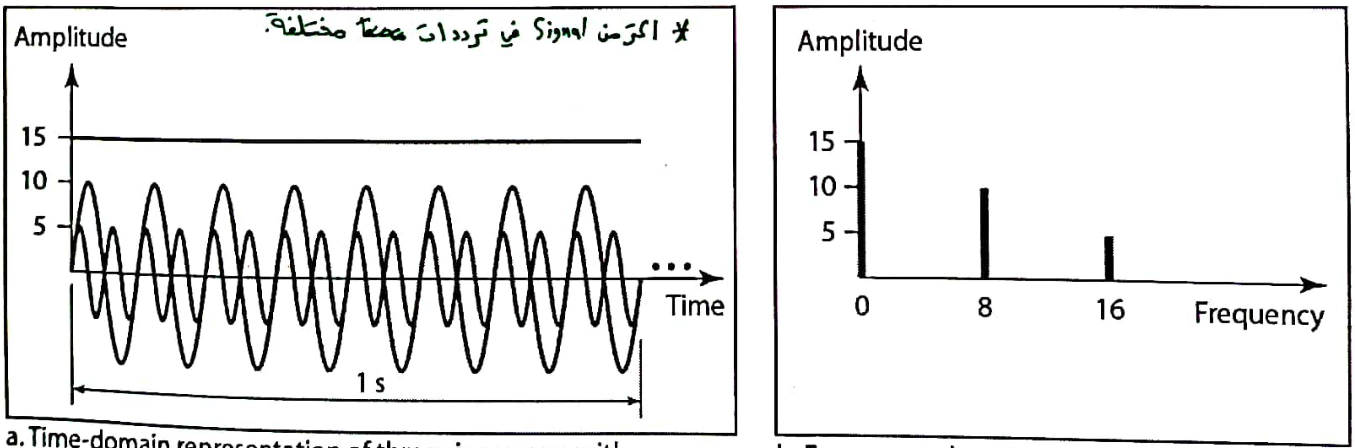
b. The same sine wave in the frequency domain (peak value: 5 V, frequency: 6 Hz)

3.24

Note

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

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

Example 3.7

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.

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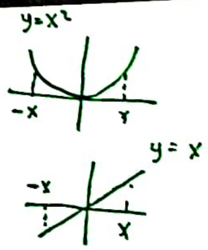
Note

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.

مكونة من اكثر من freq مجموعين مع بعض.



even
$f(x) = f(-x)$
odd
$f(x) = -f(-x)$

Bandwidth = اعلى freq لل Signal - اوطى freq لل Signal

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

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.

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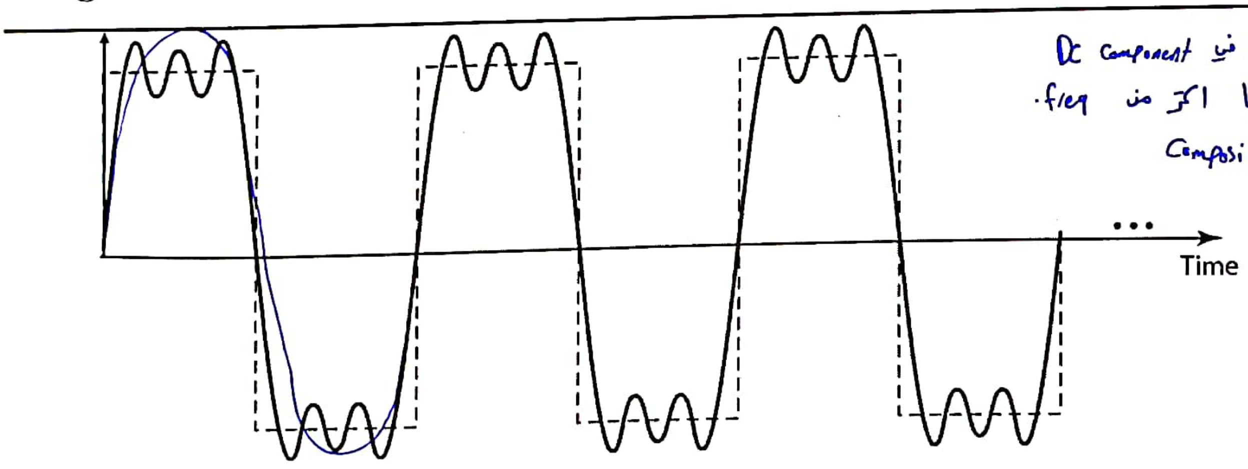
Note

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

زي ما يلا و اسجل صوتيا

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

Figure 3.9 *A composite periodic signal*



* ما في DC component

* فيها اكثر من freq.

* Composite

<u>*Example 3.8*</u>

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.

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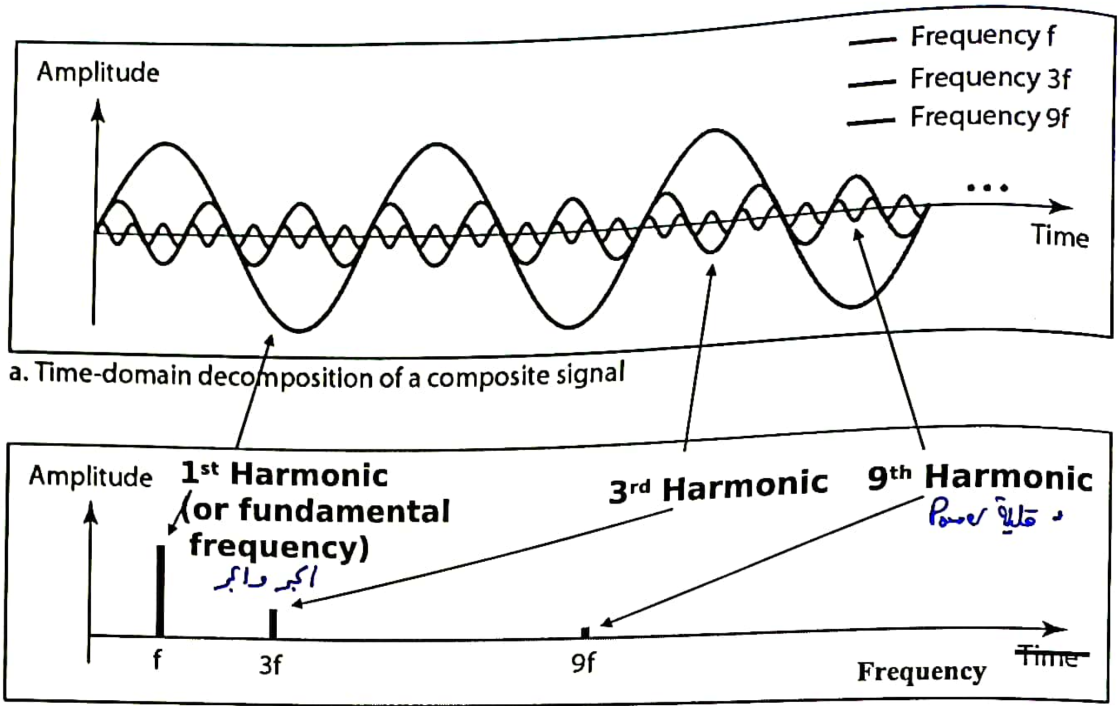
Figure 3.10 *Decomposition of a composite periodic signal in the time and frequency domains*

* كل ما زاد الـ freq قل الـ Power



a. Time-domain decomposition of a composite signal

b. Frequency-domain decomposition of the composite signal

اذا بدي ابعث الـ Square wave وكانت مو بقدر اعطا بس بده يكفه قصير.

3.31

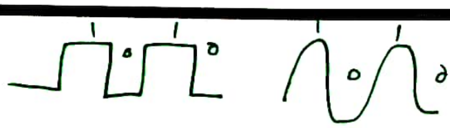
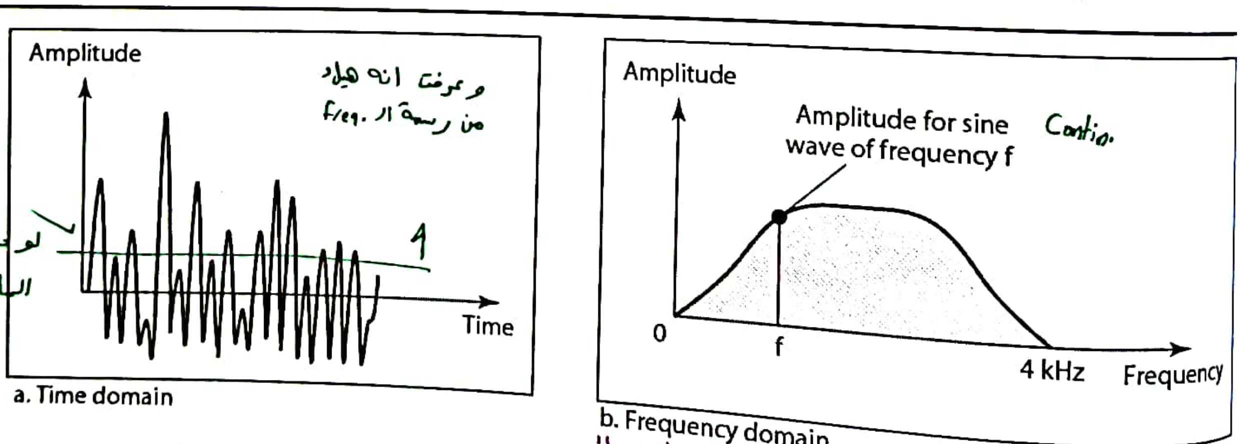


Figure 3.11 *The time and frequency domains of a nonperiodic signal*

nonperiodic ما الها شكل معين.

لو نسبنا الـ time avg الـ ال.



a. Time domain

b. Frequency domain

"Continuous"

Example 3.9

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.

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*

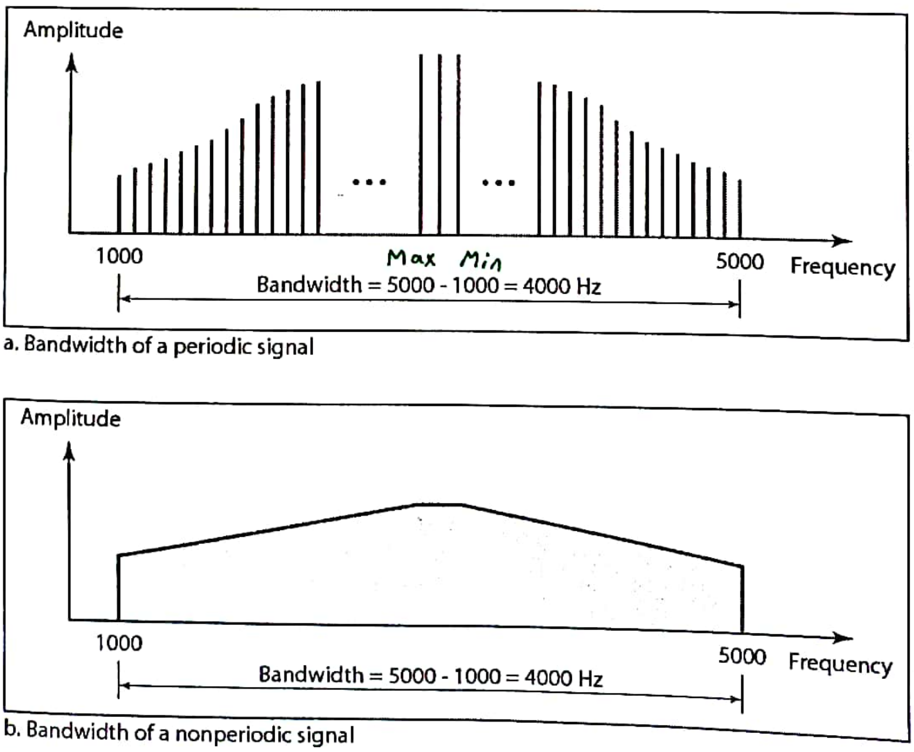
Note

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

* bandwidth signal بتلاقي عكس ال bandwidth channel

3.33

Figure 3.12 ***The bandwidth of periodic and nonperiodic composite signals***



a. Bandwidth of a periodic signal

b. Bandwidth of a nonperiodic signal

* discrete Component.
* Periodic

3.34

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

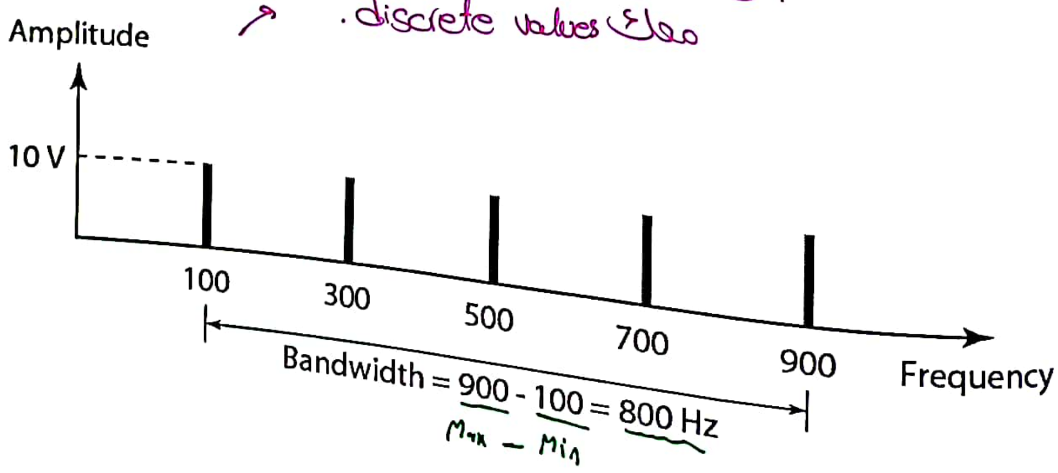
Let f_h be the highest frequency, f_l the lowest frequency, and B the bandwidth. Then

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

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***

بتعرف 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

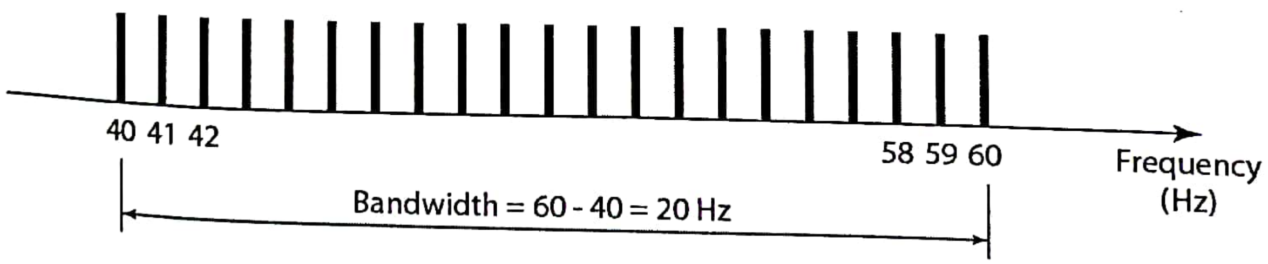
Let f_h be the highest frequency, f_l the lowest frequency, and B the bandwidth. Then

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

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***

هيك بطلع شكلها



Example 3.12

total ال

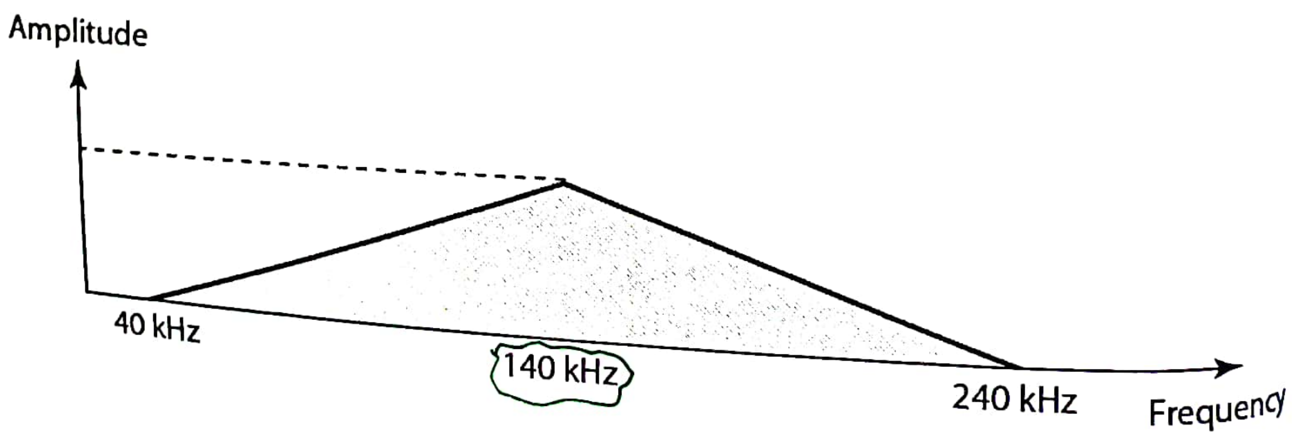
A nonperiodic composite signal has a bandwidth of 200 kHz, with a middle frequency of 140 kHz and peak amplitude of 20 V. The two extreme frequencies have an amplitude of 0. Draw the frequency domain of the signal. → باعتبار انه ال Figure حتكون symmetrical.

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.

3.39

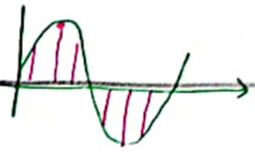
Figure 3.15 *The bandwidth for Example 3.12*



3.40

Sampling freq.

$F_s \geq 2 f_{max}$



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.

B

ال Range الـ AM

ال B للـ ∞ = Square wave

* وإذا أكثر من User بده يبعث مشكلة

* إذا بدي أرسل الـ Signal لمسافة بعيدة بدي Power مشكلة.

[*] الحل أقسم الـ Signal.

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

Bandwidth = $f_2 - f_1$

3.41

$S_c(t) = A \sin(w_c t + \phi_c)$

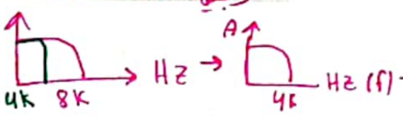
الانسان بسمع: 20 Hz – 20 KHz

وبحكي: 0 – 8000 KHz

Example 3.14

كل ما تستخدم الترددات بدون ما أخرب اشي.

بقدر أقص الـ freq لحد مثلا 4K وتضلها مفهومة.

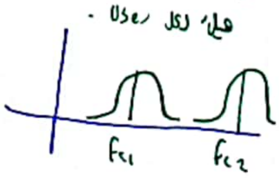


Another example of a nonperiodic composite signal is the signal propagated by an FM radio station. In the United States, each FM radio station is assigned a 200-kHz bandwidth. The total bandwidth dedicated to FM radio ranges from 88 to 108 MHz. We will show the rationale behind this 200-kHz bandwidth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

* كل يوزر بحكي بـ 4K

فلازم أعطي لكل يوزر Range مختلف.

فلي لكل User.



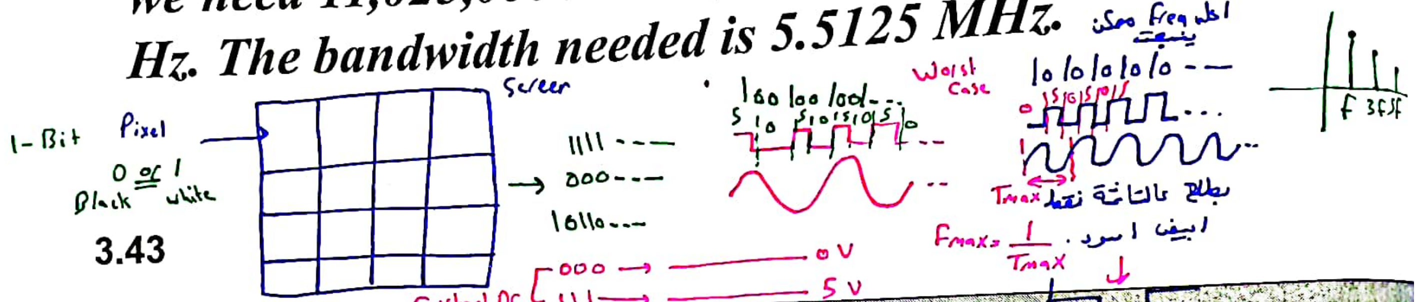
$\underbrace{4000 * 2}_{\text{\# of Samples}} * \underbrace{8 \text{ bits}}_{1 \text{ Sample}}$

= 64 Kbps

3.42

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.



3.43

3-3 DIGITAL SIGNALS

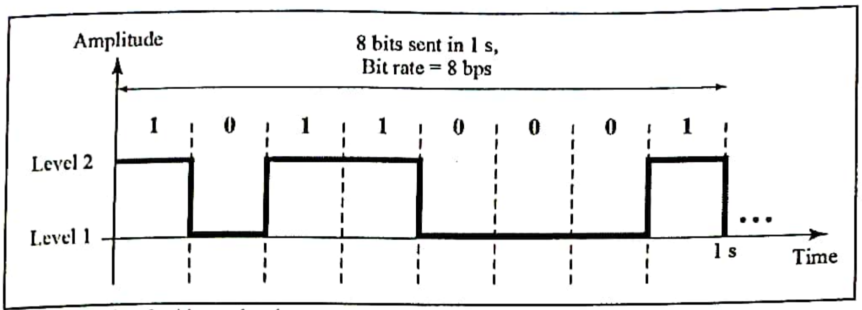
*In addition to being represented by an analog signal, information can also be represented by a **digital signal**. For example, a 1 can be encoded as a positive voltage and a 0 as zero voltage. A digital signal can have more than two levels. In this case, we can send more than 1 bit for each level.*

Topics discussed in this section:

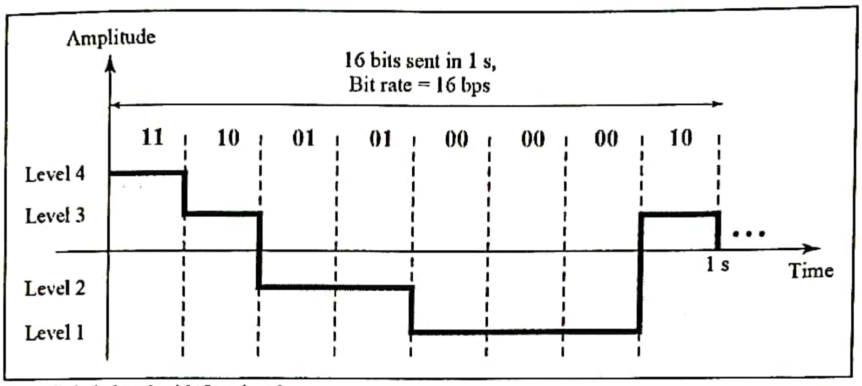
Bit Rate
Bit Length
Digital Signal as a Composite Analog Signal
Application Layer

3.44

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 الـ data Rate
برفع الـ Num of levels
بس علينا
حيكون صعب شوي
لوجود الـ 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.

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

معطيات 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

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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.

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Bit Length

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

سرعة الانتشار

Wavelength = Propagation Speed x Period

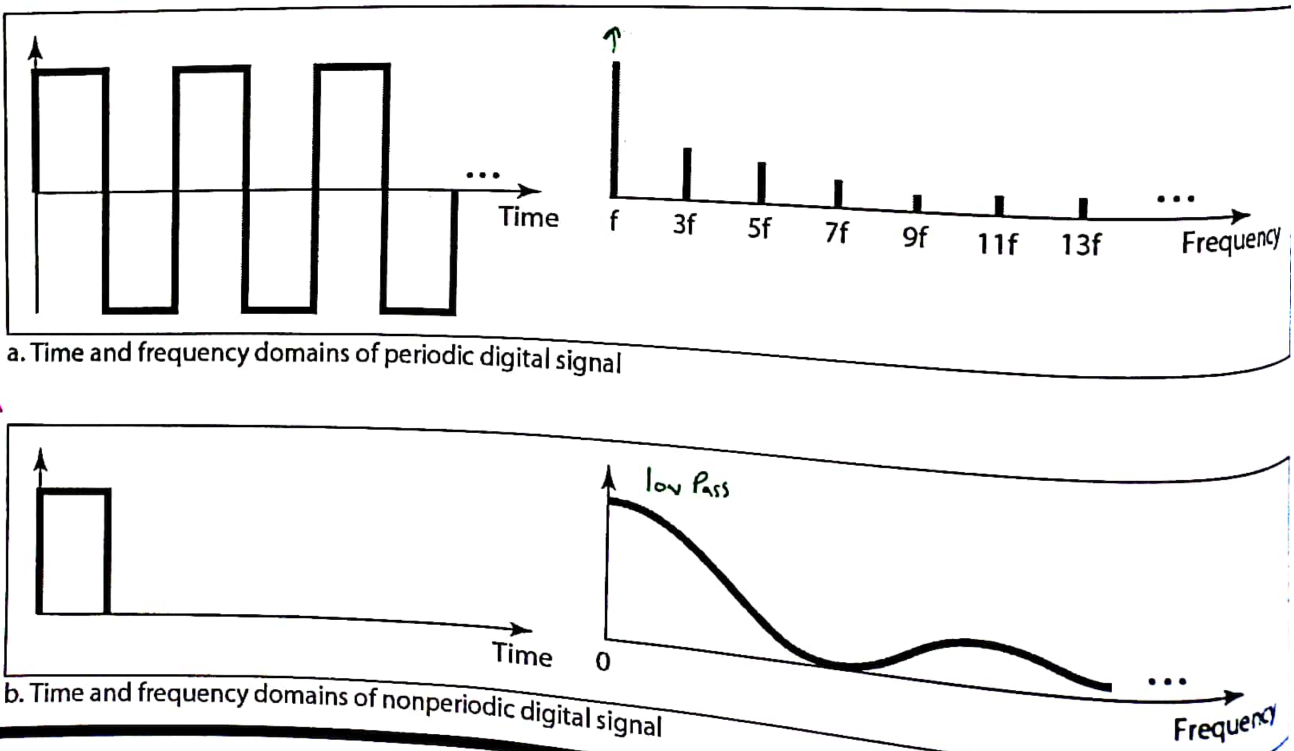
- The bit length is the distance that one bit occupies on the transmission medium, which is a characteristic of the digital signal.

$\frac{1}{\text{Bit Rate}}$

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*

harmonic اكبر ال Power بيكون مخزن بال



a. Time and frequency domains of periodic digital signal

Infinite Bandwidth

b. Time and frequency domains of nonperiodic digital signal

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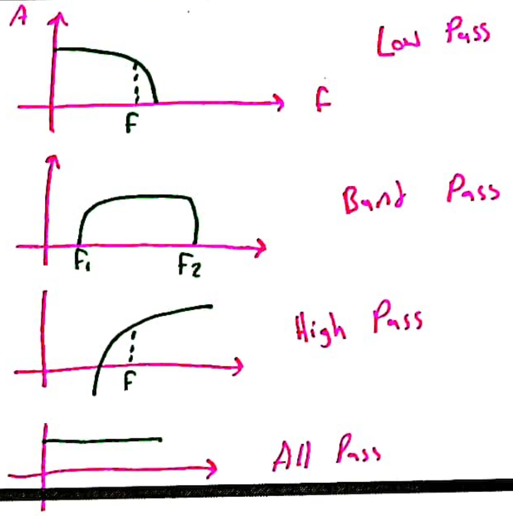
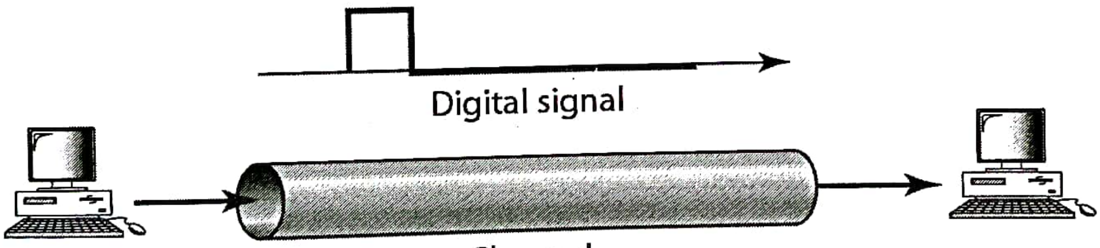
Note

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



3.53

Figure 3.18 ***Baseband transmission***



انه تبعث ال Signal كما هي (Digital)

- Baseband transmission:
  - * Sending a digital signal over a channel without changing it to analog (i.e.; without modulation)

    لم لما نحولها من تكلا لأخر
  - * 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

3.54

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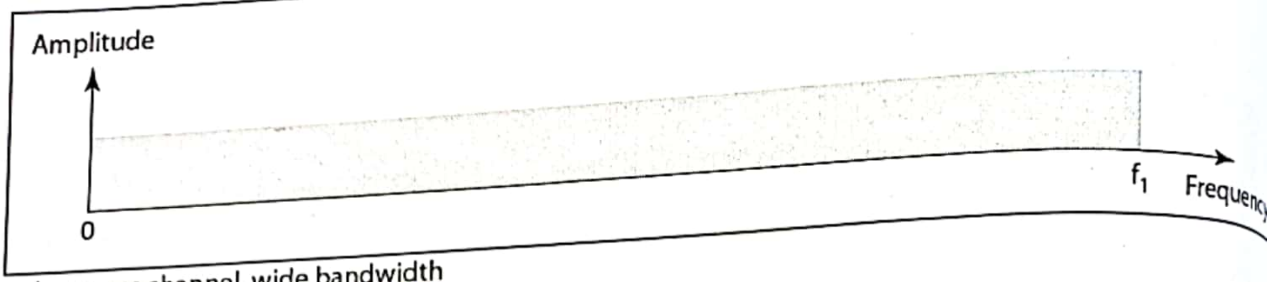
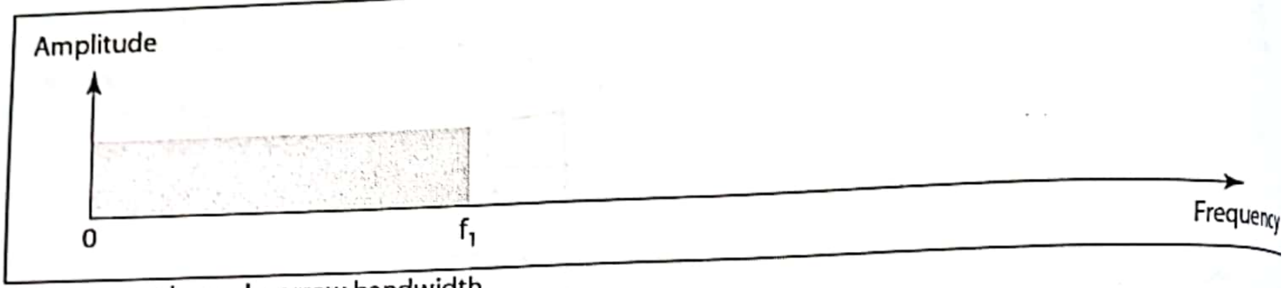
Figure 3.19 ***Bandwidths of two low-pass channels***



a. Low-pass channel, wide bandwidth

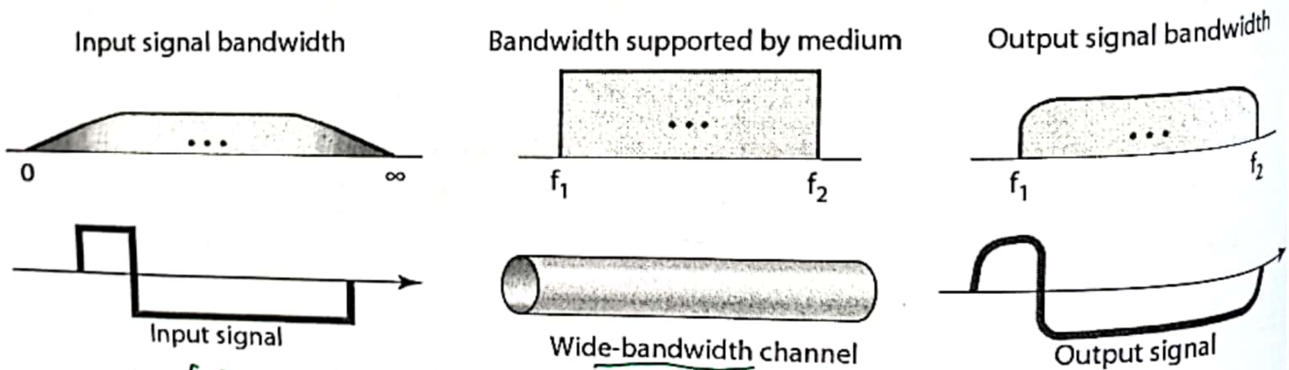


b. Low-pass channel, narrow bandwidth

3.55

* Case 1: Low-Pass Channel with Wide Bandwidth

Figure 3.20 ***Baseband transmission using a dedicated medium***



كانها Filter رح يسمح للي بيا f_1, f_2 فقط

بشبه كثير لبشكل ال Input Signal

3.56

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.

عمليا مستحيل

3.57

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.

بتمرق كل ال freq.

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.*

3.58

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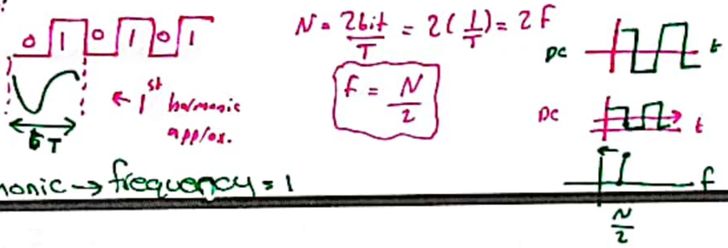
Case 2: Low-Pass Channel with Limited Bandwidth

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

Worst-Case Scenario
N : bits/s
$f = \frac{1}{T}$



* 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

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

$\boxed{f = \frac{N}{2}}$ $\boxed{B = \frac{N}{2}}$

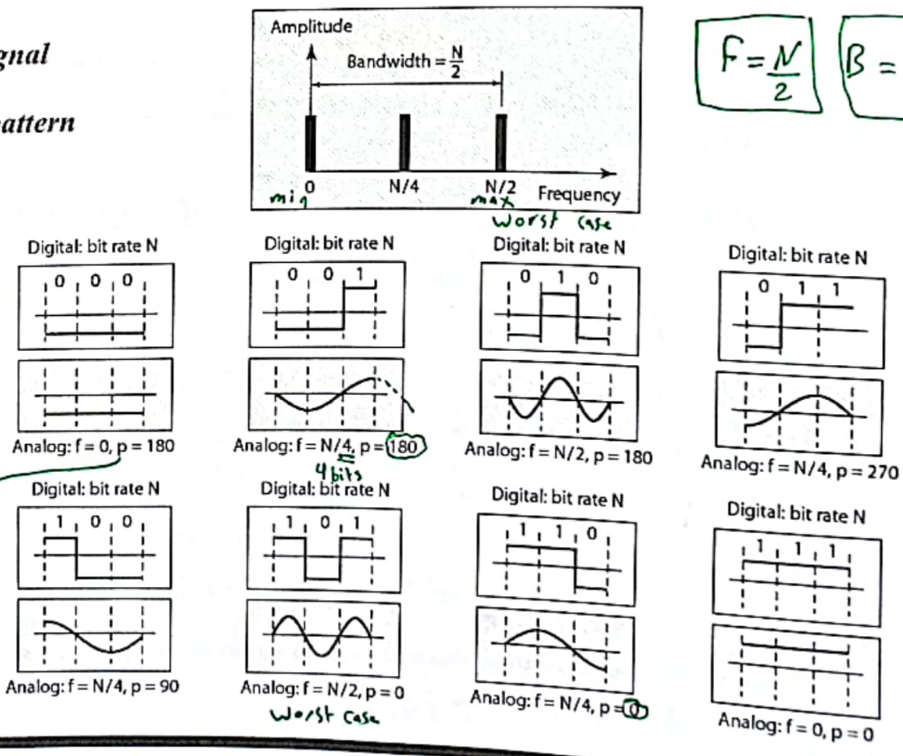


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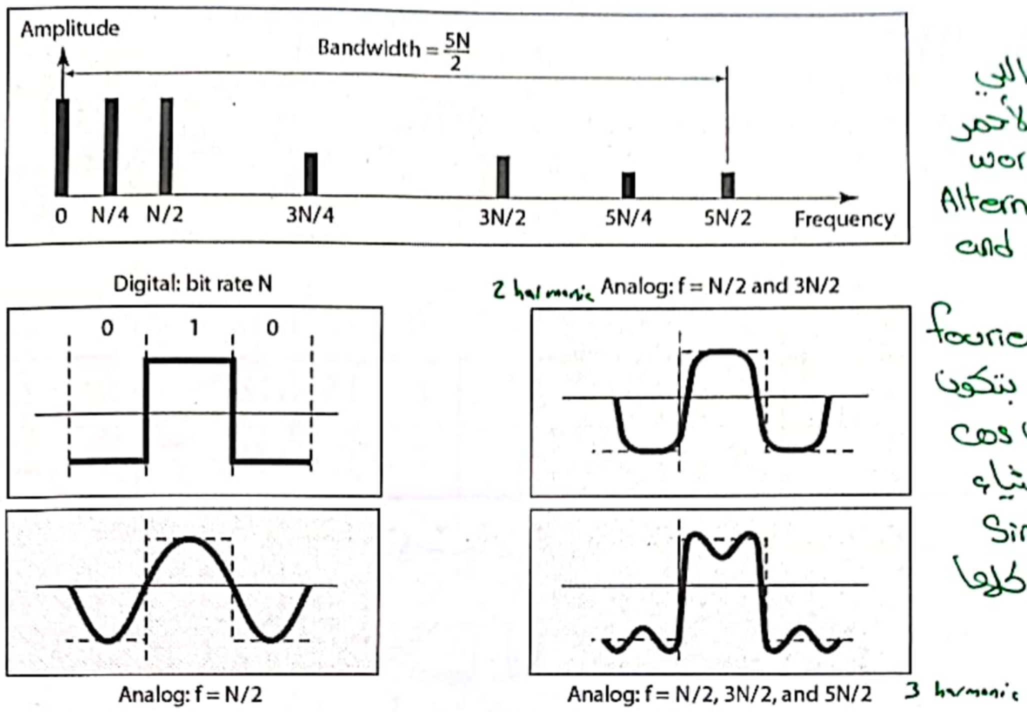
Figure 3.22 *Simulating a digital signal with first three harmonics*



* الي بالأحمر worst case Alternative zeros and ones.

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

* min bandwidth → take one harmonic.

3.61

* لو ضليت أزيد بالـharmonic لأزيد الدقة حرفع ضريبة انه الـbandwidth بزيد والـ bandwidth مشكلة بالاتصالات صار ومسافة خطرة.

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

Example 3.22

N = 2f

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

لما نعمل الـ transmittion 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.

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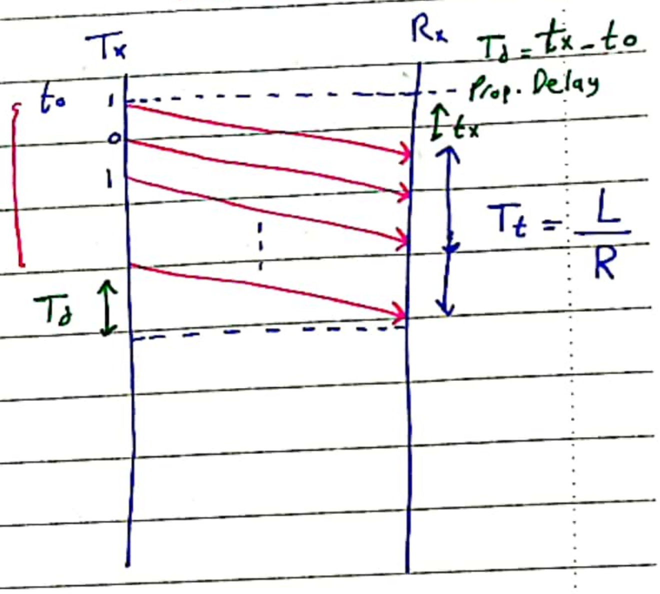
5/3

Total Delay $= T_d + \cancel{T_p} + T_t + T_Q$ ($T_p \approx 0$)

(one-way)

Link Speed $= R$



$T_d = t_x - t_0$

Prop. Delay

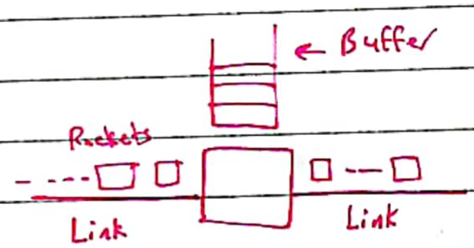
$T_t = \dfrac{L}{R}$

Total time to transmit a packet

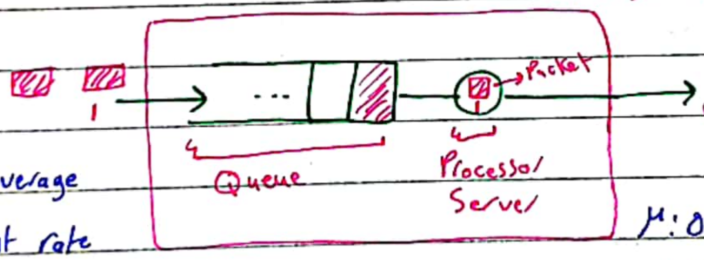
$= T_d + T_t + \cancel{T_d}$

$= T_d + T_t$

* T_Q :



Link (System)



λ: average input rate (PKts/S)

* أول Packet عندي بتدخل بال Process واللي بعدها بتظل بال Queue لحد ما تخلص الأولى وعدد ال Packets كامل بيساوي عدد اللي بال Queue + 1

μ: Outpute rate (سرعة ال link) (PKts/S) (average Value)

* لو كل الجامعة بتستخدم ال link مع بعض رح يكون ال throughput بيساوي ال μ

Performance measures :

1. Number of PKts in the link. (average value).
2. T_Q : queueing delay.
3. Through put.

Five Apple

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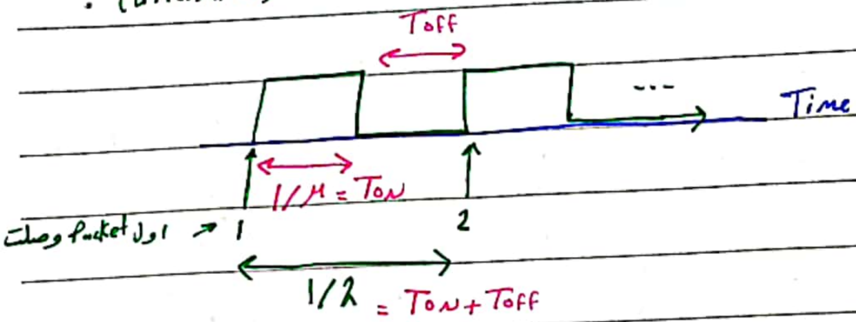
Traffic intensity $= \frac{\lambda}{\mu}$

. (utilisation)



نسبة الانشغال

Duty Cycle $= \frac{T_{on}}{T_{on} + T_{off}}$

$= \frac{1/\mu}{1/\lambda} = \frac{\lambda}{\mu}$

$$P_k = \underbrace{\left(1 - \frac{\lambda}{\mu}\right)}_{\text{empty}} \underbrace{\left(\frac{\lambda}{\mu}\right)^k}_{\text{busy}}$$

$\left(\frac{\lambda}{\mu}\right)\left(\frac{\lambda}{\mu}\right)\left(\frac{\lambda}{\mu}\right)$ k

$P(\text{"empty"} \cup \text{"busy"}) = P(\text{"empty"}) + P(\text{"busy"}) = 1$

$P(\text{"empty"}) = P_0 = 1 - \frac{\lambda}{\mu} = 1 - U$

$P(\text{"busy"}) = \frac{\lambda}{\mu} = U$

$\overline{}$ ← avg. mean

N

$\underset{RV}{X} = \{X_0, X_1, X_2\}$

$\overbrace{X_0, X_2, X_2, X_2, X_1, X_1, \ldots}^{N \text{ (Sample size)}}$

$\bar{X} = \lim_{N\to\infty} \frac{N_0}{N} X_0 + \lim_{N\to\infty} \frac{N_1}{N} X_1 + \lim_{N\to\infty} \frac{N_2}{N} X_2 + \ldots$

$= P(X_0) X_0 + P(X_1) X_1 + P(X_2) X_2 + \ldots$

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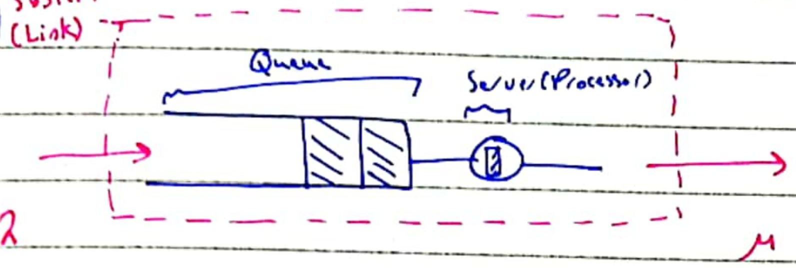
$$\bar{N} = \sum_{k=0}^{\infty} k P_k$$

7/3



average packet rate
(PKts/s)

average packet service rate
(output rate)

① $\bar{N}$: average # of PKts in the system.

② $\bar{X}$: throughput.

③ $\bar{D}$: delay (T_Q)

$$P_k = \Big(1 - \underbrace{\frac{\lambda}{\mu}}_{\upsilon}\Big)\left(\frac{\lambda}{\mu}\right)^k$$

هون لا ع اخر وحدة اجت
شافته مليان وما دخلت عشان
هيلد k مش k-1

$$= \left(1-\frac{\lambda}{\mu}\right)\underbrace{\left(\frac{\lambda}{\mu}\right)\left(\frac{\lambda}{\mu}\right)\cdots\left(\frac{\lambda}{\mu}\right)}_{k}$$

① $\bar{N} = \sum_{k=0}^{\infty} k P_k$

$$= \sum_{k=0}^{\infty} k (1-\upsilon)\upsilon^k$$

$$= (1-\upsilon)\boxed{\sum_{k=0}^{\infty} k \upsilon^k}$$

$$= \frac{(1-\upsilon)(\upsilon)}{(1-\upsilon)^2} = \frac{\upsilon}{(1-\upsilon)}$$

لو ال input rate = ال output rate
كانت $\upsilon = 1$ فبيطلع ∞ يعني ما بصير
يتساوى ~~...~~ مع بعض.

⊛ $R = 10$ Mbps

$L = 8$ bits

$$R_p = \frac{R}{L}$$

R ← constant; R_p ↑ random; L ↖ random

⊛ $P(A \cap B) = P(A)\,P(B|A)$

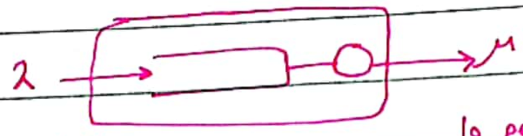
$= P(A) + P(B)$ → {Independent?}

⊛ $\bar{X} = \sum_{x} x P_x$

⊛ $\sum_{k=0}^{\infty} r^k = \frac{1}{1-r}, \ |r|<1$

$$\frac{d}{dr}\left[\sum_{k=0}^{\infty} r^k\right] = \frac{d}{dr}\left[\frac{1}{1-r}\right]$$

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



10 people/1 min 10 people/1 min

10 لو كان 7 و هو 3

يكون في 3 اشخاص جوا

عشان ال System يكون عندي Stable لازم

$\boxed{\lambda < \mu}$ → Stability Condition

* بما انه λ اقل من μ لازم

ال throughput يكون λ.

② $\bar{X} = \underset{\text{when busy}}{\text{Rate}}\ P(\text{"busy"}) + \underset{\text{when empty}}{\text{Rate}}\ P(\text{"empty"})$

$= \mu(U) + \phi(1-U)$

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

① $\bar{N} = \frac{U}{1-U}$

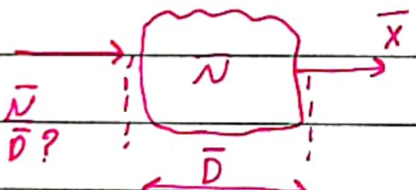
$\lambda < \mu$

② $\bar{X} = \lambda$

③ N of Packets

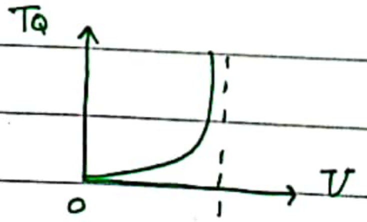
③ $\bar{D} = T_Q = \frac{1}{\mu - \lambda}$

الزيادة بال D مش linear ← exp



$\bar{D} = \frac{\bar{N}}{\bar{X}} = \frac{U/\lambda}{1-U}$

$= \frac{1/\mu}{1-\frac{\lambda}{\mu}} = \frac{1}{\mu - \lambda}$



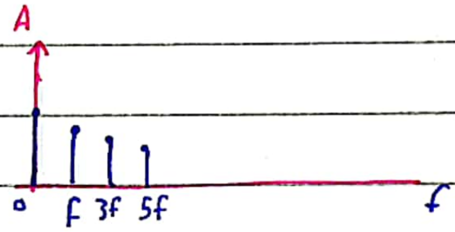
Slide How do loss and delay occur?

* ال loss اشي خطير وال delay بيكون زي منو انو قريب يصير الخطر.

23/3

* Maximum Bit Rate (N) at 1^{st} harmonic approx

Why?

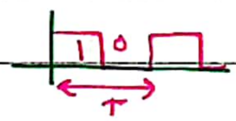


$B = f_{max} - f_{min}$

$= f - 0 = f$

$B = \boxed{f = \frac{N}{2}} \longrightarrow N = 2B$

$N = \frac{2\ bits}{T} = 2f$ worst case Scenario



* Assume 2^{nd} harmonic approx

$B = \frac{3N}{2} \longrightarrow N = \frac{2}{3} B$

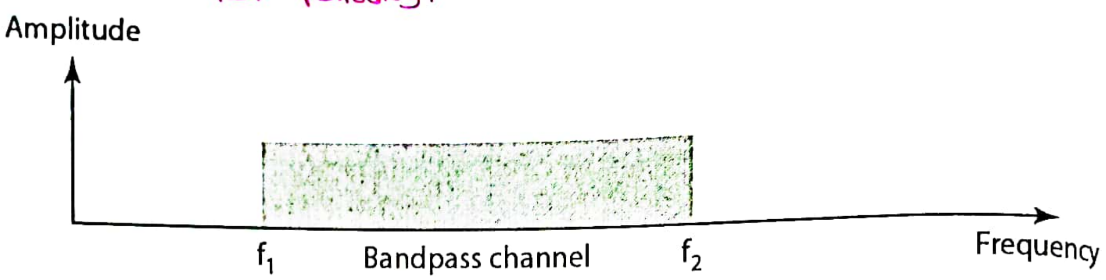
* Assume 3^{rd} harmonic approx

$B = \frac{5N}{2} \longrightarrow N = \frac{2}{5} B$

Figure 3.23 *Bandwidth of a bandpass channel*

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

اللي ممكن نستوعبها أما ال Signal تصنف ك Periodic
أو non-Periodic



* ليه ما في channel بتمرر كل ال frequencies ؟ لأنه في frequencies
تمر وفي لا يعني زي كأنه بفلتر ال 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.

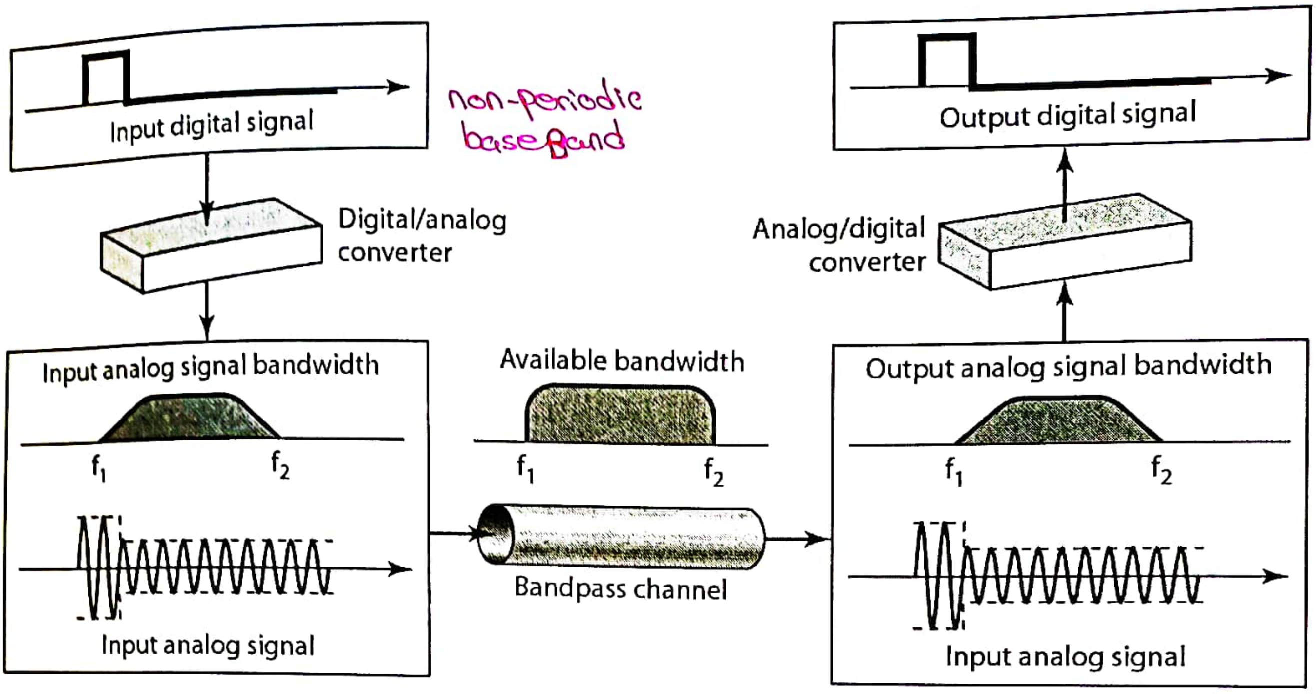
لأنه Squarewave ←
إذا كانت Periodic
العامل ال DC
component إلى
∞ - إذا عندك

Channel معينة تحوط
ال freq. اللي تحت وفوق
وحيمشي جزء من ال Channel
بالوسط.

إذا وصلنا ال freq اللي فوق (العالي) مو مشكلة
لأنه أحسن (لأنه ال Power قليل وال Amp
قليل) أما ال freq اللي تحت موكويس لأنه Power و Amp
عالي. في مشكلة إنه ما بنقدر نبعت Squarewave
على bandpass
channel لازم نحولها زي ما مكتوب .

3.68

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



3.69

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 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.

0→4

3.70

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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.

ضعف

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

مُضاد

Figure 3.25 *Causes of impairment*

Signal

Impairment causes

- **Attenuation**
  - Loss of signal energy due to channel resistance
  - في كمان Capacitance و Inductors
  - 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
  - * لما نجمع (signals) لما بطلع square wave بسبب تجمع بعض الـ sin wave
- **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

$P_s > P_R$

Channel

S —— R

3.73

R_s P_s P_R V_i R_L V_o C

$V_o = \left(\frac{R_L}{R_L+R_i}\right)V_i \rightarrow Z_c = 1/j\omega c$

$1/j2\pi f c$, لو كانت $f=0$ بطلع ∞ open circuit

لو كانت $f=\infty$ بطلع zero short circuit

f_1 f_2

Figure 3.26 *Attenuation*

$\frac{P_R}{P_s} = \frac{1}{2}$ ← معناها انه الـ receiver فقد نصف الـ Power.

S — 1/3 — 1/4 — R

P_s $\frac{P_s}{3}$ $\left(\frac{P_s}{3}\right)\frac{1}{4}$ P_R

$P_R = \frac{P_s}{12}$

$\frac{P_R}{P_s} = \frac{1}{12}$

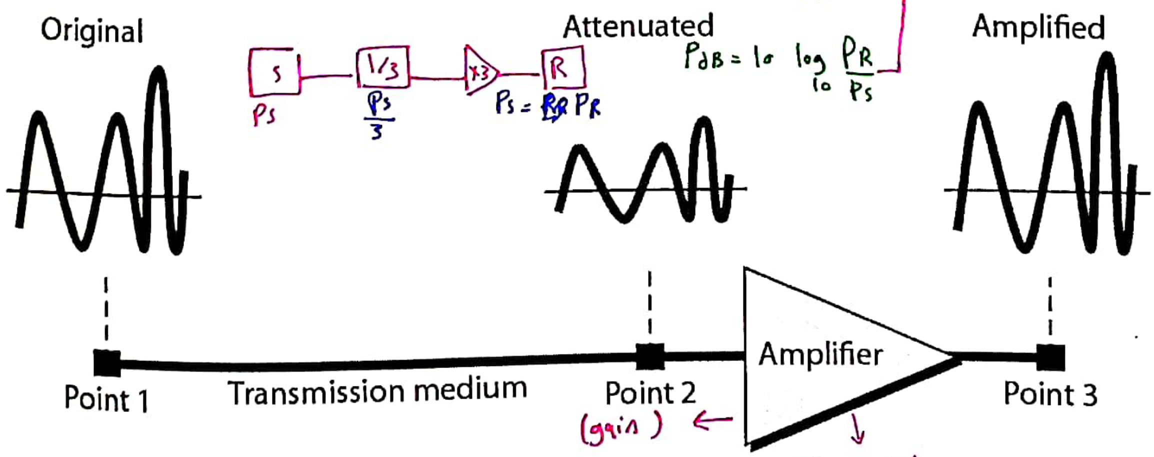
* اذا طلع (+) بيكون تضخيم.

* " " (−) Attenuation.

S — 1/3 — x3 — R

P_s $\frac{P_s}{3}$ $P_s = P_R$

$P_{dB} = 10 \log_{10} \frac{P_R}{P_s}$



(gain) ← لنعوض الـ energy اللي فلتت.

S — −2dB — −1dB — 10dB — R

$P_{s\,dB}$ $P_{R\,dB}$

$P_{R\,dB} = P_{s\,dB} - 2 - 1 + 10$

$= P_{s\,dB} + 7\,dB$

3.74

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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

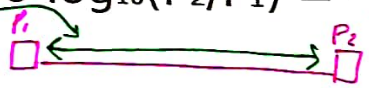
اقرب لل منطق

- * 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



$P_1\,dB = 10 \log_{10}\left(\frac{P_1}{1W}\right)$

3.75

$dB = 10 \log_{10} \frac{P_2}{P_1} \rightarrow$ in watts

$P_1\,dBm = 10 \log_{10}\left(\frac{P_1}{1mW}\right)$

Example 3.26

لازم لحالي اعرف من هون.

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

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

← بنعرف إنه نص ال Power راحت.

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

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Example 3.27

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

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

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

3.77

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.

وراء بعض

In Figure 3.27 a signal travels from point 1 to point 4. In this case, the decibel value can be calculated as

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

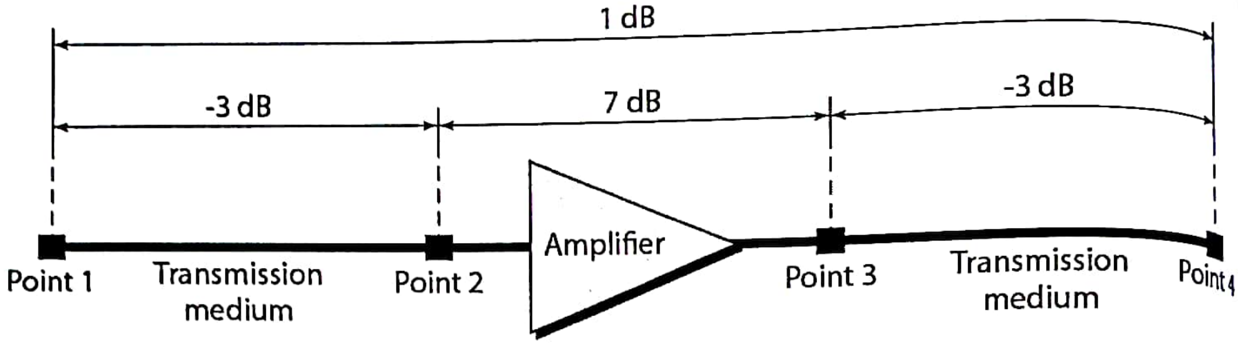
صرت اجمع
واطرح مباشرة
بدل ما اضرب واقسم

$$10\log_{10}\frac{P_4}{P_1} = 1 \text{ dB}$$

Amplification.

3.78

Figure 3.27 *Decibels for Example 3.28*



بدل ما أضرب وأقسم بصير أجمع
وأطرح.

3.79

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 $dB_m = -30$.

Solution

We can calculate the power in the signal as

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

لا هون ما في نقطتين
(بس نقطة وحدة)

$$dB_m = 10 \log_{10} P_m = -30$$
$$\log_{10} P_m = -3 \qquad P_m = 10^{-3} \text{ mW}$$

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

3.80

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Example 3.30

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

في نقص بالـ Power .

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 ← طوله

Solution

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

We can calculate the power as

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

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

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

total loss من البداية للنهاية

لو طلع أكثر من 2mw بكون غلط لأنه أنا بنزل فأكيد مش حفقد أكثر من 2mw وأنا أساساً ما عندي إلا 2mw

3.81

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

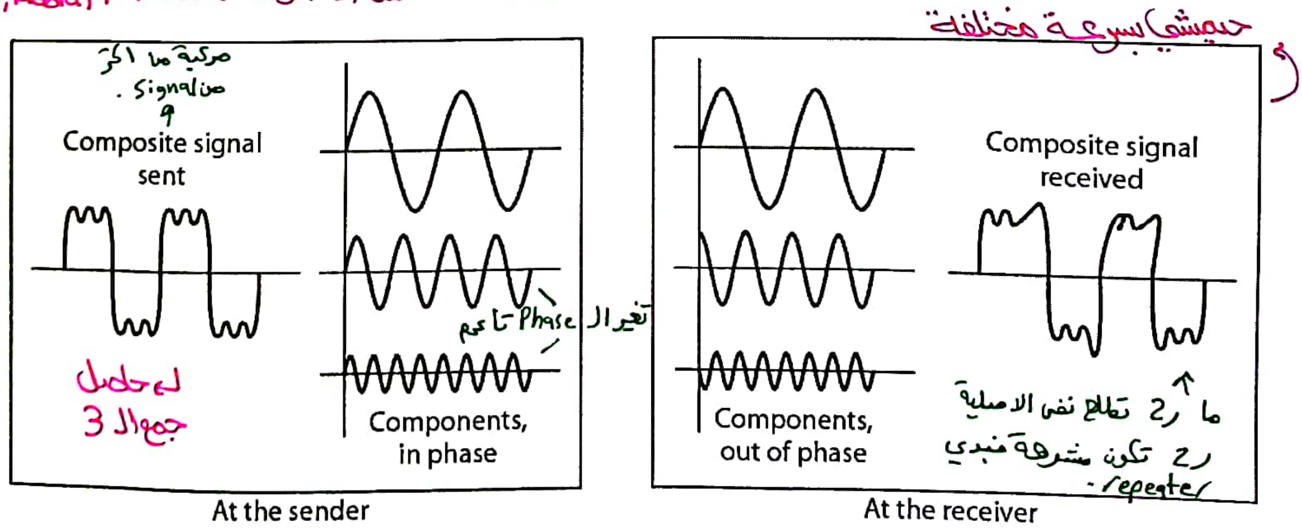
Figure 3.28 *Distortion*

Phase Distortion بالأذى

كلما زادت الـ freq نقصت طول الموجة .

$\lambda f = u$ phase : الأدق ليست دقيقة * $\lambda f = c$

الـ u ← function لـ (f, Media) من الـ f والـ Media $= \frac{1}{\sqrt{\mu \varepsilon}}$ ← Velocity



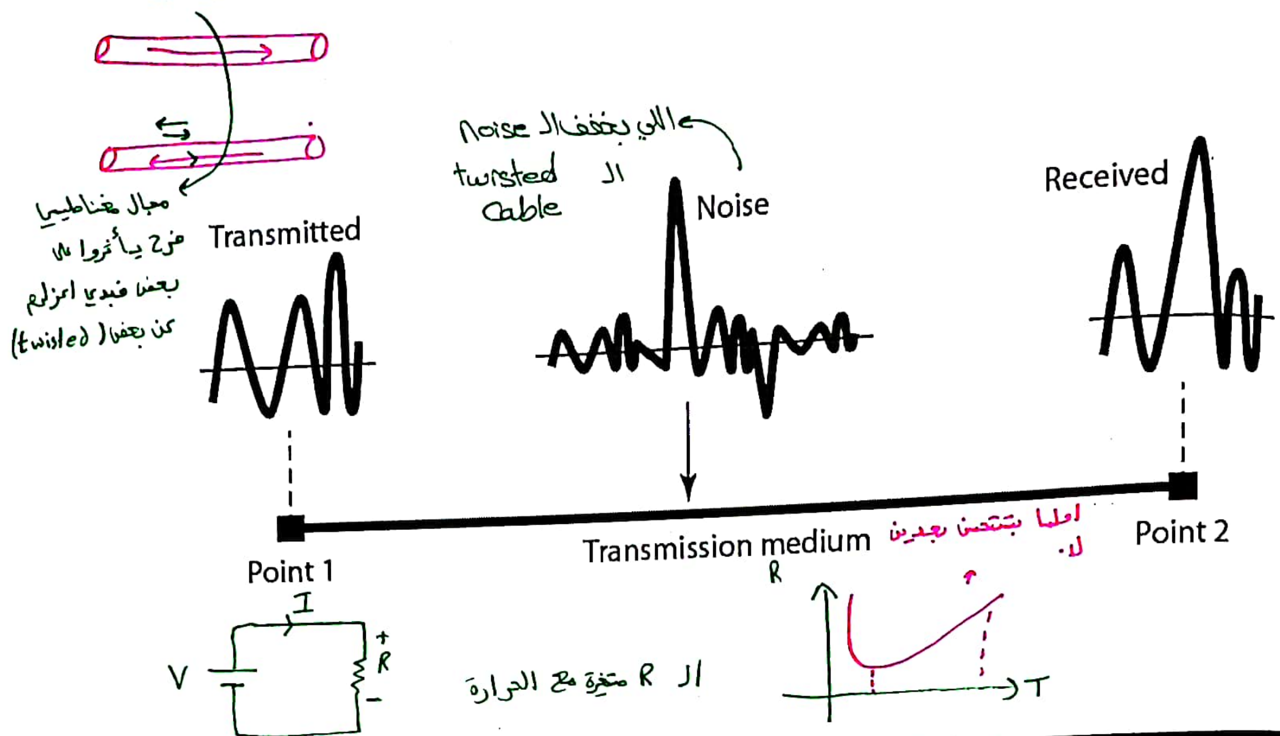
* different freq. في الـ Media تمشي بسرعات مختلفة وهذا يسمى distortion .

لما 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 الـ Power تاعت الـ Signals إلى نسبة الـ Power تبعت الـ 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}
- $SNR_{dB} = 10 \log_{10} SNR = 10 \log_{10} (P_s / P_n)$ (العادية)

لو كانت noise power = 0 بطلع SNR = ∞
وهاد مش موجود.

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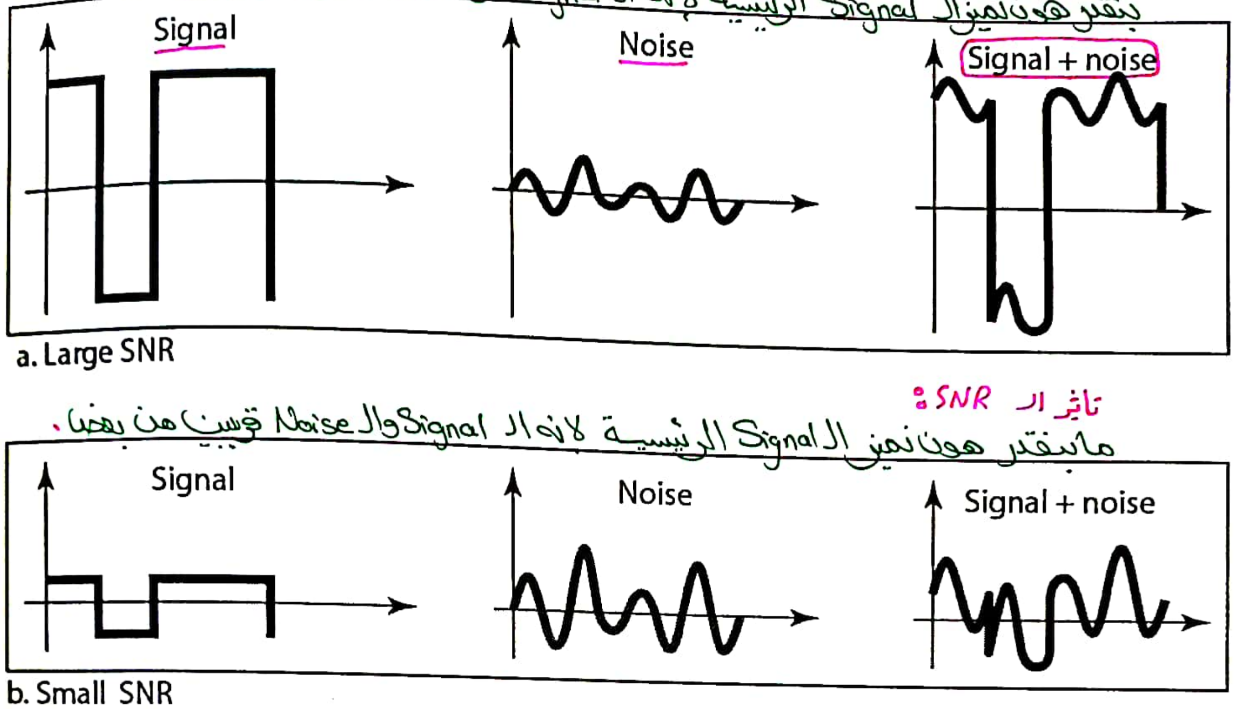
Figure 3.30 *Two cases of SNR: a high SNR and a low SNR*

بتقدر هون نميز ال Signal الرئيسية لأنه ال Signal و ال Noise بعاد عن بعض.



تأثير ال SNR:

مابنقدر هون نميز ال Signal الرئيسية لأنه ال Signal وال Noise قريبين من بعض.

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:

$$\boxed{\begin{aligned} \text{SNR} &= \frac{10{,}000\ \mu\text{W}}{1\ \text{mW}} = 10{,}000 \\ \text{SNR}_{\text{dB}} &= 10 \log_{10} 10{,}000 = 10 \log_{10} 10^4 = 40 \end{aligned}}$$

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Example 3.32

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

ideal case
مش موجودة بالواقع .
(نظرية)

$$SNR = \frac{\text{signal power}}{0} = \infty$$

$$SNR_{dB} = 10 \log_{10} \infty = \infty$$

→ 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 i how fast we can send data, in bits per second, over a channe

Data rate depends on three factors:

1. The bandwidth available (channel)
2. The number of signal levels used
3. The quality of the channel (the level of noise)

نوعية ال Channel الموجودة .

اكتشفها Shannon

Topics discussed in this section:

- Noiseless Channel: Nyquist Bit Rate → افضل data rate اذا ما كان Noisy . عمليا ما بتقدر نحصلها
- Noisy Channel: Shannon Capacity → افضل data rate اذا كان Noisy .
- Using Both Limits

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Noiseless Channel: Nyquist Bit Rate

- Defines the theoretical maximum bit rate possible for noiseless channels

مثالي مش موجود بالطبيعة

- $\text{Nyquist Bit Rate} = \boxed{2} \times \text{Bandwidth} \times \log_2(L)$ → 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 $\log_2 2 = 1$.

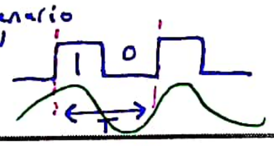
- Increasing the number of levels is not practical:
  - It imposes a burden on the receiver → *حيتأثر استقبال الـ reciever للـ Signal. (كيف بدكون)
  - It complicates the receiver design → * الـ receiver بصير معقد
  - It reduces the reliability of the system (i.e.; increase the probability of reception errors)

* أيضًا بكون فيه noise.

3.89

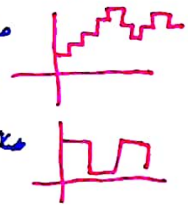
$\text{Bit rate} = 2 \times B \times \log_2(L)$

Worst case scenario



صار اصعب اميز بين بعض

هنا سهل اميز بينهم



$\text{Bit rate} = \frac{2 \text{ bits}}{T} = 2f$

but, in best case scenario

$f = 0$ (transmitting all 1's or all 0's)

$\text{Bit rate}_B = 2 \times B \times \log_2(2)$

$= 2 \times B$

Note

عشان ازيد الـ bit rate بزيد الـ (L) نظريا بقدر ازيدها قدما بدي

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

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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.

3.91

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

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

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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

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

telephone line → 4 KHz
↳ Bandwidth

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:

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

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

→ 128 bit rate
اقرب اتي عليها.

↳ 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.

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Noisy Channel: Shannon Capacity

· Noisy ← Channel لـ هاي بتاخد بعين الاعتبار انه ال

- Defines the theoretical maximum bit rate possible for noisy channels

  * كل ما زاد ال Bandwidth
  كل ما زاد ال data Rate
  وكل ما زاد ال SNR كل ما زاد ال bit Rate

Bits/sec

- Channel Capacity = Bandwidth x $\log_2(1+\text{SNR})$
  - * 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

3.95

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

كأنه بقسم ٥٥ لا

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.

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Example 3.38

لو ما حكا بنفرض 4000

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.

تقلل ال noise
أو نزيد ال Signal Power
نزيد ال bandwidth

Example 3.39

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

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

Example 3.40

ر 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 نقرب أو لا *

* زي كانها linear

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

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

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

عالدفتر 30/3

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.

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

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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.

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

اخذنا شوي أقل
من ال max
(6Mbps) عشان ما نمشي
عالحفة.

3.101

*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.

3.102

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 → units cycles/s (Hz) → بعاملاً بتغير → bits per second
Throughput
Latency (Delay)
Bandwidth-Delay Product

Note

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

تعريف ال bandwidth :

① ← 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 ال freq الي ممكن ال channel تمرقها أو إنها تكون موجودة بال Signal.

② ← سياق مختلف لمجال الاتصالات =

The second, bandwidth in bits per second, refers to the speed of bit transmission in a channel or link.

bps

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Example 3.42

The bandwidth of a subscriber line is 4 kHz for voice or data. The bandwidth of this line for data transmission can be up to 56,000 bps using a sophisticated modem to change 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
زي النظام الاوروبي ، اخر 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.

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بوقف عند ال Sender ويعد كم Packet عم بيبعت Per second

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

Throughput versus Bandwidth

Throughput = Bandwidth ← في أحسن الأحوال ال

← في أسوأ الأحوال بكون ال loss كبير جداً.

reference لـ ❑ The Bandwidth is a potential measurement of how much we can send data over the link
الآب.

* أهم من ال Bandwidth *

على مستوى ال data link layer

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

كمية الشغل الي بتنجزه.

❑ 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

بس مو كتير مستغلة ٢٠٪ من الي بحتاجه.

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

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

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Latency (or Delay)

one way delay

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

* في نوع تاني round-trip time (رايح وراجع)
بيكون Send و Ack (مش شرط .Sync)

- The total delay consists of four components: → موجود بالسلايدات خارجية
  - Propagation delay = Distance / Propagation Speed
  - Transmission delay = Message Size / Bandwidth (bps)
  - مشكلة Random ← Queuing delay: depends on the current network status
  - Processing delay: depends on the speed and capabilities of the nodes هي سرعة الاجهزة تأثيره صار اقل.

يعتمد ع ال Comm link الي عندي عشان اعرف مين المسيطر اكتر فيهم.

- The Jitter defines the variations in the delay
قيمة التغير في ال

ال delay بيكون مضر لل multimedia Apps.
اذا كان كبير (صوت وفيديوهات).

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.

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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.

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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.

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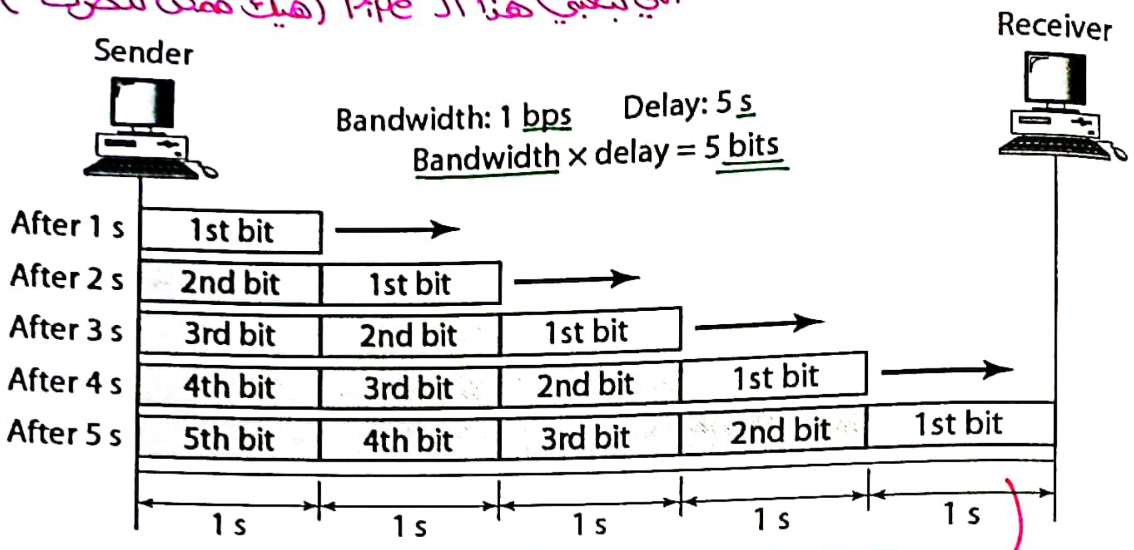
بالدفتر 2/4

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

bits عدد الـ ← # bits * Delay = bandwidth delay product (bps)

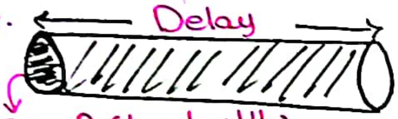
اللي بتعبي هذا الـ Pipe (هيك ممكن نتخيله).



بعد 5 ثواني يادوب وصلت الـ bit 1 الى الـ Receiver.

* كل الـ Pipe تعبت خلال فترة 5 ثواني.

بالـ Transport بنعرفها ب Round trip time.



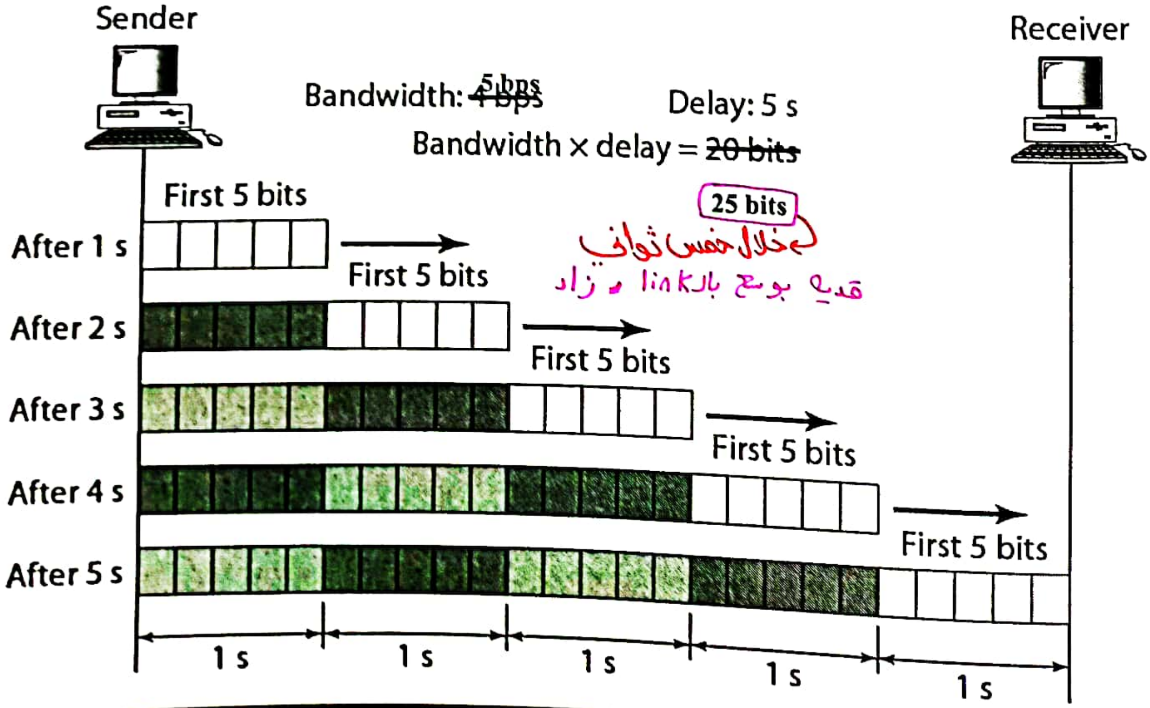
* الـ Pipe capacity in bits/s :

Volume = B x D

BDP → Bandwidth delay product

* كلما زاد الـ Delay أو الـ Bandwidth كلما زاد الـ Pipe volume.

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



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Note

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

3.117

30/3

Example 3.40

Show that

$$C = B \times \frac{SNR_{dB}}{3}$$, when SNR is large

Sol.

given $\longrightarrow$ SNR

$SNR_{dB} = 10 \log_{10} SNR$

$$10^{\frac{SNR_{dB}}{10}} = \cancel{10}^{\cancel{\log_{10}} SNR} = SNR$$

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

$$= B \log_2 (1 + 10^{\frac{SNR_{dB}}{10}})$$

if SNR is large then $10^{\frac{SNR_{dB}}{10}}$ is large.

$$\approx B \log_2 (10^{\frac{SNR_{dB}}{10}})$$

$$= B \frac{SNR_{dB}}{10} \log_2 10$$

is large

$$= B \frac{SNR_{dB}}{\cancel{\frac{10}{\log_2 10}}_{3}} = \boxed{B \frac{SNR_{dB}}{3}} \#$$

Example 3.41

Sol: Use both Nyquist's formula and Shannon's formula.

① Using, $C = B \log_2(1 + SNR)$

$N < 6$ Mbps

دايما ناخذ قيمة N اقل من C

$C = 10^6 \log_2(1 + 63)$
$= 6$ Mbps

② Use lower bit rate say 4 Mbps. → دايما نختار اقرب Rate من ال C ويكون من مضاعفات ال 2.

$N = 2B \log_2 L$

$4 \times 10^6 = 2 \times 10^6 \log_2 L$
$2 = \log_2 L$

$\Rightarrow L = 4$

*Performance Metrics:

① Bandwidth ← (bits per second) ← data rate يعني
② Throughput
③ Latency (Delay)
④ Bandwidth – Delay Product

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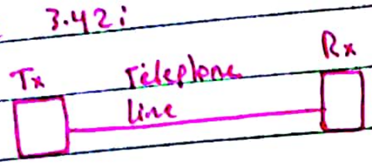
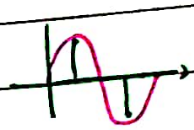
① Bandwidth :

Example 3.42 :



Bandwidth of the line 4 KHZ. but using the second definition the bandwidth is



$4 \times 10^3 \times 2 \times 8$

$\frac{\cancel{\text{Cycles}}}{s} \times \frac{\cancel{\text{Sample}}}{\cancel{\text{Cycle}}} \times \frac{\text{bits}}{\cancel{\text{Sample}}}$

$= 64$ Kbps

يعطي 2 Samples

اقدر ابعث ال Signal بضبط ازيدها

كمان بس رح يزيد ال Bandwidth

② ThroughPut :

↳ كم وصل للـ Receiv. بنجاح

Example 3.44

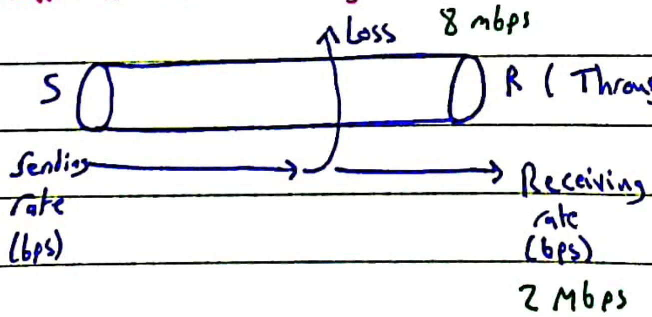
* Bandwidth = 10 Mbps.

وصلت الطرف الثاني (Throughput)

* Can pass only 12,000 frame per minute.

* Frame Size = 10,000 bits (average)

What is the throughput ?



Sol. $T_n = 12{,}000 \frac{\text{frame}}{\text{min}}$

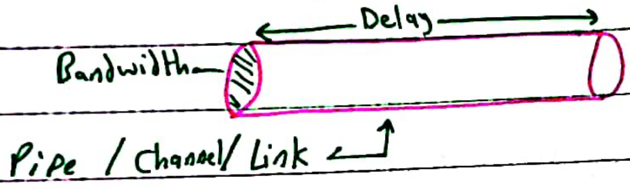
$= 12{,}000 \left(\frac{1}{60}\right) \left(\frac{10{,}000}{1}\right)$

$= \frac{\cancel{\text{frame}}}{\cancel{\text{min}}} \times \frac{\cancel{1\,\text{min}}}{60\,\text{s}} \times \frac{\text{Size of frame (bits)}}{1\,\cancel{\text{frame}}}$

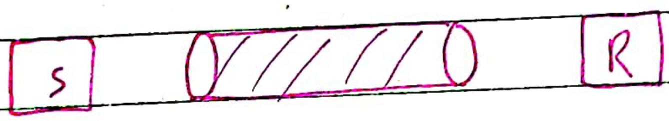
$= 2$ mbps

$\boxed{2/4}$ 3.31

Bandwidth - Delay Product



Volume = Bandwidth X Delay (bits)



* الفائدة بنقدر نحسب كم حجم ال bits الطايرة الي عندي .

Data Communications and Networking Fourth Edition

Forouzan

1 1 1 1 0 1 1 1 1

هون ال Avg رح يرتفع ويصير كبير

مرح يصير عندي يخربط فبدي احول

(Digital-To-Digital) من

Chapter 4

Digital Transmission

S → R

+5v

0v

2.5 V

فوق 2.5 بعتبره 1

تحت " " 0

$\bar{S} = \frac{1}{T} \int_T s(t)\, dt$

Avg.

تحويلات ال Signal

↳ Digital to Digital

↳ Digital to Analog

وهكذا.

4.1

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* ابعت bit rate اعلى

* تحسين ال features تبعت ال Signal

اتفاد الرسالة عشان توصل احسن

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

4.2

Line Coding

بسولنا 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 $\boxed{\text{sender}}$ data elements are encoded into signal elements
- At the receiver, signal elements are decoded into data elements

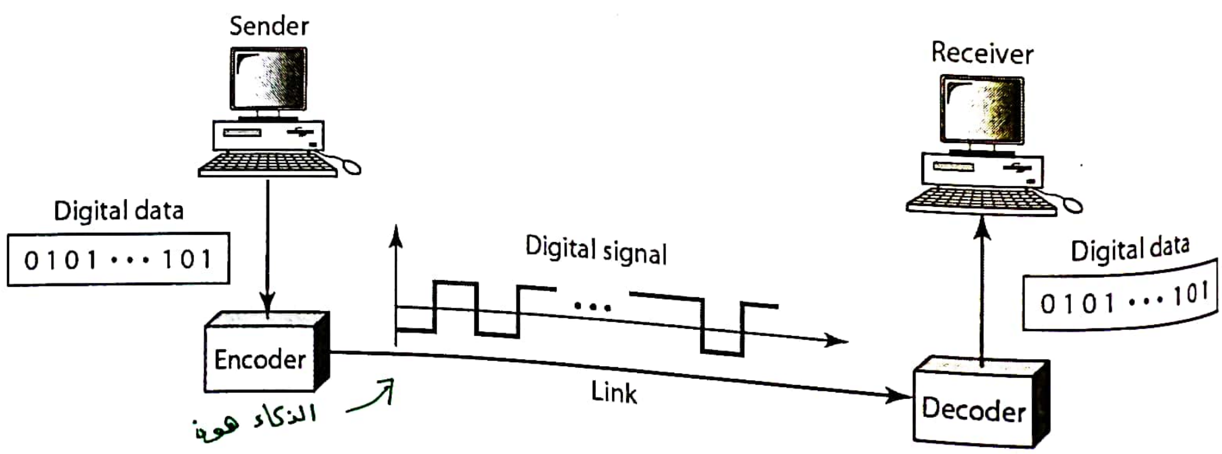
العملية العكسية.

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 ال bandwidth وهكذا.

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

Figure 4.1 *Line coding and decoding*



الذكاء هون

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Characteristics of line coding

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

1 ▪ Data Element vs. Signal Element

2 ▪ Data Rate vs. Signal Rate

3 ▪ Bandwidth → بمعنى ال (Hertz) (حزمة من الترددات) مش bps.

4 ▪ Baseline Wandering

5 ▪ DC Components → * غير مرغوب بال DC في ال Signal وال line coding
* ال frequency للـ DC component = 0.
* لو ال digital signal فيها DC component بدي أرسلها ع ال channel ال B تبعي مابعرف zero frequency حيصير عليها blocking فلازم نحولها من digital ل digital تاني مافيه DC component

6 ▪ Self-Synchronization

7 ▪ Built-in Error Detection

8 ▪ Immunity to noise and interference → ممانع.

9 ▪ Complexity → يكون بسيط.

* ال channels الي مابتمرر DC component مثلاً ع ال transformer (كذا السيركت كهربائياً).

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)

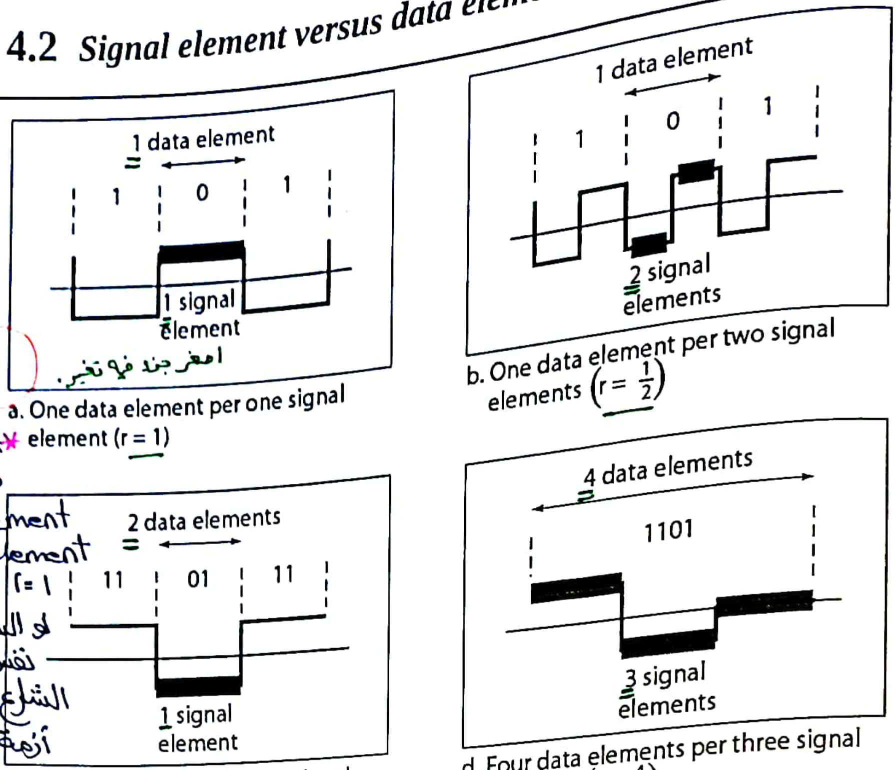
Signal Elements vs. Data Elements

أقصر إشي ممكن يحمل Information.

- 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 → $r = \frac{\#\text{data elements}}{\#\text{Signal elements}}$ ↓ كلما كانت أكبر كانت أحسن.

* ممكن نحمل Data element على Signal element, (بالدفش).

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 ($r = \frac{4}{3}$)

data عدد ال = r
element نسبة
Signal الى ال
element.

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

اعرف جد فهو تغير.

* افرض انا بسيارة وراكبين فيها أشخاص ، الأشخاص data element والسيارة هي Signal element ، لو شخص واحد اللي راكب r = 1 ، ولو شخصين r = 2 ، لو السيارة وراها قاطرة وبيسوق نفس الشخص بتصير $r = \frac{1}{2}$. الشارع هو ال channel . فالي بسوق أزمة السيارة مش الأشخاص فال Signal rate هو اللي بأثر.

4.7

* هو اللي بأثر على ال bandwidth (يهمني يكون قليل جداً).

Data Rate vs. Signal Rate

* يهمني يكون عالي.

تدي بعث data elements.

- The data rate (or bit rate), N, is the number of data elements (bits) sent in one second

how many ← bps انت قاعد بتبعث.

- 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 ←

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

how many* signal element بتبعثها بالثانية.

عشان ال bandwidth.

Signal rate * data rate

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

لـ عامل بعتمد على case معين اللي حتقوم بارسالها.

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

لو حد c ذهنا $\frac{1}{2}$

$C \in [0,1]$ حسب ال case.

$N = \frac{1}{c} \times S \times r$ ∝B بتنتج $\log_2 L$

4.8 Case Factor average $C = \frac{1}{2}$ ⟶ $= 2 \times B \times \log_2 L$

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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?

$c = \frac{1}{2}$

Solution

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

Singal rate

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

* كلما r بتزيد بنزل ال Signal Rate *

The Bandwidth

لأنو ال Information بتكون Random

- 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 عال low frequencies بنزل ال Amplitude بال fourier transform وهذا جيد لأنه 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

لأنه بننسى الترددات العالية بس بناخذ الي بـ range معين

- The bandwidth is proportional to the signal rate

متناسب.

الي Power العالي بتركز فيه بناخذهم.

- Given N: min bandwidth $\boxed{B_{min}} = c \times N \times \frac{1}{r}$

max bit rate
اي ممكن تحمله كل هذا ال bandwidth.

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

* كلما بزيد ال Data rate بزيد ال Bmin *

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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 elementالـ

bit rateالـ .

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

دائماً بنتعامل معه

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

* الـ reciever لما يستقبل الإشارة بحدد اذا هي 0 ولا 1 فبحسب الـ avg الـ Power في الـ Signal (running/time avg)
↓ base Line

تجول ←

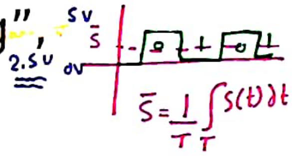
Baseline Wandering

اي اشي فوقها ← 1
اي اشي اقل ← 0

- 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

* ممكن يسبب مشاكل اذا اجا 1 كثير او 0 كثير للوضع التالي

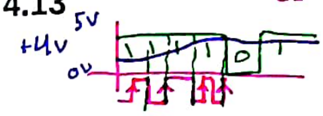
- Long strings of 0's or 1's can cause a drift of the baseline, called "baseline wandering", which makes it hard for the receiver to correctly decode the signal



$\bar{S} = \frac{1}{T}\int_T S(t)\,dt$

- A good line coding scheme should prevent baseline wandering ← لازم تكون بتعالج هاي النقطة

اختل عندي الـ baseline فما بقدر اقرا الـ Signal بشكل صحيح



Dc component لازم ما يكون فيه good line coding عشان مرات الـ transmission line بتكون طويلة فبالغالب بكون فيها low pass effect ومرات بحتاج أستخدم transformer (أفصل أجزاء كهربائية بالسيركت عن بعضها) فهذا بعمل block للـ DC ، أما بال LAN عادي بيكونوا في DC component.

DC Components

f=0 بتعيد حالها

بعدنا نبعد انه يكون كل 0 او 1
ممكن ما اعتمد على 0v ، 5v
اعتمد هل انا بطلع او بنزل

- 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

مثلا
هون بخلي الـ 1 على +5v والـ 0 على -5v

- A good line coding scheme should have no DC components

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

متى تبتدأ ال bit ومتى بتنتهي ال bit.

Self-synchronization

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

وعكسه ← asynchronized ما بعتمد clocks

كلما تزيد المسافة كلما بتصير معقدة أكثر

- Bit duration, the start, and the end of the bits should be exactly (معرف) identified by the receiver

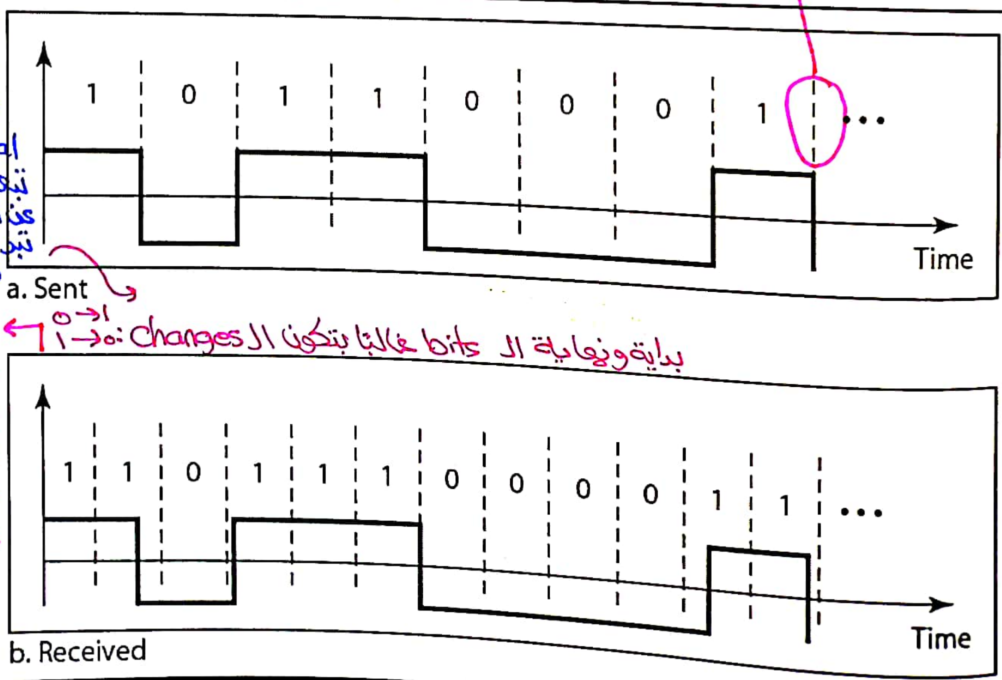
بعتمد S start and stop bit.

- 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 تبعث معها معلومات عن ال Clock أو متى تبدأ ال bit ومتى تنتهي



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

تغيير الفترات فتأثر استقبال ال bits ، فاعد بستلم data مختلفة تماما عن اللي تم إرسالها.

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Example 4.3

← مثال بيتلك 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

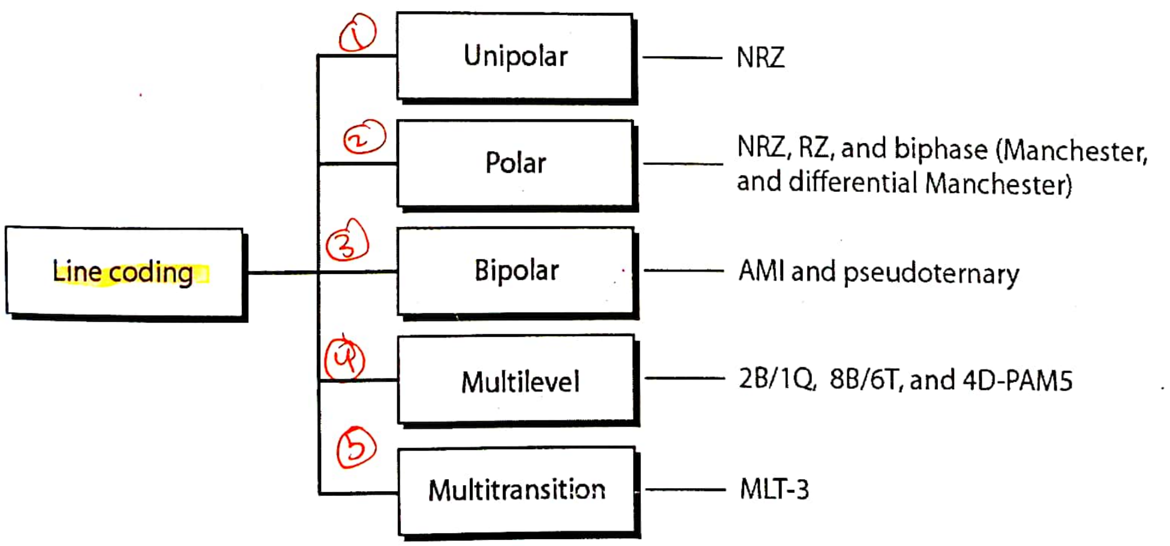
At 1 kbps, the receiver receives 1001 bps instead of 1000 bps.

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 ال "



لا من ٧ الى ٥ ولا من ٥ الى ٧٠.

لأنك خلال فترة استقبال الـ bit ما بترجع لـ Zero بتخلك تنفس level الاستقبال الـ Signal

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

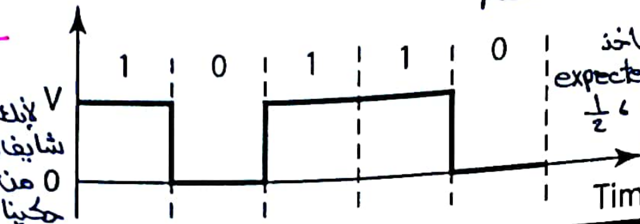
$*P = \frac{V^2}{R} \rightarrow R = 1\Omega \rightarrow P \cdot V^2$

Arithmatic Mean ← $* \bar{X} = \sum_{x} x P(x)$

← مشاكله:

1- Baseline wandering → لأنك شايفهون Snapshot من لحظة معينة بس حكينا انه في cases، فلو اجا مثلاً Sequence طويل من الـ 0's حتستقبلوا كانوا 1 ولو DC كله ٠٧.

2- بتصرف Power



بناخد الـ expected value ½، بتكون فوق و تحت ½

$$\boxed{\frac{1}{2}V^2 + \frac{1}{2}(0)^2 = \frac{1}{2}V^2}$$

Normalized power (avg)

- 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 → لأنه مابترجع لـ 0 خلال فترة إرسال الـ bit
- Relatively very costly scheme in terms of normalized power → مكلف بسبب الـ Power
- Hence, not a popular scheme nowadays ← مش منستخدم حالياً كثير

4.19

مثلاً هنا كل رح ابعث ٥ وراء بعض مثلاً

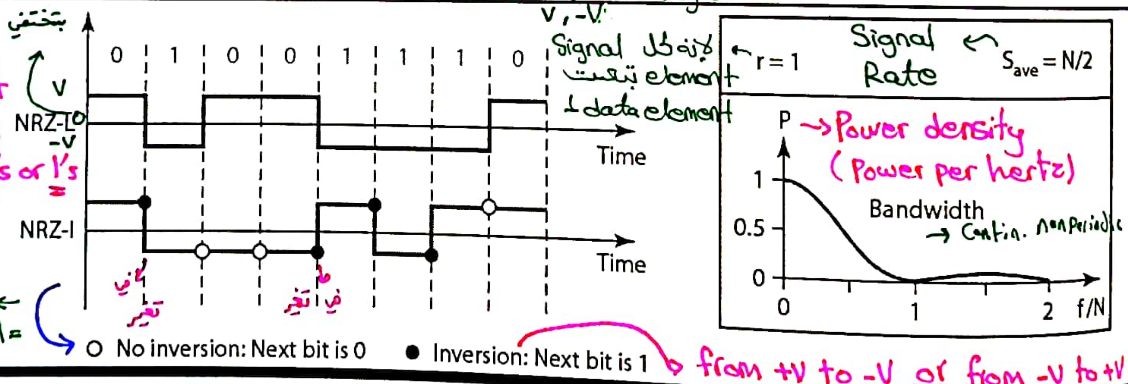
*logic 0 = V / logic 1 : -V   A: ?   B: ?

Figure 4.6 *Polar NRZ-L (NRZ-Level) and NRZ-I (NRZ-Invert) schemes*

↳ because we have +, - Voltages

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

← اذا الـ Next bit = 1 بعمل inversion ولكن اذا 0 مابعمل Inversion (يعتمد على الـ 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
- Both have the DC Component problem

خففنا مشكلة الـ Baseline wandering في حالة الـ sequence of 1's

اذا فيا لازم نستخدم NRZ-I احسن !!

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

4.20

بيعانوا من مشكلة ال .Sync فبدي يكون عندي معلومات
اكتر من ال Clock الي عندي ال Sender .

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

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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.

$S_{avg} = C \times N \times \frac{1}{r}$

$= \frac{1}{2} \times N \times \frac{1}{\frac{1}{2}} = N$

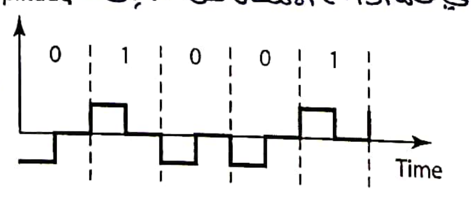
← لإنه أثناء فترة إرسال ال bit انت بتنتقل من V ل V أخرى. بتكون ب V بترجع ل 0 او العكس من V- الى 0.

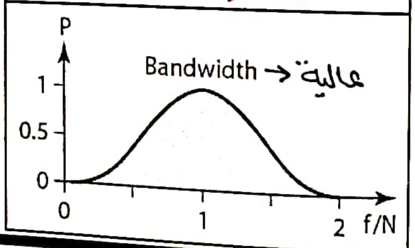
كل 2 Signal element بيعث 1 data element وهنا بزيد ال Bandwidth

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

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

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





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

* ← لإنه صرت اعرف متى برسل وبستقبل ال bit من خلال ال change تبع ال Signal لإنه مجرد ما يصير بعرف إنه هو حاليا بنص ال bit

* أصعب على ال reciever تصميمه.

(LAN)

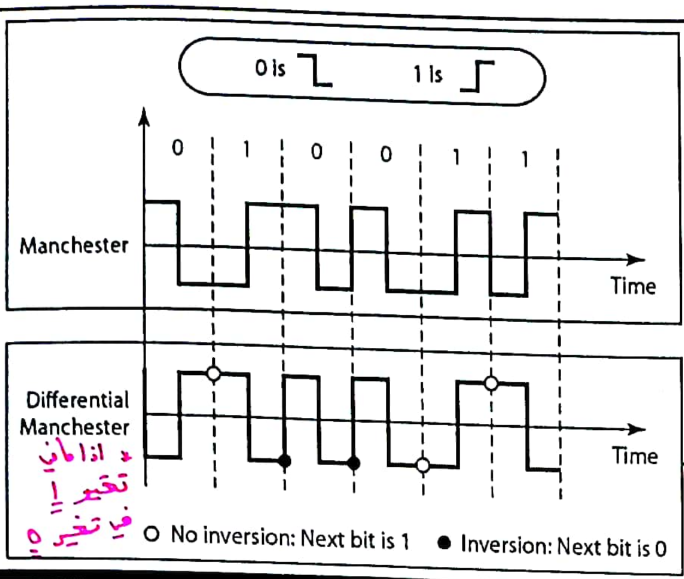
مستخدم في ال Ethernet

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

في return-to-zero

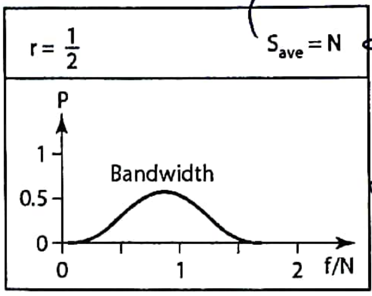
(انتقال في منتصف ال bit)

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



إذا ما في تغير 1

في تغير 0



(مناسب نستخدمه في ال LAN فقط لإنه ال Bandwidth كثير عريض فبمرق ال Data.

أقل Power بس لسا ال Bandwidth عالي.

"2 levels of Signal"

(عنصرين 3)

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

اذا بعث 0's او 1's ما في DC Component

و عشان في trans. بنص ال bit بحل مشكلة ال Sync.

4.25

* ما بقدر استخدمه على WAN لانه بيتطلب bandwidth كبير.

Note *

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

Note

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

هاي تعتبر سيئة.

4.27

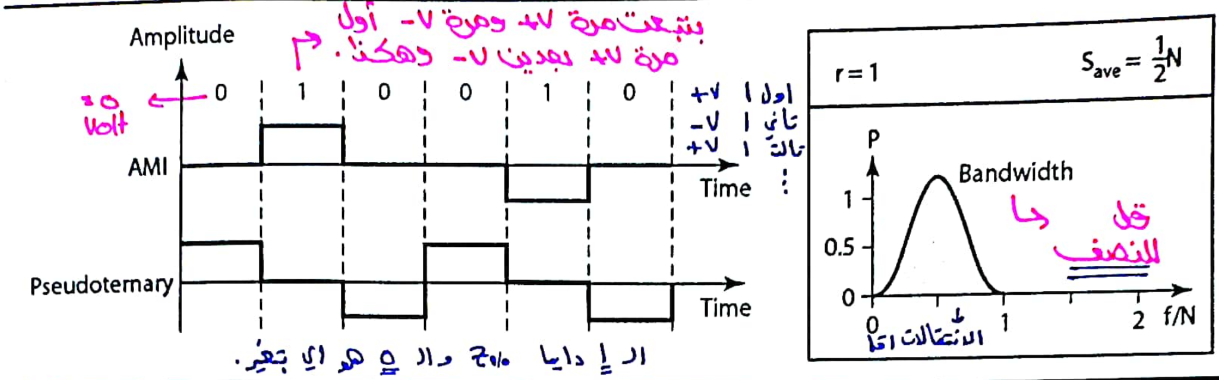
Note

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

4.28

لإنه 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. → AMI عكس
- No DC component. Why? → because avg = 0
- What about synchronization? → AMI → بحلها لو كانت 1 لو 0 لا
  → Pseudoternary → بحلها بمرحلة ال 0 بدال 1 لا

4.29

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

* ال Synchronization بتكون مضروبة بالعادة لما يكون كل ال Signal 0 أو 1 لإنه بضيع لما عندنا معلومات عن ال clock.

Multi-Level Line Encoding Schemes (mBnL)

بداية ونهاية ال bit

- 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 بيبعت 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

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Note

In *m*B*n*L 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

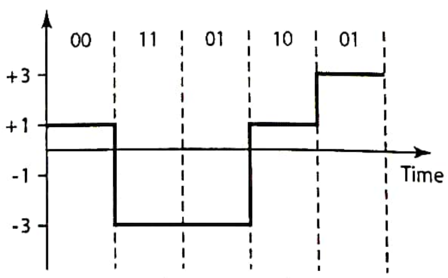
2 data element encoded in 1 SE

L=4 levels

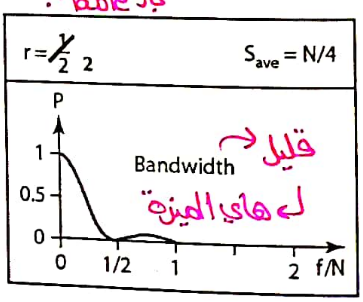
Figure 4.10 *Multilevel: 2B1Q (2 Binary 1 Quaternary) scheme*

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

Transition table → لازم نلتزم بال table.



Assuming positive original level



مشكلة

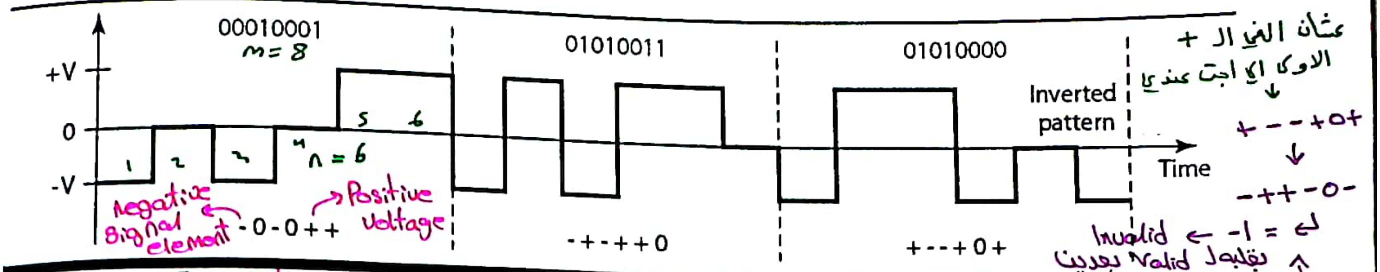
- Faster data rate. Why? because r=2
- L = 4 → complex receiver & $2^2 = 4^1$ → no extra signal patterns
- What about the DC Comp.? Not balanced (scrambling used to help)
- What about synchronization and baseline wandering? (same)

↑L → ↑complex receiver

لسا موجودة

4.32

Figure 4.11 *Multilevel: 8B6T (Eight Binary 6 Ternary) scheme* يستخدم في ال Ethernet



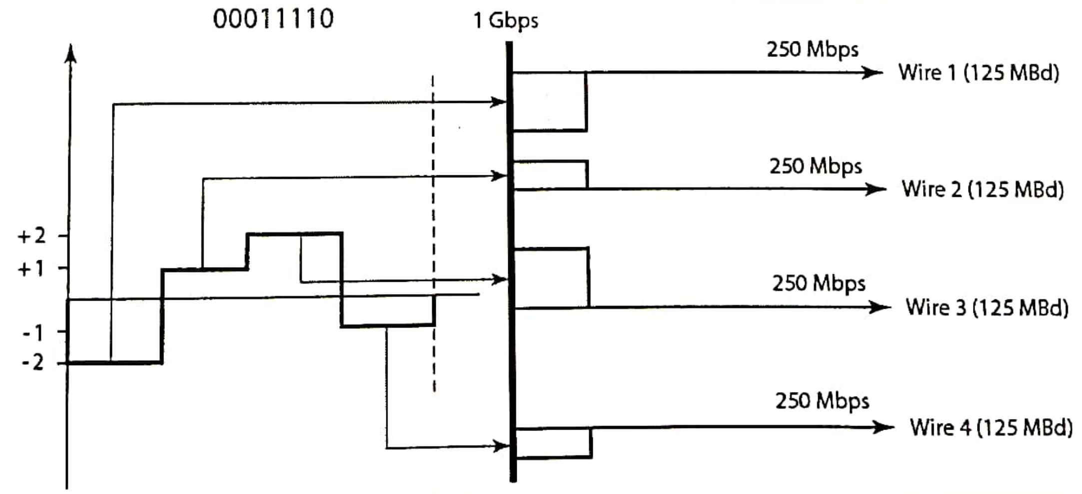
- $3^6 - 2^8 = 473$ extra signal patterns ➔ good synchronization and error detection
- Each pattern has a weight of either 0 or +1 DC values
- If two consecutive patterns with +1 weight, the sender inverts the second pattern. Why? to reduce baseline wandering
- How does the receiver recognize the inverted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

* ال receiver اذا لقى منه (-) بعرف انه عندي وحدة مقلوبة فبعكسها وبترجع زي ما كانت.

* لتقليل ال Baseline wandering.

* بـ Appendix D موجود الجدول اللي بمثل كل ال 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

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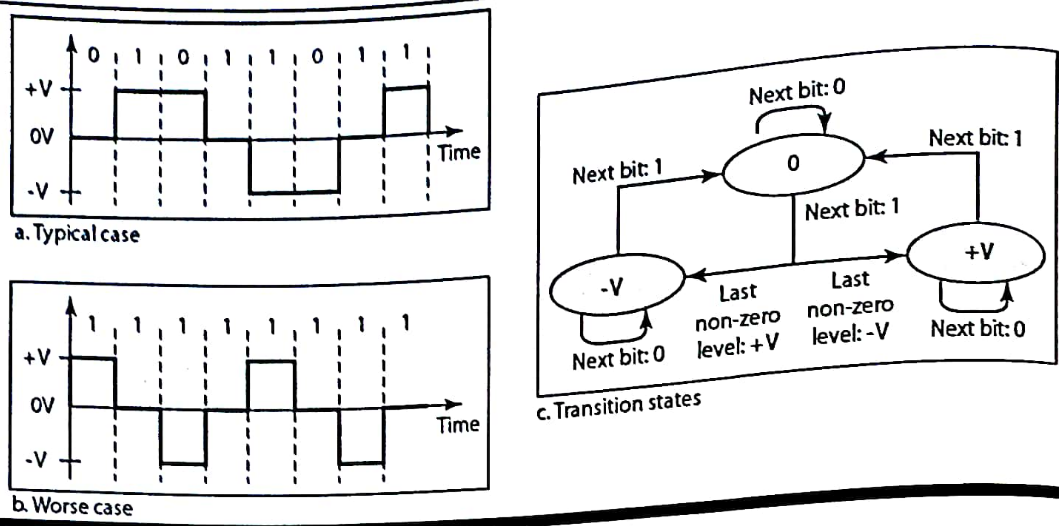
Figure 4.13 *Multi-transition: MLT-3 scheme*



- 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

4.35
- ➔ $S_{max} = N/4$

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

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m bits بعمل عملية استبدال للـ*
بـ n bits حيث n>m →

Block Coding

- 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)

  > m

- This is referred to as *mB/nB* coding

لعملك 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.

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Steps in Block Coding Transformation

- Step 1: Division
  - The bit stream is divided into groups of m bits
- Step 2: Substitution

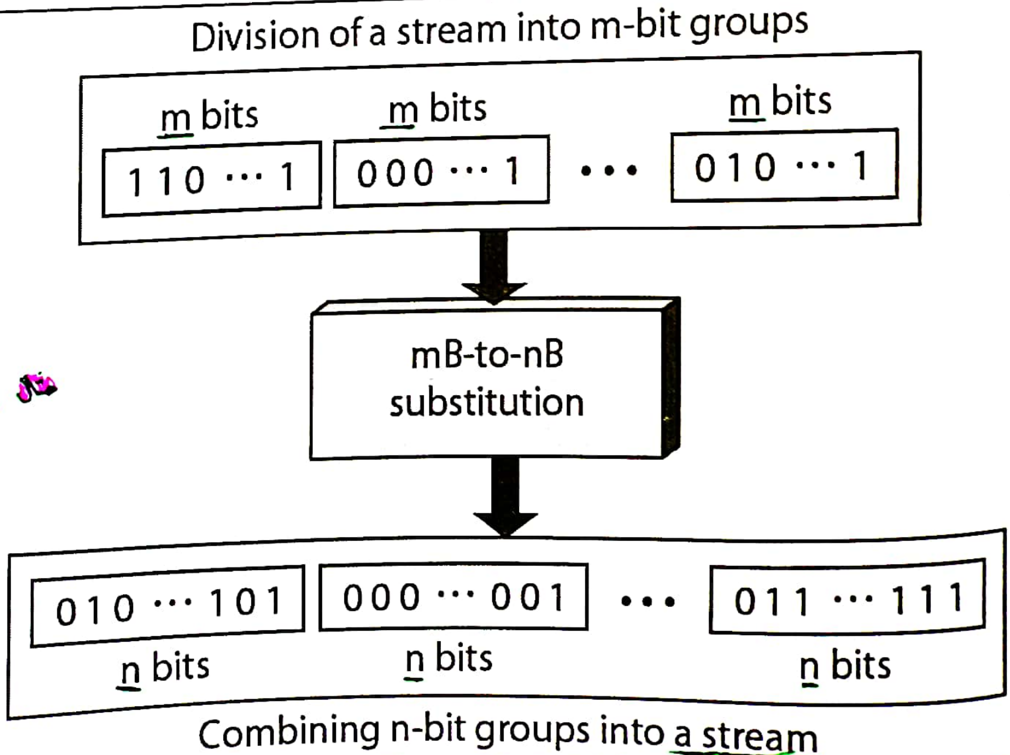
  n-bits من ال Pattern
  - The m-bit groups are substituted with n-bit codes, where n>m
  - A number of n-bit codes are carefully chosen to ensure that the synchronization and error detection are achieved
  - Notice that at most only one half of the n-bit codes are needed. Why? because n is most larger 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

لـ بتحولوا ال Block coding

رح يزيد ال Signal rate.

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) لو عملنا هيك تعديل بولع
  - 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

4 bits

Sub

5 bits

? → 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*

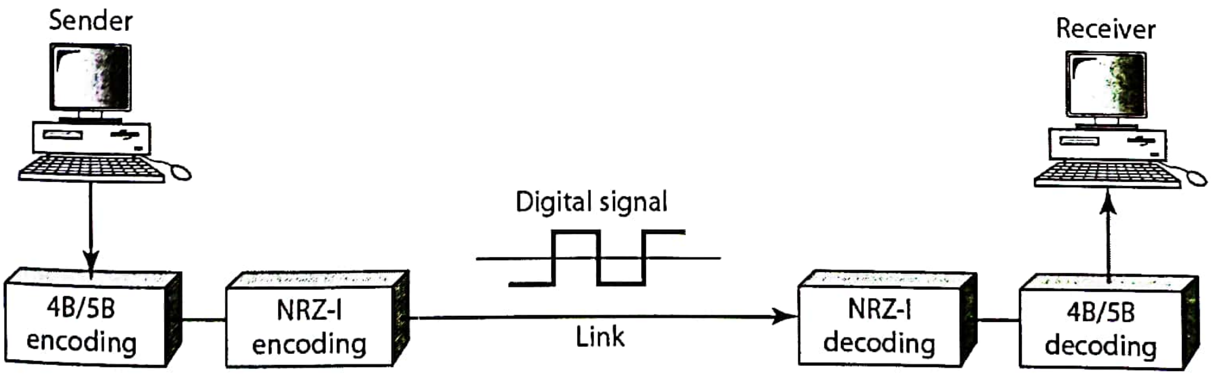


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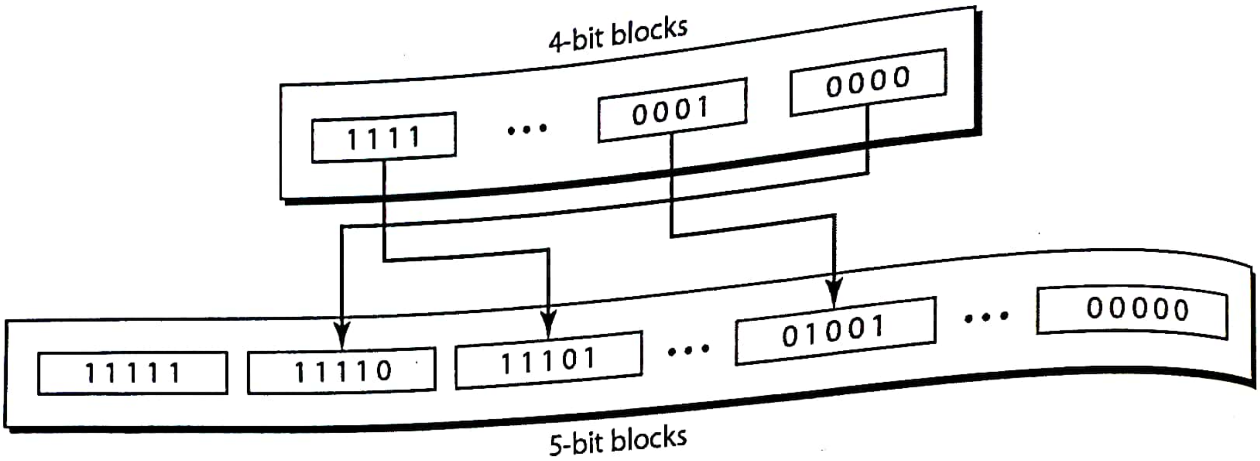
Figure 4.16 *Substitution in 4B/5B block coding*



111 00 0 1001
trailing leading

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

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

لم يتم اختياره بشكل عشوائي.

Table 4.2 *4B/5B mapping codes*

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

أرسالهم مش متكرر وفالنا ما برسلوا ورا بعض.

4.44

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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?

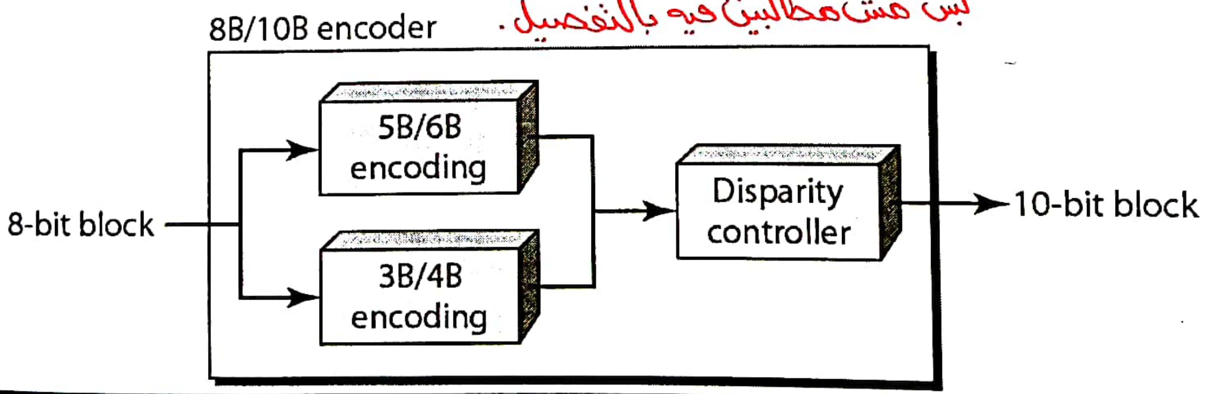
Solution

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 نعمل 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

كل شيء صايرين نحوله لـ Digital
حتى الصوت وهيك.

Topics discussed in this section:

Pulse Code Modulation (PCM)
Delta Modulation (DM)

4.47

لازم نحول الـ digital الى digital
من نوع اخر
الصوت بحوله لـ Digital تا
ادخله عالحاسوب.

* لما احول من Analog لـ Digital
بصير عندي losses.
Quantised level.

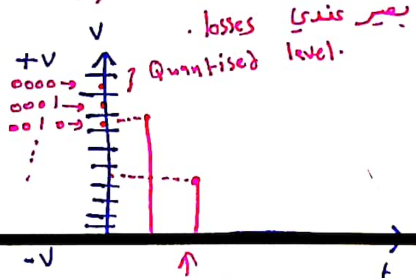
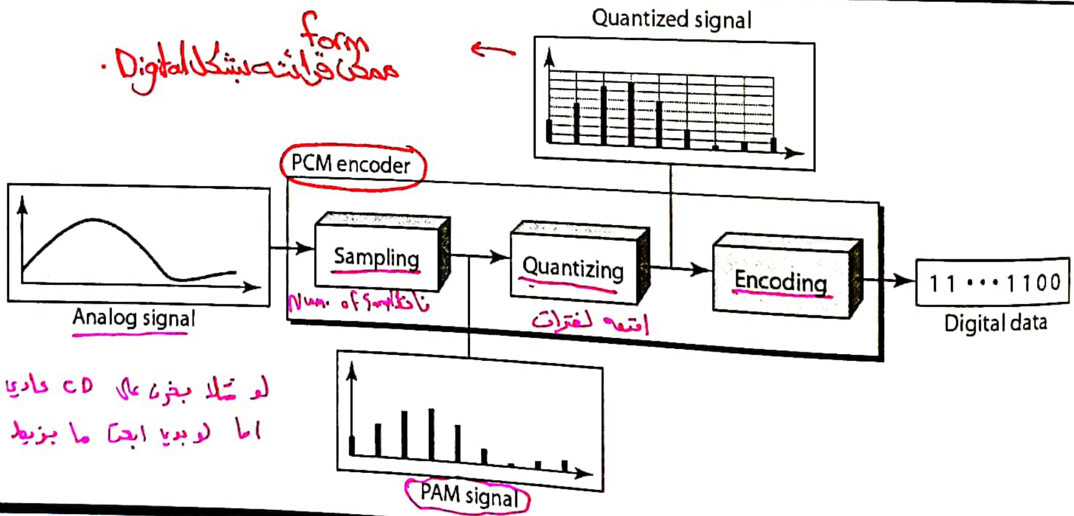


Figure 4.21 *Components of PCM encoder*

ممكن قرائته بشكل Digital form.

Composite Signal



(دقة)
لو شلنا بجزء على CD عادي ازيد الـ Quan. level
اما لوبدي ابحا ما بزبط

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

4.48

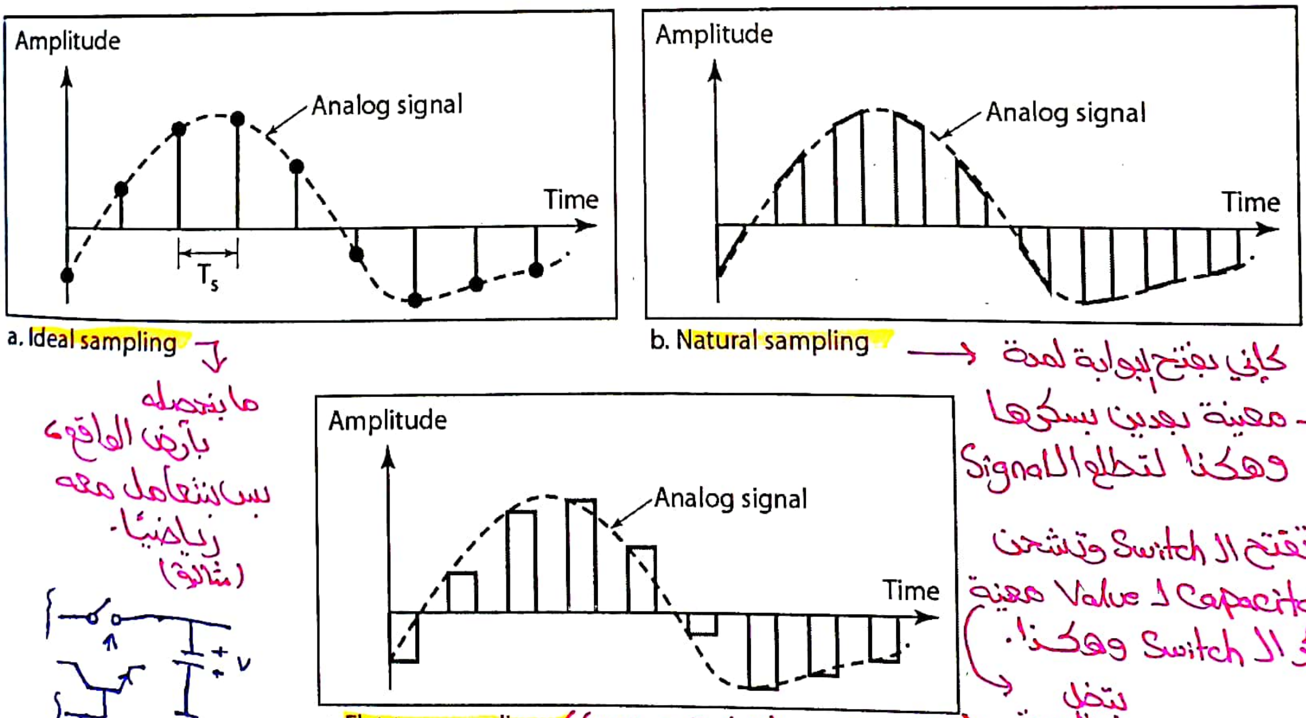
Sampling: Definition and Background

- 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

لما اضخم الاشارة بضخم الـ noise كمان.

* إذا ما قدرت تعرف أعلى frequency موجودة عندك بالـ Signal ماح تعرف شو الـ Sampling Rate.

Figure 4.22 *Three different sampling methods for PCM*



(Sample and hold) مستخدم أكثر.

Ideal sampling: ما بنحصله بأرض الواقع، بس نتعامل معه رياضياً. (مثالي)

Natural sampling: كأني بفتح بوابة لمدة معينة بعدين بسكرها وهكذا لتطلع الـ Signal — T_s

أو تفتح الـ Switch وتشحن الـ Capacitor لـ Value معينة وتسكر الـ Switch وهكذا. — بتضل ثابتة

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Sampling Rate: Nyquist Theorem

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

لـ كلما كان اكبر كان ال bandwidth اعلى وهيك احسن واحسن.

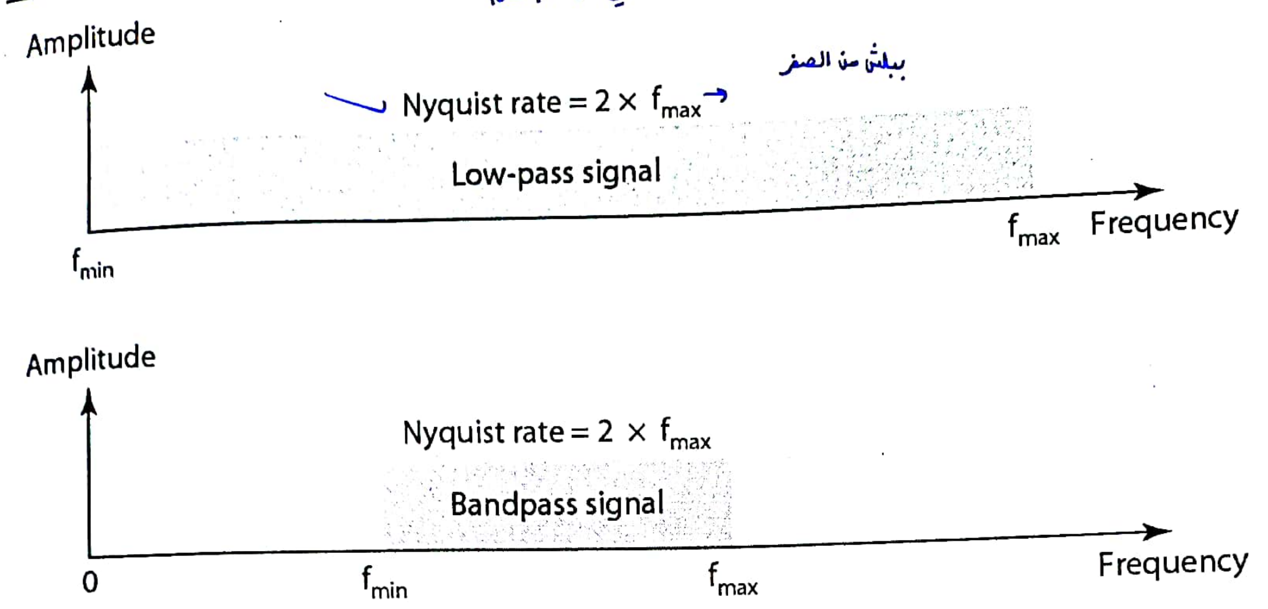
- Nyquist Theorem: the sampling rate should be at least twice the highest frequency component in the original analog signal

4.51

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

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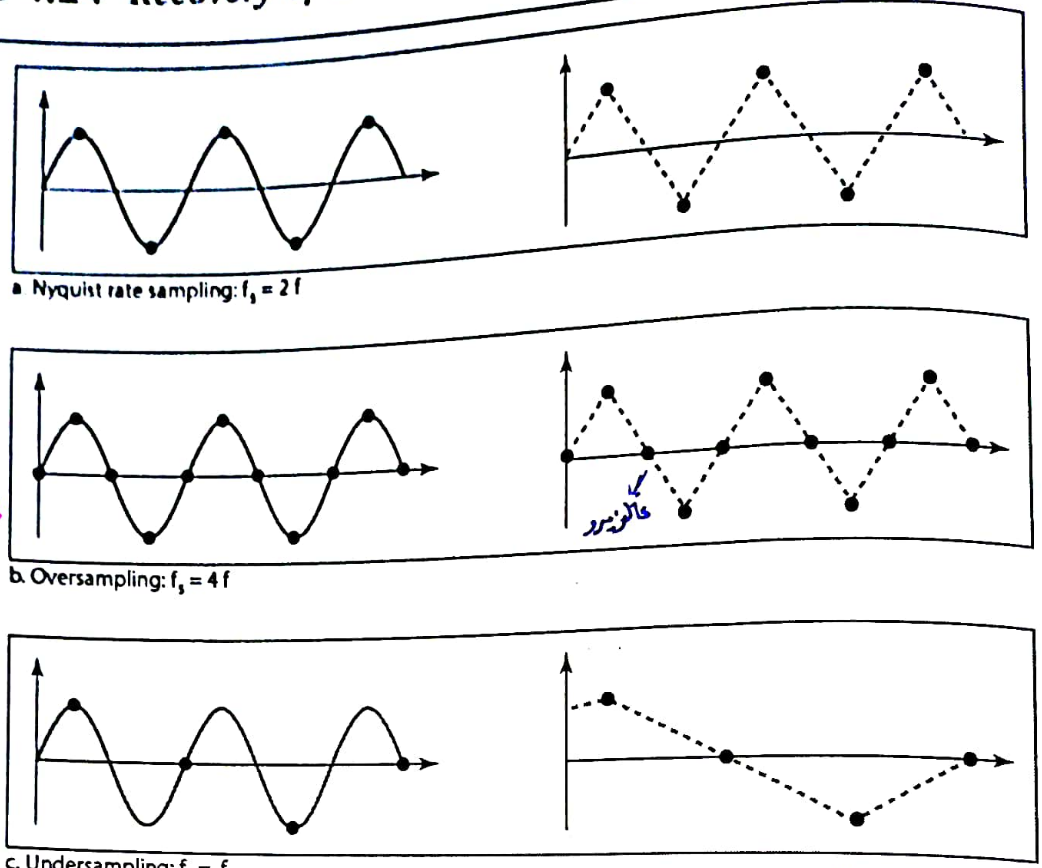
Figure 4.24 *Recovery of a sampled sine wave for different sampling rates*



← بتقدر توصلنا للـ Original Signal.

← ممكن ما بكون مفيد لأنه باخد زيادة Samples ما أثرت عالشكل. (الي طلعت عال zero)

← مابتقدر توصل للـ Original Signal.

مثال كتاب صفحة 118 Figure 4.25 بالـ Undersampling صرت اعرف بالعكس لانه بشوف كل 45 ثانية

4.55

و Example 4.8

$f_r \geq 2f_{max}$ ✗

$f_r > 2f_{max}$

Example 4.9

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

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

Example 4.10

أكثر من frequency

مبلش 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.

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ch.4

$\boxed{13/4}$ Line of codding (summary)

B^m: number of combinations of data pattern.

L^n: number of Combinations of Signal pattern.

B : number of available data elements.

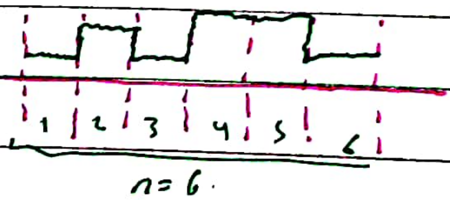
m: number of data elements in a data pattern.

L: number of ~~the~~ available Signal levels.

n : number of Signal elements in a signal pattern.

Data pattern. $\longrightarrow$ Signal pattern

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



$B^m \leq L^n$

Quantization

ممتاز في ال PCM ، كل ما بنزيد عدد ال bits تمثيل ال Signal الأصلية يكون أدق ويقل ال Quantization error بس الضريبة إنه ال bandwidth بزيد

- 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:

  لازم اكون عارف ال range تاع ال Signal

  - 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.

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

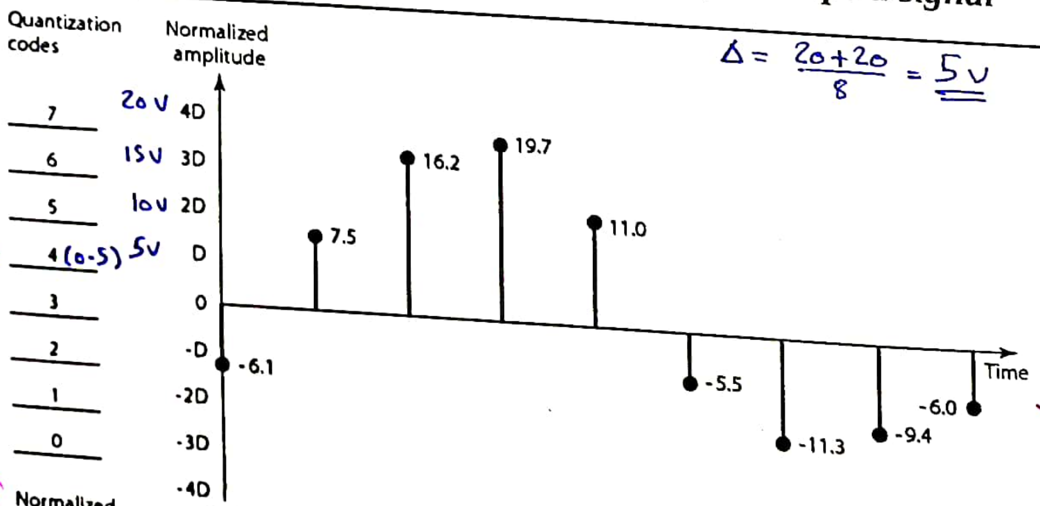
عدد النقاط

  - 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 Value تقع بين الفترة الصغيرة بقربها لل midpoint تاع هاي الفترة وهذا بعمل error.

Figure 4.26 *Quantization and encoding of a sampled signal*

$\Delta = \frac{20+20}{8} = 5V$



معطيات

- $V_{min} = -20$ v
- $V_{max} = +20$ v
- $L = 8$
- $\Delta = 5$ v

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

Normalized PAM values	-1.22	1.50	3.24	3.94	2.20	-1.10	-2.26	-1.88	-1.20
Normalized quantized values	-1.50	1.50	3.50	3.50	2.50	-1.50	-2.50	-1.50	-1.50
Normalized error	-0.38	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

Value مقسوم Δ

منتصف الفترة

$e = V - V_{mid}$

binary

error في هاي

error في

أعلى من ال Mid

أعلى من ال Mid

$[7] \to 2^3$

$2^{3} = 8$

bits

error = measured or estimated value − actual value

error = 3.5 − 3.24

$\boxed{error = +0.26}$

لأنه ال mid أكبر من قيمتي الحقيقية.

Quantization Levels and Error

- The number of levels, L, depends on:
  - The amplitude range of the analog signal → بعتمد على الموجة.
  - The accuracy needed in recovering the signal → ال (L)
- 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 $\Delta/2$ ($-\Delta/2 \leq$ 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

$n_b = \log_2 L$ ← عدد ال bits كبير

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

level بال ← (number of bits)

← كل ماتزيد عدد ال bits ال SNR بتتحسن، الممانعة لل noise بتكون أكبر لأنه.

* بعد ما احول ال analog الى digital نشوف شو تأثيرها على المعادلات الي قبل

Example 4.12

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

* لأحسب ال SNR لازم أحسب ال # of bits ، ولأحسب ال # of bits لازم احسب ال # 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 بزيد ال bandwidth.

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Example 4.13 (2014) دفتر

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.

→ Pulse Code Modulation

PCM Bandwidth

تعتبر عملية معقدة

- Consider a low-pass analog signal

بتبدأ من zero مثلا low-pass.

- 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)

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

$= c \times 2 \times B_{analog} \times n_b \times 1/r$

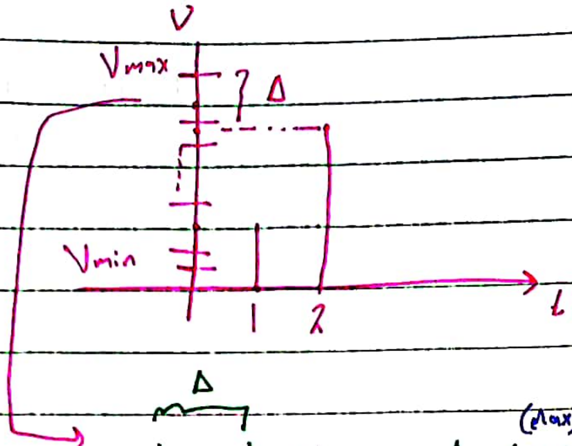
- When r=1 (for NRZ or Bipolar) and c=1/2,

كل ما ازيد عدد ال bits

لازم ازيد ال min ← bandwidth

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

$\boxed{20/4}$

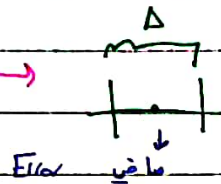


L : # of levels

Δ : quantisation period

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

كل ما زدنا L قلت Δ
وقلت ال error بس ال bandwidth
ح يزيد بسبب انو بدي عدد
bits اكبر.



$$-\frac{\Delta}{2} \leq \underset{(max)}{error} \leq \frac{\Delta}{2}$$

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

$$\underset{B_{min}}{S} = C * N * \frac{1}{r}$$

$$r = \log_2 L$$

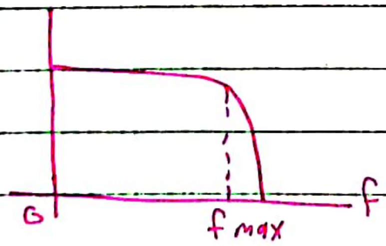
$$= 10 \log \frac{P_S}{P_{N_Q}}$$

$$e = V_{mid} - V$$

$$= 10 \log \frac{\frac{V_S^2}{R}}{\frac{V_N^2}{R}}$$

$$= 10 \log \left(\frac{V_S}{V_N}\right)^2$$

$$= 20 \log \frac{V_S}{V_N}$$



$$f_s > 2 f_{max}$$

$$f_s > 2 B_{analog}$$

$$B_{analog} = f_{max} - 0 = f_{max}$$

$$* \text{ Bit rate} = \frac{\text{Sample}}{\text{Sec}} * \frac{\# \text{bits}}{\text{Sample}}$$

$$f_s = \frac{\# \text{bits}}{\text{Sec}}$$

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.

CS CamScanner

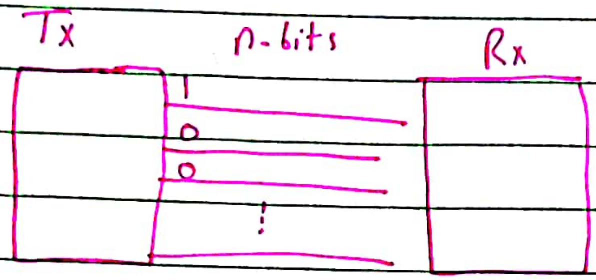
25/4

اذا T_b ثابتة الـ Parallel اسرع.

اما لو لعبت بـ T_b ممكن Serial اسرع

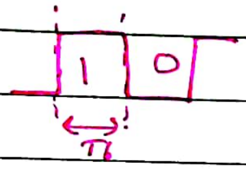
* Parallel Trans

$$\boxed{R_P = n R_s}$$



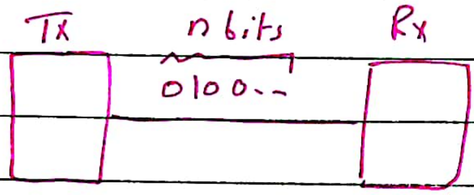
$\rightarrow$ n bits $\xrightarrow{in}$ T_b

$R_P = \frac{n \text{ bits}}{T_b} = n\left(\frac{1}{T_b}\right) = nR_s$



$R_s = \frac{n \text{ bits}}{n T_b} = \frac{1}{T_b}$

* Serial Tx :



$\rightarrow$ 1 bit $\xrightarrow{in}$ T_b

n bit $\xrightarrow{in}$ nT_b

Delta Modulation :

$\delta = \text{d}$, $S_\Delta(t)$: Signal at time t.

$\delta = S_\Delta(t) - S_\Delta(t-T)$

الفارق بين 2signals.

① Compute :

$$\delta = \begin{cases} \delta , & S(t) > S_\Delta(t-T) \\ -\delta , & S(t) \leq S_\Delta(t-T) \end{cases}$$

(above $S(t)$: الحالية; above $S_\Delta(t-T)$: اللي قبلها و delay)

② Time : (Next-bit)

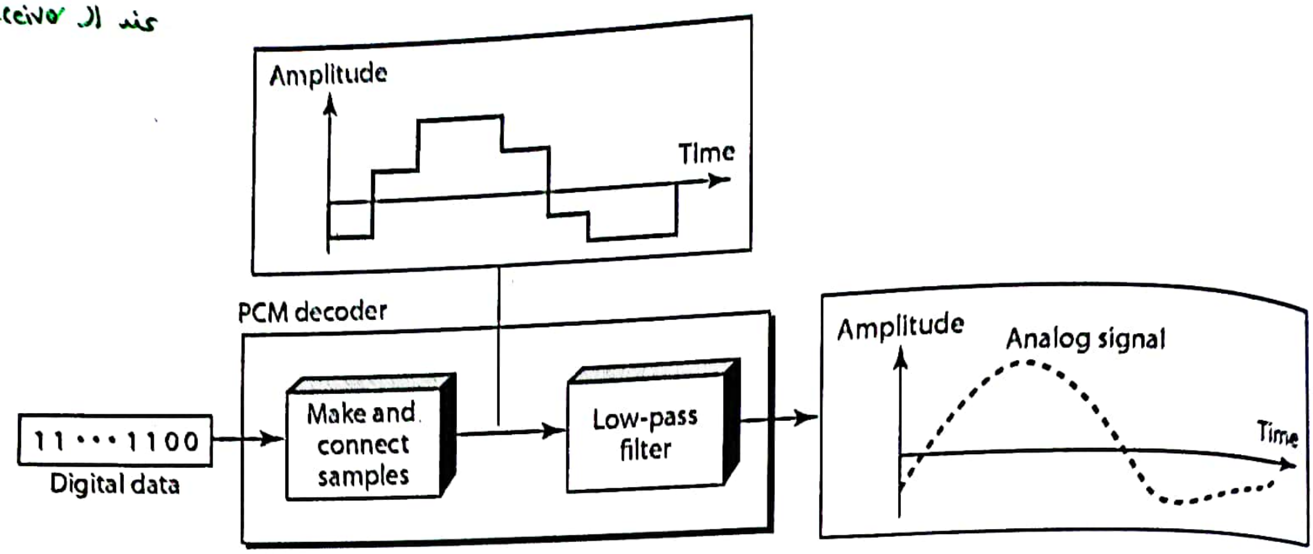
Digital bit اللي بنبعتها بأخرها $\leftarrow$ $D = \begin{cases} 1 , & \delta > 0 \\ 0 , & \delta \leq 0 \end{cases}$ D: Next-bit

③ Delay by T

④ $\underset{\text{Update}}{S_\Delta(t)} = S_\Delta(t-t) + \delta$

Figure 4.27 *Components of a PCM decoder*

عند ال receiver .



حلت مشاكل PCM

* هي الطريقة البسيطة والأسهل بدل ال PCM

Delta Modulation (DM)

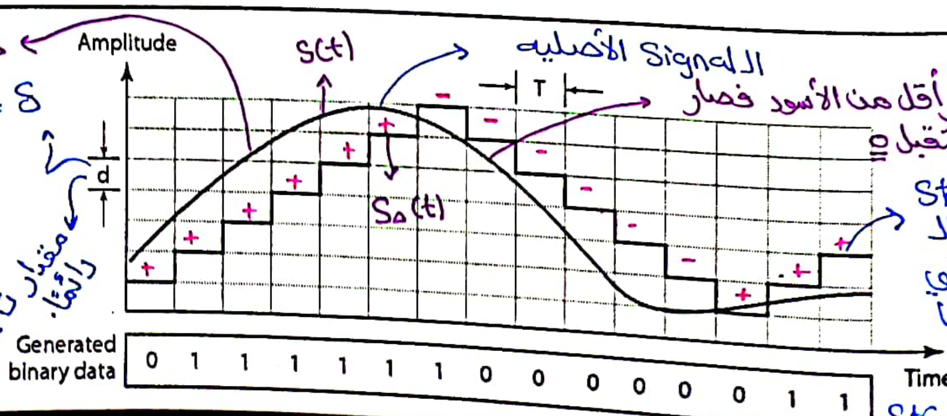
- PCM is a relatively complex A-to-D technique
- DM is a much simpler technique than PCM:
  - Finds the delta change of the current sample compared to the previous sample
  - If the current sample is larger, it sends a 1. Otherwise, it sends a 0

Figure 4.28 *The process of delta modulation*

طول ماهاي أعلى بضل أضيف 1 (موجب)

d = δ

مقدار ثابت دائماً



S(t)

ال Signal الأصليه

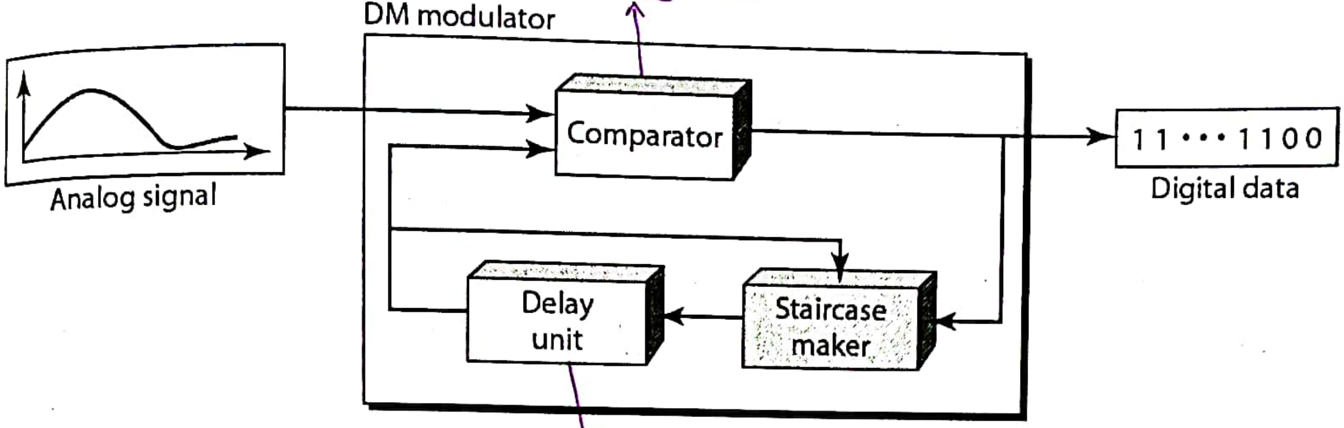
$S_Q(t)$

صار الأحمر أقل من الأسود فصار negative فبستقبل 0

Stair-case كأنها ظل ال Signal اللي بستخدموها لأطلع ال Streams of 0s and 1s

Figure 4.29 ***Delta modulation components***

بلزمك معلومتين الأولى
ناتج الطرح (Δ)



والثانية كشان يعرف T
شو لتساوي مشان ال Delay

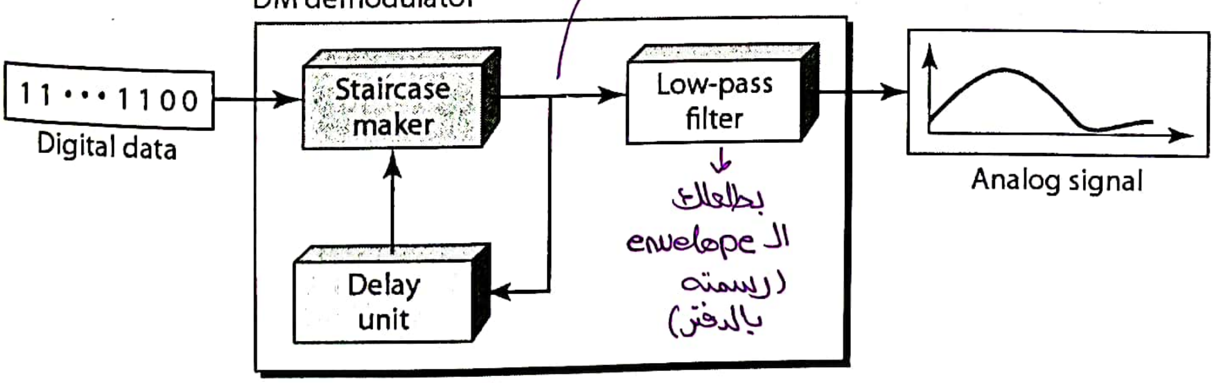
ابطل لما احول
من analog ل digital.

4.69

Figure 4.30 ***Delta demodulation components***

العملية العكسية:

بطلع ال Stair-case



بطلع 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.

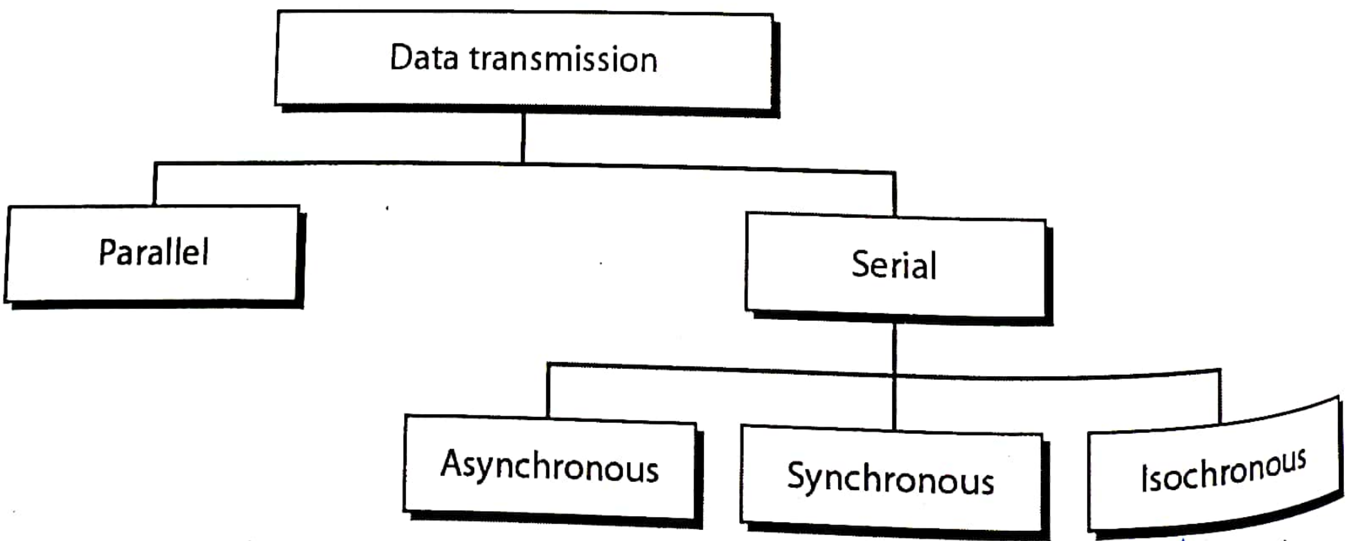
نوع ثالث
وجديد من
ال transmission

Topics discussed in this section:

Parallel Transmission
Serial Transmission

4.71

Figure 4.31 *Data transmission and modes*

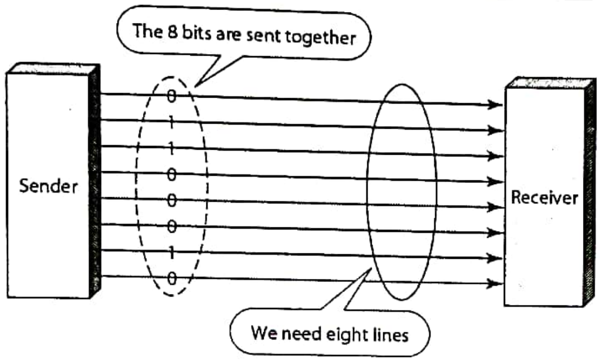


* Delay between frames not bits.

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 لوجود الأسلاك due to the extra wiring
- Usually limited to short distances → ماينفعه 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***



n wires

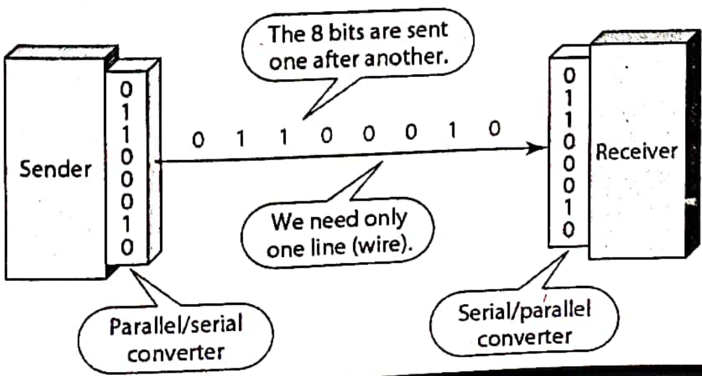
* مين أسرع ال Parallel ولا ال Serial ؟ الجواب بالدفتر

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***



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اللي 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 (كل byte لحاله .) (عشان التنين يحطوا مع بعض)
- 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.

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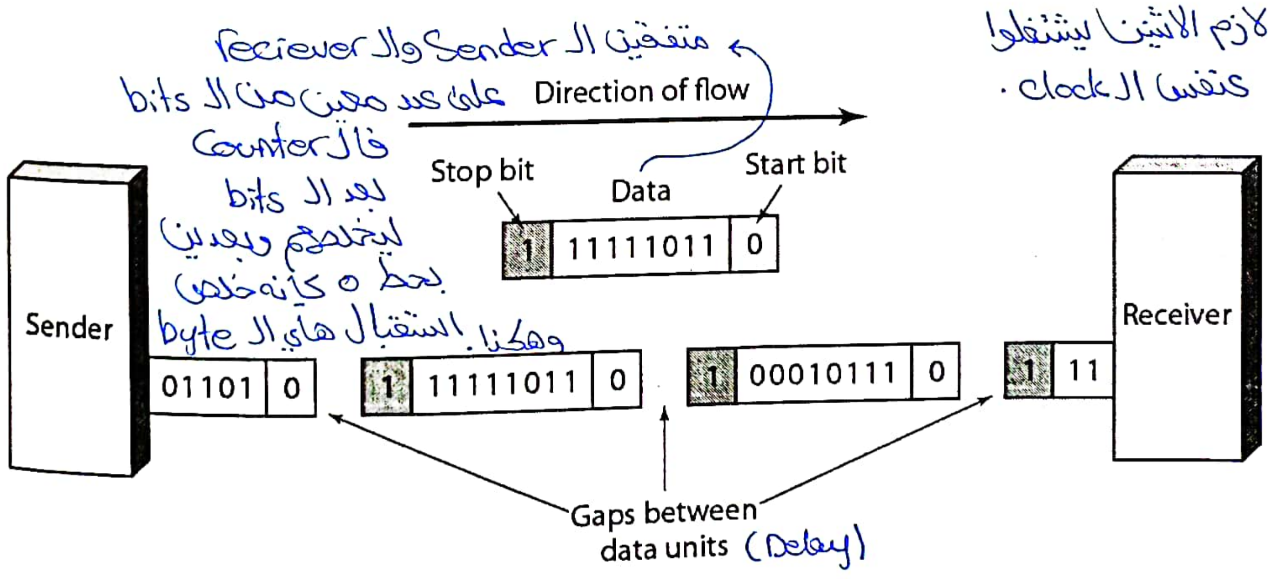
Note

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

4.77

Figure 4.34 *Asynchronous transmission*

* لازم الاثنين يشتغلوا بنفس ال 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 يبعت 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

لـ أسرع، مافي (لإنه bits زيادة لل 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.

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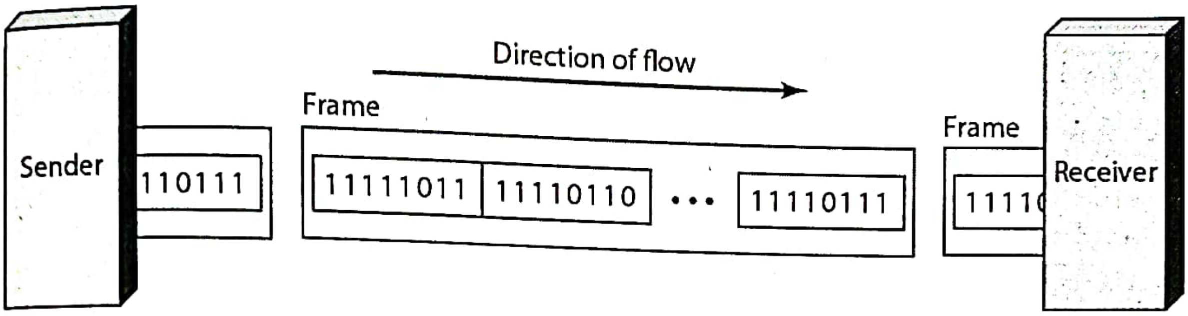
Figure 4.35 *Synchronous transmission*



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
  - The data is received at a fixed rate

    بين ال Frames اسمه Jitter
- Examples: real-time audio and video streaming

Data Communications and Networking Fourth Edition

Forouzan

Chapter 5

Analog Transmission

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5-1 DIGITAL-TO-ANALOG CONVERSION

* مثلا بدي ارسل لمسافة ابعد او ازيد عدد ال Users ضروري Analog
او بدي ابعث زي ال Radio ds Freq معينة عشان ما يتداخلوا

***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 إنو ال Channel تبعتنا محدودة ال bandwidth فما بنقدر نبعث ال Square wave الي فيها عدد هائل من ال Frequency.

Carrier → $S\_c(t) = A \sin(wc * t + theta)$ (θ)

A: يا بغير ال Amplitude — wc: $2\pi f_c$ او بغير ال freq — theta: او بغير ال Phase

هدول ال 3 characteristics الي بقدر اغيرهم عشان اوصل

Size of Antenna $\propto \lambda$

$\lambda f = c$

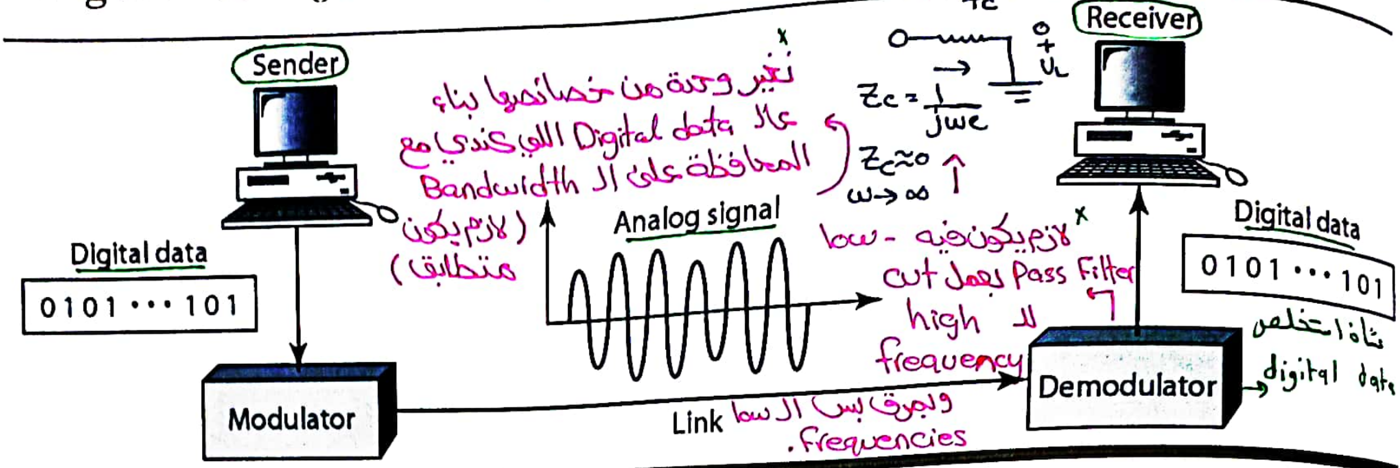
$\lambda = \frac{c}{f}$

كل ما قلت ال freq كل ما كانت ابعاد اوحجم ال Antenna كثير كبيرة ضروري ↳ Modulation.

Topics discussed in this section:

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

Figure 5.1 *Digital-to-analog conversion*



- 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)

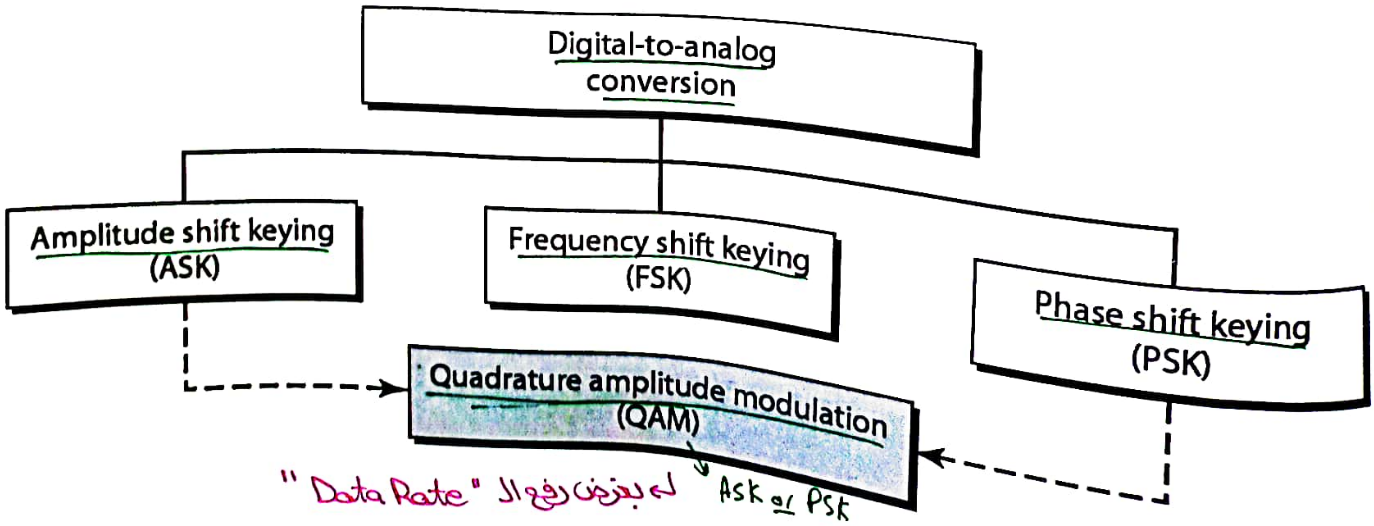
  بس تتدخل within the bandwidth of the link.
- The digital information can be carried as predefined changes to one or more of the characteristics (e.g., no-change = 0 and some-change = 1)

  تغيرات واضحة

  * ال Signal اللي بدي اغيرها اسمها Carrier Signal.

5.3

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



Aspects of Digital-to-Analog Conversion

العلاقة الي كانت تربط Data Rate مع Signal rate

$S = C * N * \frac{1}{r}$

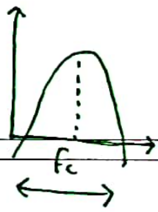
صارت هون $\boxed{S = \frac{N}{r}}$

$S \leq N \quad , \quad r \geq 1$

- 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 (s)
- Carrier Signal: ← مرتبطة بال Bandwidth
  - The digital data changes the carrier signal by modifying one of its characteristics (Amplitude, Freq, Phase)
  - This is called modulation (or Shift Keying)
  - The receiver is tuned to the carrier signal's frequency

ال Receiver بلتقط ال Carrier Signal وبتخلص شو مابديا

ممكن كل يوزر ابعت على Freq.



زي بالراديو لما اغير ال freq بقدر استقبل قنوات مختلفة

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.

r

S

N = ??

Solution

In this case, r = 4, S = 1000, and N is unknown. We can 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?

N

S

r = ?? L = ??

Solution

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

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

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

تغيير الـ Amplitude للـ Carrier بناء على الـ digital Signal .

Amplitude Shift Keying (ASK) = OOK

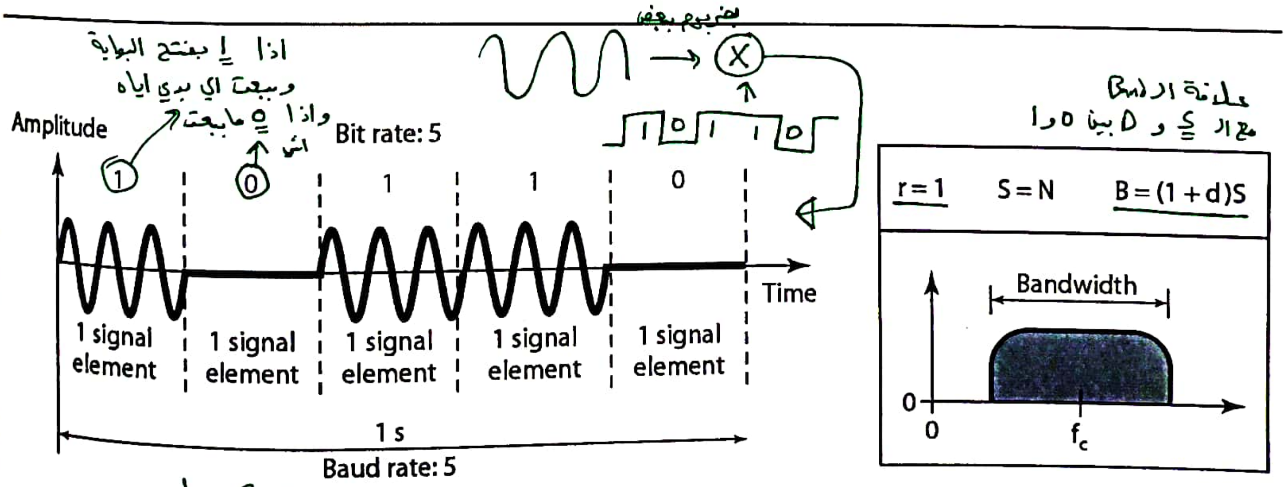
- ASK: varying the peak amplitude of the carrier signal to represent binary 1 or 0
- Other signal characteristics remain constant as the amplitude changes
- During each bit duration, the peak amplitude remains constant
- Transmission medium noise is usually additive (i.e.; affects the amplitude), therefore, ASK is very susceptible to noise interference → بتأثر بال Noise .
- Binary ASK (BASK) or On/Off Keying (OOK) modulation technique:
  - One of the binary bits is represented by no voltage — يعني لو كان في شكل +5V و -5V لو مثل صفر بطلع في الـ Signal عند الـ 0
  - 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
- 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 — الـ Parameter المتعلقة بالـ Modulation والـ Sampling. تعوض مكان الـ C اللي كان
- The min. bandwidth required to transmit an ASK is equal to the baud rate — بالـ S
- The baud rate is the same as the bit rate → N = S (يعني هون حطينا ناخذ الـ Cases بعين الاعتبار)

* 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*



1 Data / 1 Signal — الـ r = 1 لأنه بيبعث على

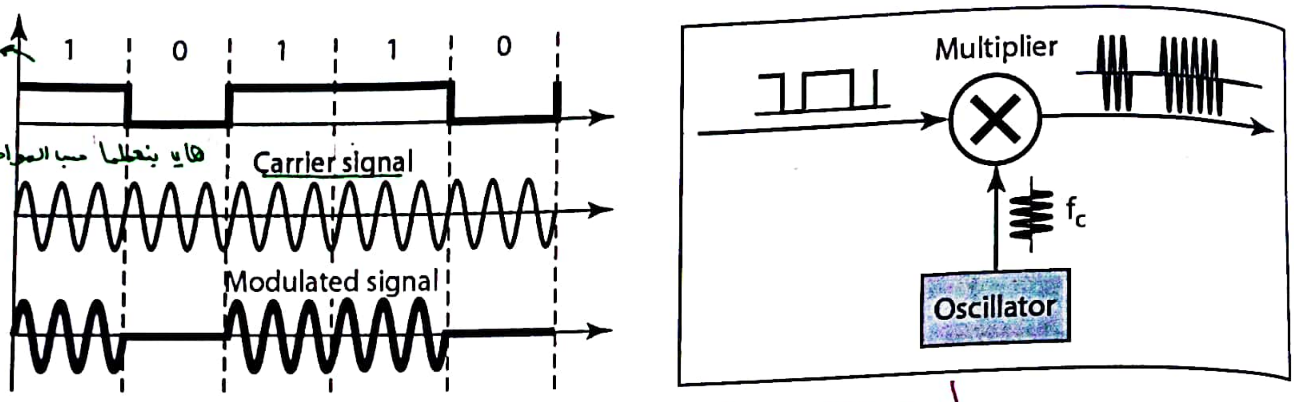
* ASK-T1 : بيوضح كيف بتم ضرب الـ 2 Signal من خلال الـ Resistor.

* Bandwidth → Symmetric about carrier frequency ↳ around

Figure 5.4 *Implementation of binary ASK*

Information Signal الإشارة العادية

هاي بنحللها حسب المواصفات الي بدي اياها.



عمليًا لو بدي أبنيها بال Circuit عشان أضرب ال 2 signals

* ال bandwidth ما بتغير مع Modulation او بدونا.

Example 5.3

لـ نبعث ال Data باتجاه واحد.

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

Solution

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

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

Bandwidth

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

لـ نفسه 50 baud لأنه S=N.

Example 5.4

لازم أخصص band للإرسال و band للاستقبال.

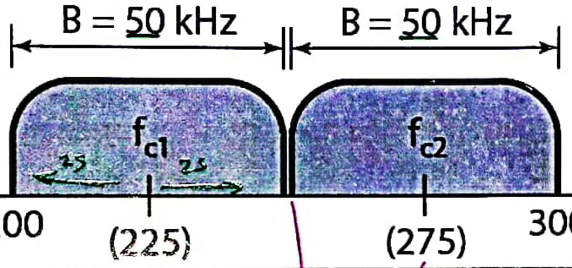
أرسل واستقبل مع بعض

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* *

* بدي ابعت ال 2channels زي الأذاعات كل وحدة على اثر لحال وكل وحدة الها Bandwidth, Freq مختلف.

Nonperiodic* Cont.



*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 ما يصير بينهم تداخل

guard Band ← بتاخذ من الbandwidth.

Frequency Shift Keying (FSK)

اغير ال Freq

حاط ال Carrier

- 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 (مناعة) 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

زي كأنه 2ASK مع بعض بعملوا FSK

* 00 → f_{c1} , 01 → f_{c2}

(ممكن ابعث 2 bits بس مبدأيا خلينا 1 bits.)

- f_1 and f_2 are $2\triangle f$ apart
- The BFSK required bandwidth is $B_{BFSK} = (1+d)\ S + 2\triangle f$
- What is the minimum value of $2\triangle f$?
- The baud rate is the same as the bit rate ➔ N = S

* Minimum Bandwidth = 2S

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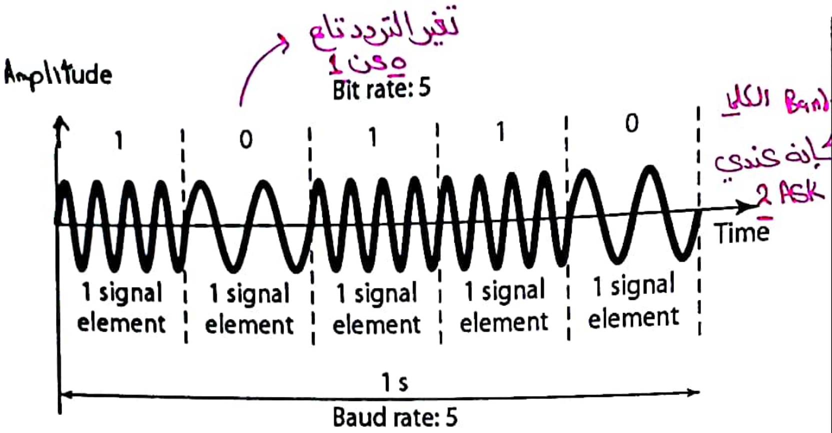
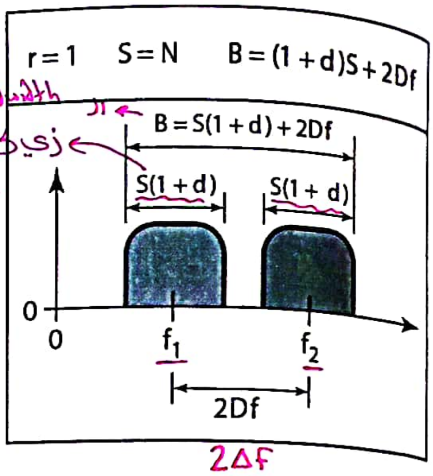
Voltage Controlled osc

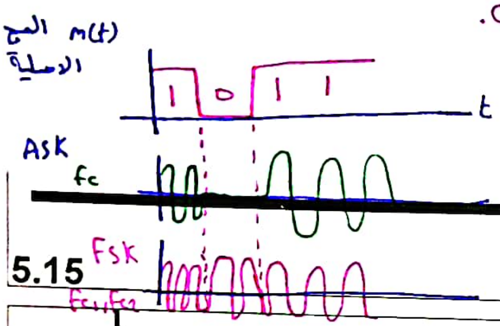
re 5.6 *Binary frequency shift keying*





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

* هاد محتاج لل Noise والBandwidth اكبر لانه بلعب بال Freq.



5.15

Example 5.5

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

« يكون دايمًا بالوسط »

Solution

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

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

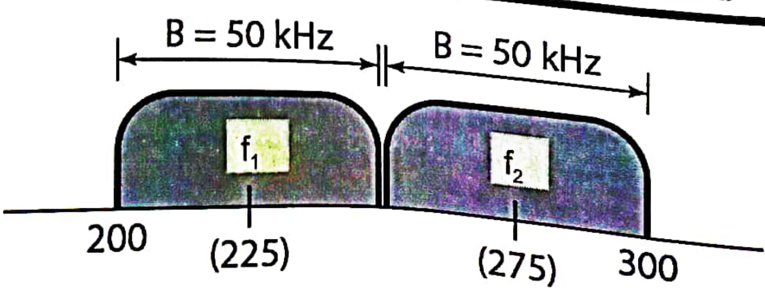
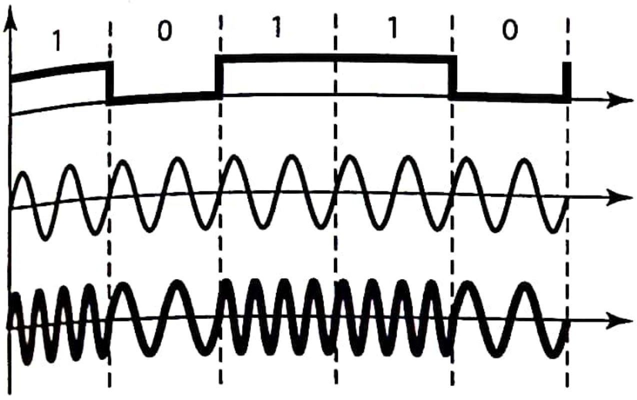
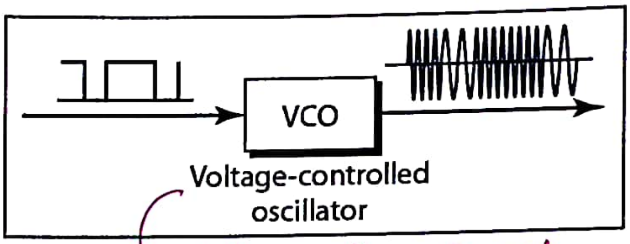


Figure 5.7 *Implementation of BFSK*





↳ Voltage of one Signal
يغير ال Frequency of another Signal

اكتر من 2bits بدي ابعت كيف اتعامل معهم.

Multi-Level FSK (MFSK)

→ بزيد ال bit rate ← ابعت أكثر من Pattern بأكثر من 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 ال bandwidth عندي بزيد لإنه دوره
  - * For example: four different frequencies can be used to send 2 bits at a time
    ال Freq. بزيد
- However, each adjacent pair of frequencies must be $2\Delta f$ apart ← المسافة بين 2 freq
- The MFSK required bandwidth is → عدد ال levels

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

→ لو كان 2 فكأنه رجعنا لل FSK

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

↲ بيعتمد برضه على ال # of levels.

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 $\rightarrow = L$ *(different frequencies), the baud rate, and the minimum bandwidth.* $\rightarrow L \times S$

$8 = 2^3$ عندي

فيدي 8 من ال freq.

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

$S = 3 \text{ Mbps}/3 = 1 \text{ MHz}$

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

The bandwidth is $B = 8 \times 1\text{MHz} = 8 \text{ MHz}$

Figure 5.8 shows the allocation of frequencies and bandwidth.

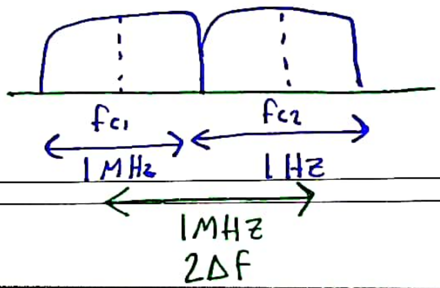
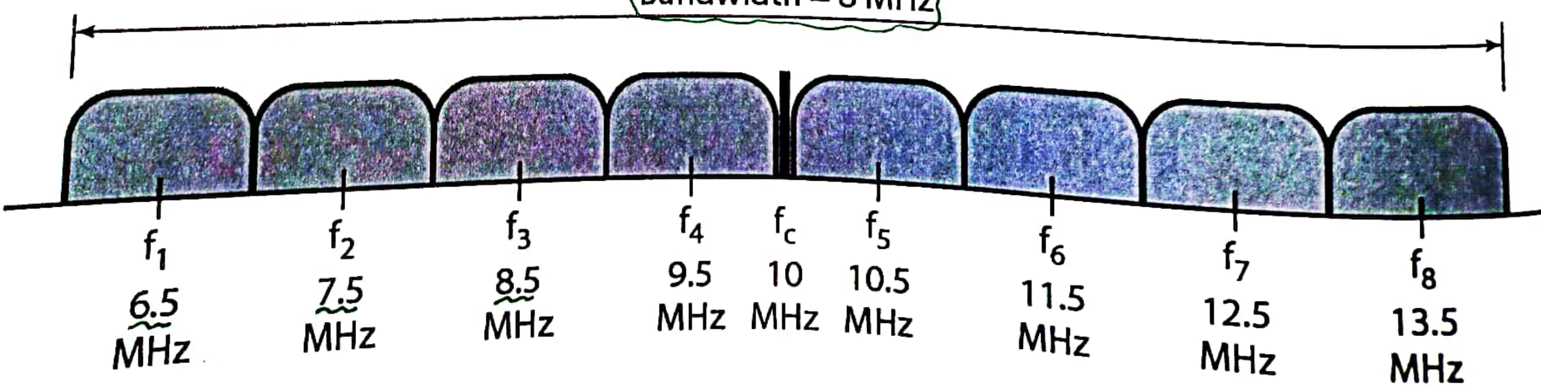


Figure 5.8 *Bandwidth of MFSK used in Example 5.6*

لو كان عندي guard band

الجواب ط اكبر



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

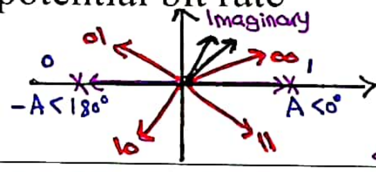
Phase Shift Keying (PSK)

بس بغير الـ Phase

- 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

  مرات بكون صعب جدة الـ reciever إنك تميز بين الـ Phases.
- PSK spectrum and bandwidth requirements are similar to ASK (نفس معادلات 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

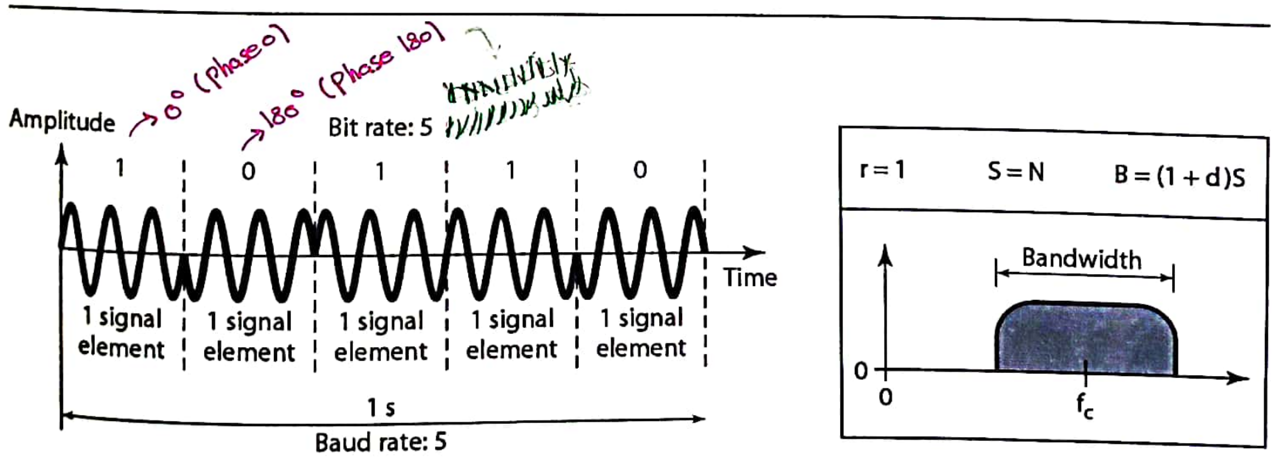
  الرسم الكوكبي
- PSK is limited by the ability of the receiver to distinguish small phase difference, therefore, limiting the potential bit rate



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 يبعت عدد أكبر من الـ bits.

* الـ PSK أفضل من الـ FSK لإنها بتحتاج Frequency واحد والـ Bandwidth محدد وأقل وممانع للـ Noise. (كأنه أخذ ميزات الـ ASK والـ FSK بس)

Figure 5.9 *Binary phase shift keying*



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

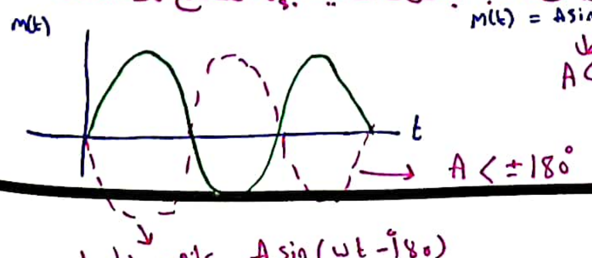
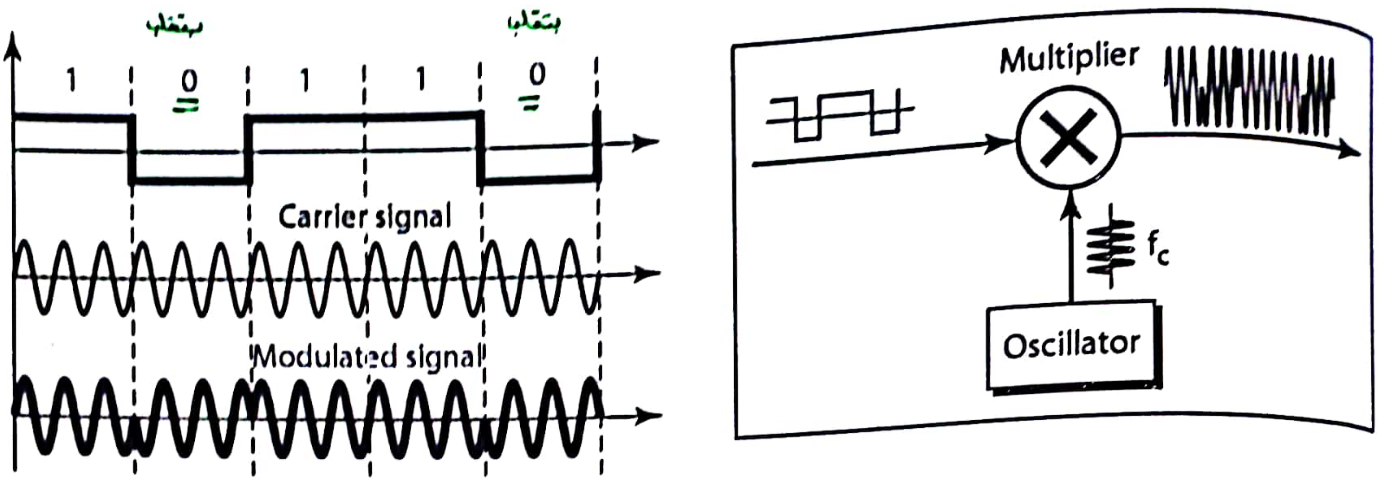


Figure 5.10 *Implementation of BPSK*



5.23

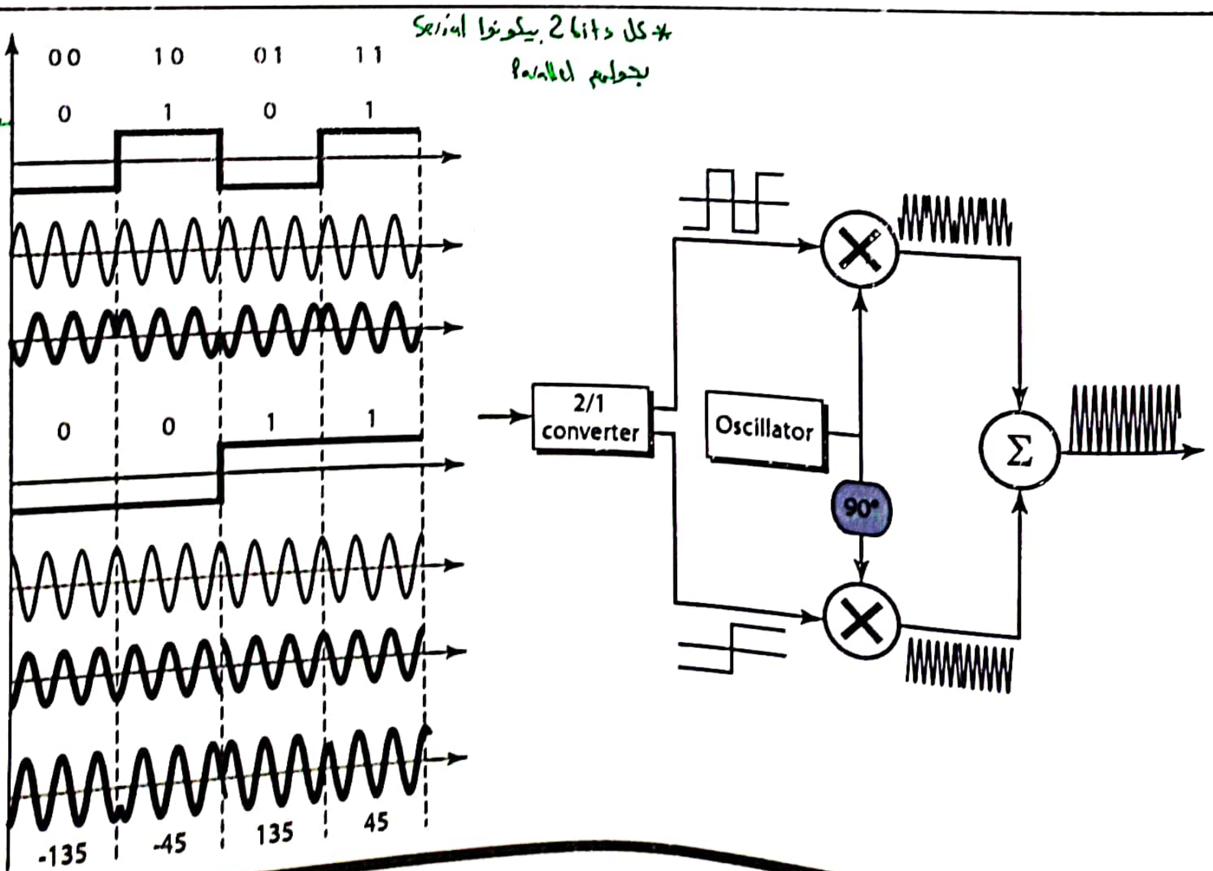
Figure 5.11 *QPSK and its implementation* $\boxed{9/5}$ عالدفتر

* كل 2 bits بيكونوا Serial بحولهم Parallel

T اول 1 كانت

2T بالـParallel

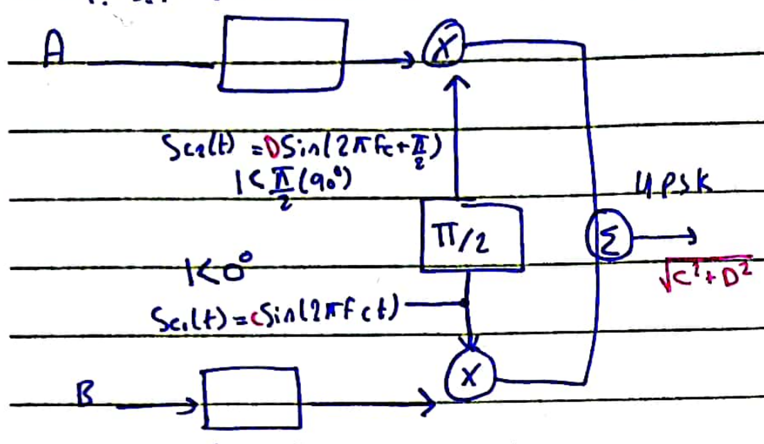
90 درجة من الـCarrier الاول



9/5

ثاد لحاله عال Y axis

level shifter (بيكوجحم الـ bit)



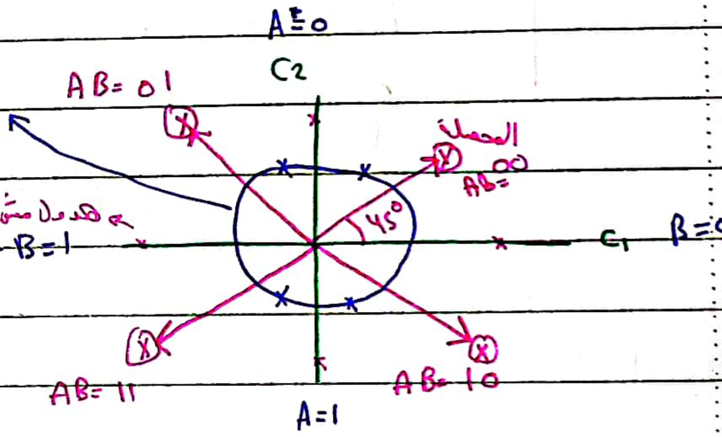
Serial to parallel

Data	10	01	01	11	00
A	1	0	0	1	0
B	0	1	1	1	0

ثاد لحاله عال X axis
بس بدنا المحصلة.

اذا تغيرت الـ Amplitude
كمان بيكون عندي 8 نقاط
بدل 4 . فبدل التنين مع بعض

هدول مش موجودين لانه في ٤ (محصلة)
B=1 ← زي ما بدي بجمعهم



A	B	Σ
0	0	$\frac{\pi}{4} \rightarrow 45°$
0	1	$\frac{3\pi}{4} \rightarrow 135°$
1	0	$\frac{5\pi}{4} \rightarrow 225° \rightarrow -135°$
1	1	$\frac{7\pi}{4} \rightarrow 315° \rightarrow -45°$

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Example 5.7

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

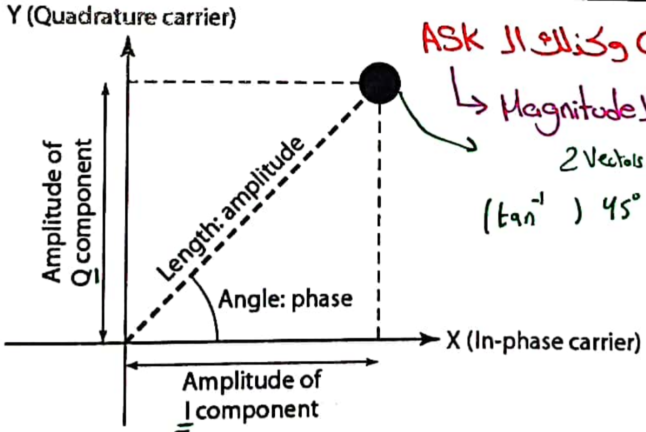
ASK معادلات

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.

$\boxed{S}$ $= S(1+d)$

Figure 5.12 *Concept of a constellation diagram*



* بنقدر نرسم ال QPSK وكذلك ال ASK

بنغير فقط ال Magnitude

المجموع تاع 2 Vectors

الزاوية 45° ($\tan^{-1}$)

- 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.

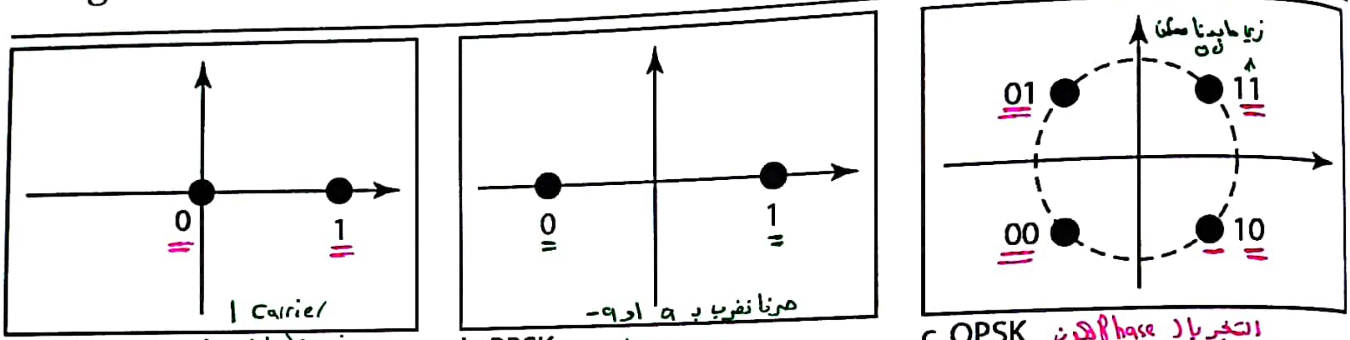


Figure 5.13 *Three constellation diagrams*

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

التغير بال Phase فقط
وهيك زدت r (عدد ال bits على ال Signal Element)

مزيج بين التنين بس الاحسن نغير ال Phase لانه ال Amplitude في noise وصعب اعمل ال Receive.

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

نفس معادلات ال ASK و PSK

Note

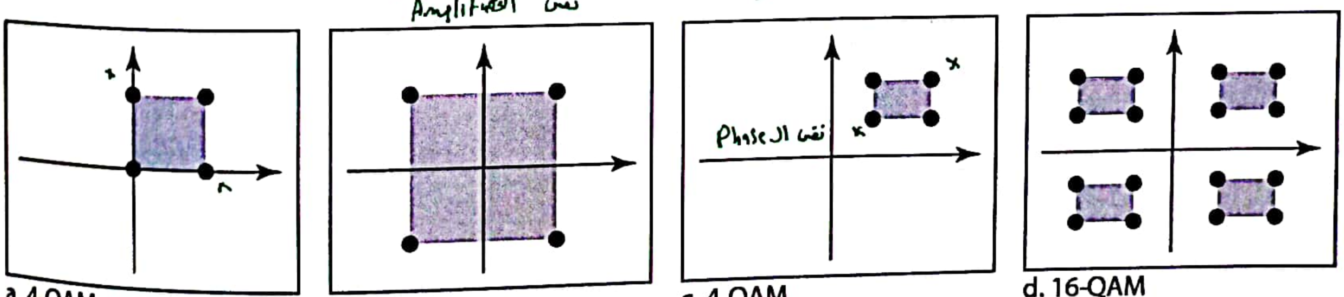
*Quadrature amplitude modulation is a combination of ASK and PSK.

5.29

Figure 5.14 *Constellation diagrams for some QAMs*

4 diffirent Phase
نفس ال Amplitude

4 different Amplitude
3 // phase



a. 4-QAM → 2 bits

$\log_2 4 = 2$

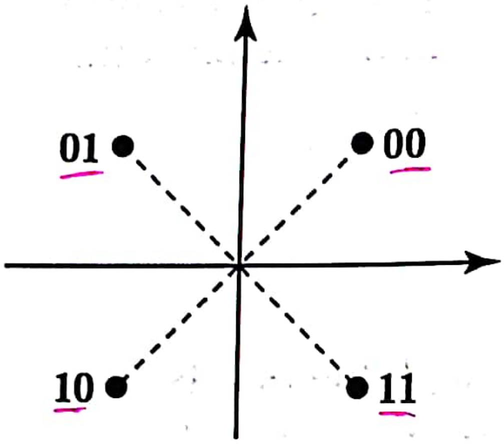
b. 4-QAM

c. 4-QAM

d. 16-QAM

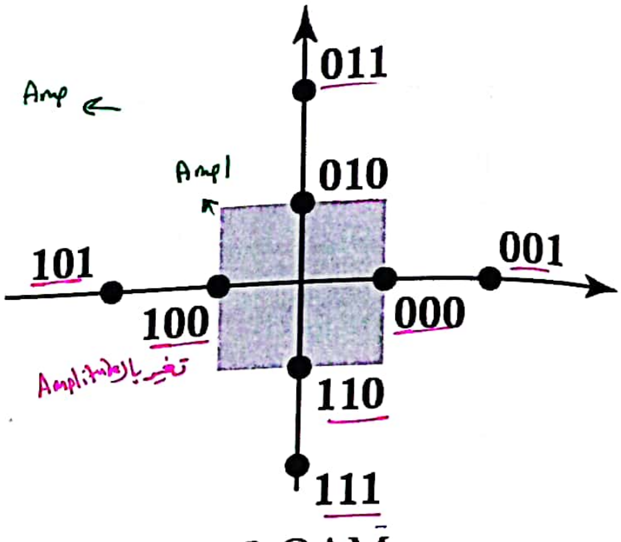
4 bits

Examples of 4-QAM and 8-QAM constellations



4-QAM
1 amplitude, 4 phases

2 amp 2phase هاي احنا من



8-QAM
2 amplitudes, 4 phases

Magnitudes

لو أخذنا الـ X-axis لحالة:

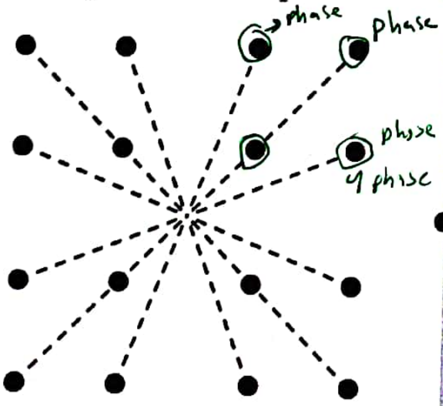
→ phase ثابت / 2 Amplitudes

8-QAM → $2^3 = 8$ → 3 bits

5.31

Examples of 16-QAM constellations

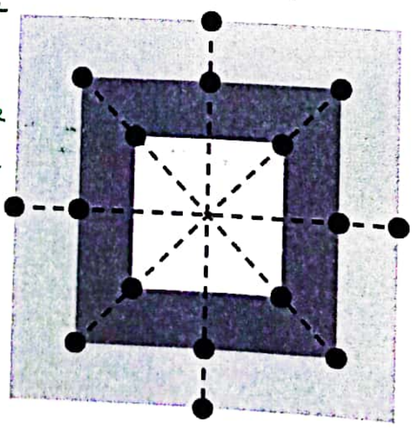
* 3 amplitudes, 12 phases



16-QAM

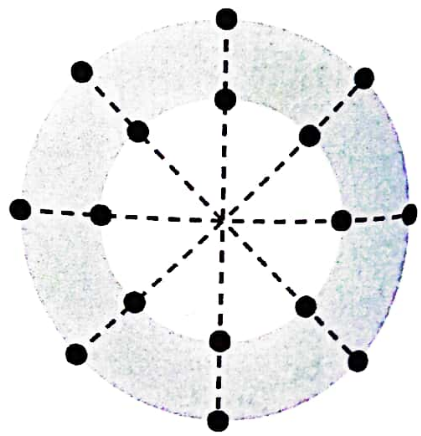
4x3 ~~4~~ = 12 phase

* 4 amplitudes, 8 phases



16-QAM

* 2 amplitudes, 8 phases



16-QAM

5.32

Example

* كل الفكرة ب ؟

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
(S)

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

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

5.33

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.
(N) (S)

له مناسب لإنه 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,

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

كل ما ازيد r بزيد Bit rate

حطيناه N كرمز بس

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

بيرفع ال Data Rate على نفس ال bandwidth عن طريق ال r

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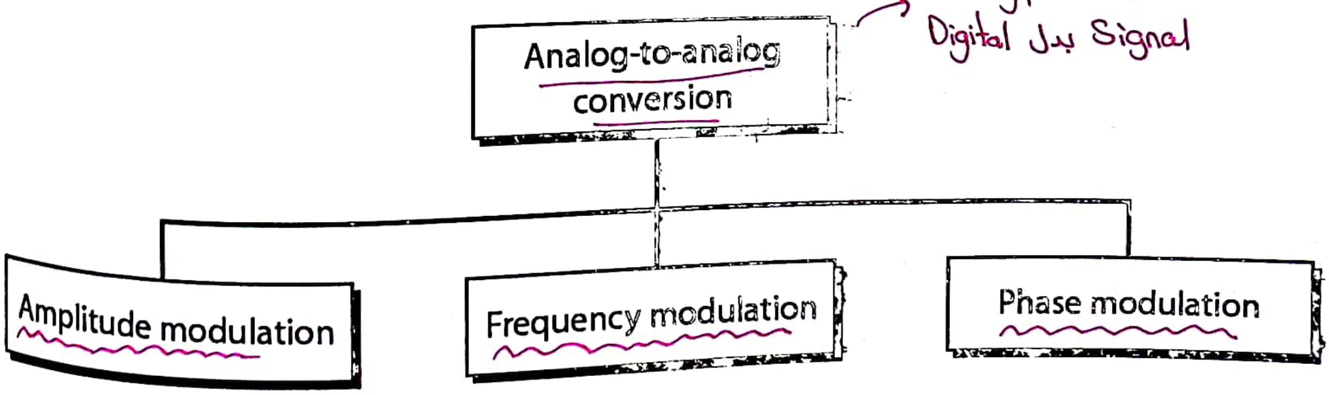
5-2 ANALOG-TO-A

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*



باستخدام Analog
Digital بدل Signal

Amplitude Modulation (AM)

Video Signal أو Voice signal بالعادة بتكون

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

ثابتين

- 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 عندك 10 kHz) محددة من قبل جهات معينة ← Standard)

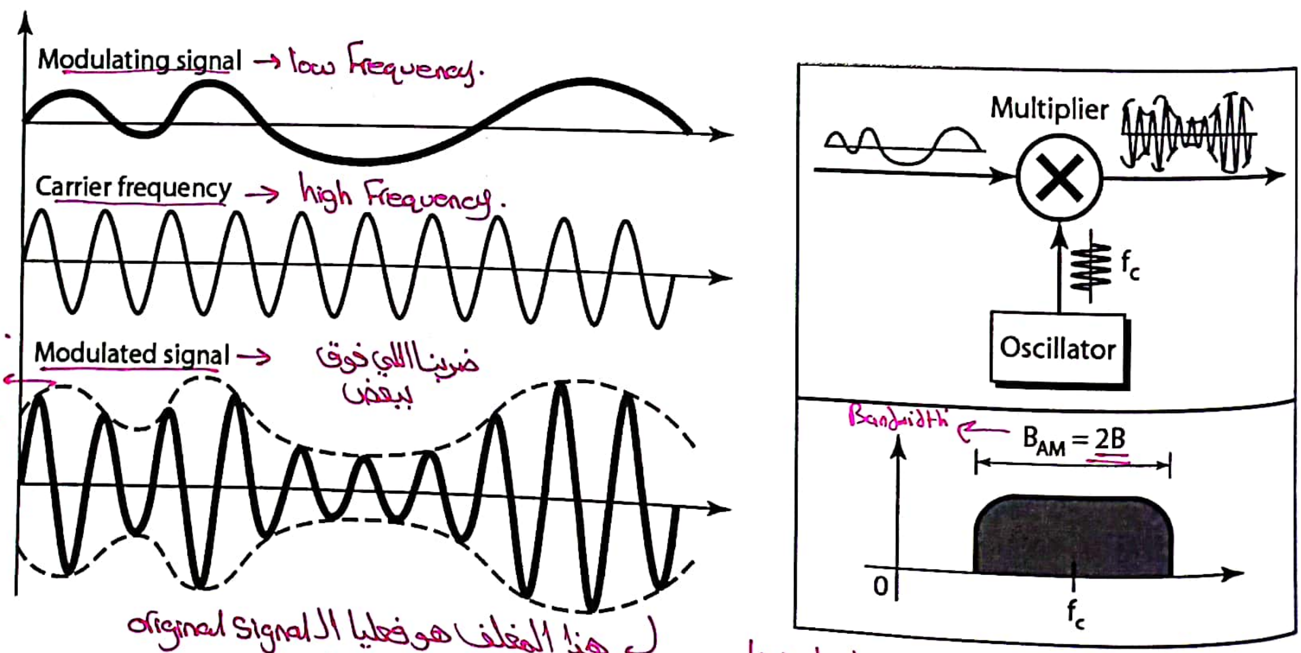
- Every other band is used as a guard band to prevent interference among adjacent channels

لـ مساحة كشان ما يتداخلوا مع بعض 2 bandwidth.

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

Figure 5.16 *Amplitude modulation*

الخط المتقطع مش موجود

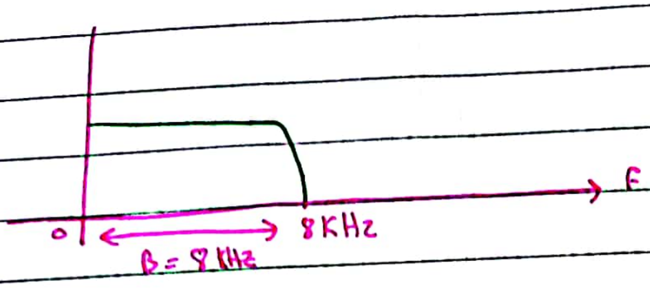


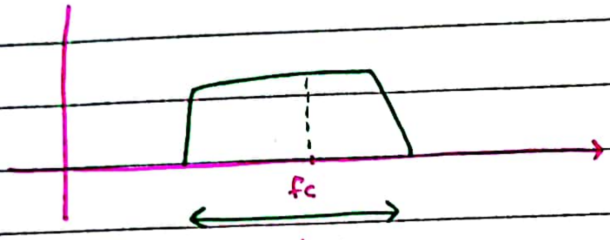
لـ هذا المغلف هو فعليا ال original signal اللي كانت موجودة عندك (envelope)

low-band filter ← ال receive

11/5 عالدفتر

11/5





$B_{AM} = 16\ KHz$

$\boxed{B_{AM} = 2B} = f_c + f_m$

Example (AM):

$m(t) = \cos(W_m t)$ ... Message (Modulating Signal) $\longrightarrow$

رسمناهم بالصفحة التالية

$S(t) = \cos(W_c t)$ ... Carrier Signal (modulated Signal) $\longrightarrow$

حاصل ضربهم ببعض

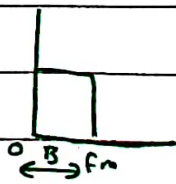
$$S_{AM}(t) = m(t) \cdot S(t)$$
$$= \cos(W_m t) \cdot \cos(W_c t)$$
$$= \frac{1}{2}\left[\cos((W_m - W_c)t) + \cos((W_m + W_c)t)\right]$$

$$\downarrow \qquad\qquad \downarrow$$

$$f_c - f_m \qquad\qquad f_c + f_m$$

$\cancel{B_{AM} = 2B = f_c + f_m}$

$B_{AM} = f_c + f_m - (f_c - f_m) = 2f_m$



$B = f_m - 0$
$= f_m$

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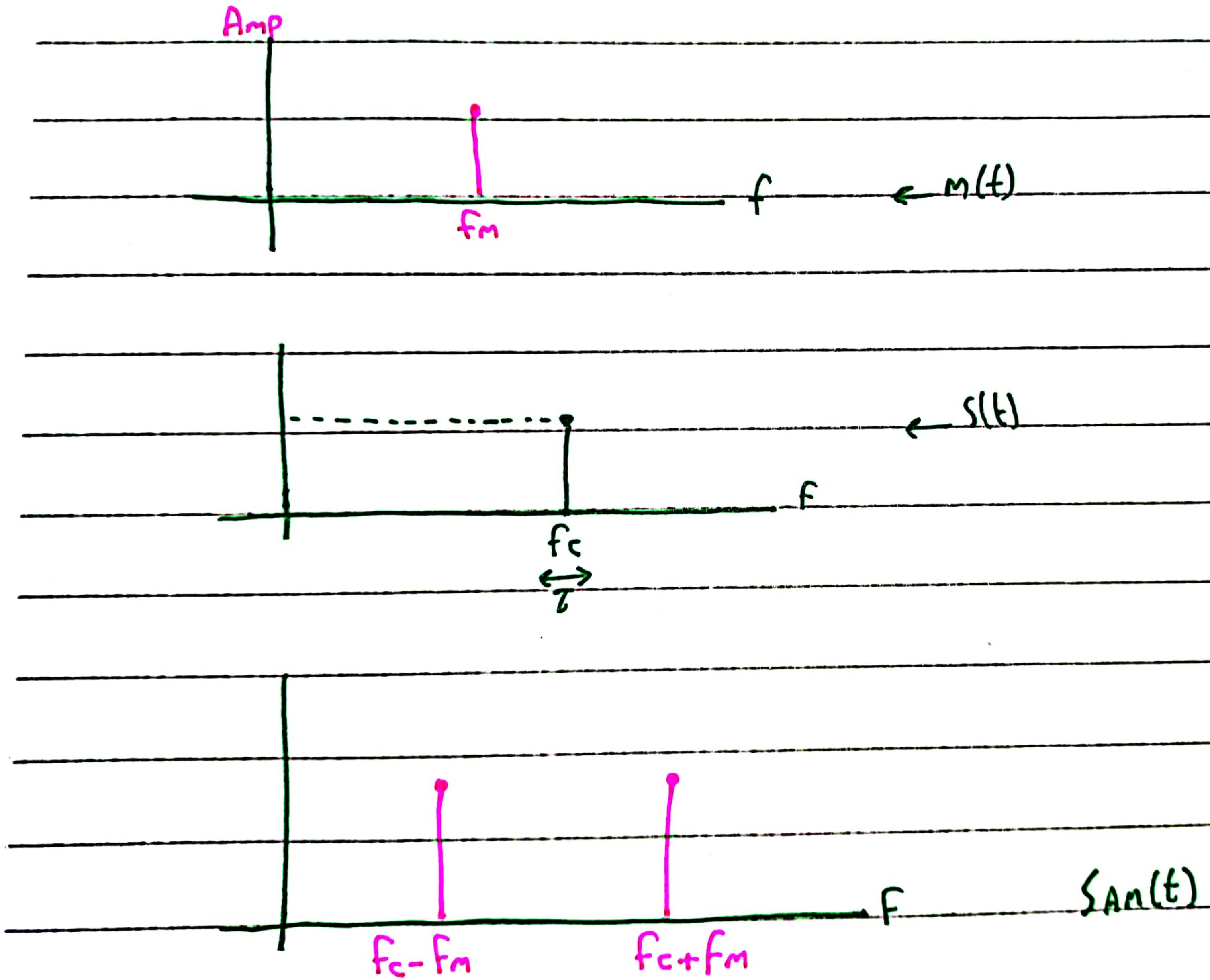


Amp
f
m(t)
fm
s(t)
F
fc
τ
F
SAm(t)
fc-fm
fc+fm

Note

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

« أهم نقطة »

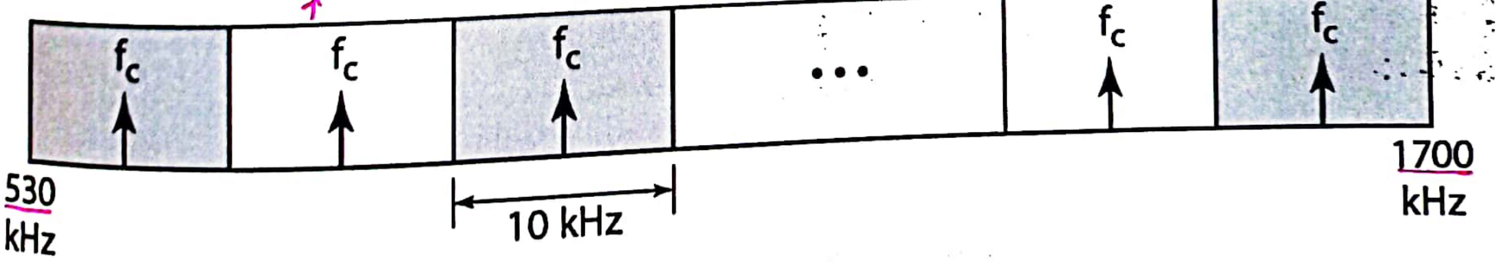
5.41

Figure 5.17 *AM band allocation*

guard band

بالعادة في GB بين كل قناة وقناة

guard



بنغير الـ Frequency بدل الـ 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
- The bandwidth of the FM signal is usually about 10 times the bandwidth of the modulating signal

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

١٠ أمثال

- The bandwidth of a stereo audio signal (voice and music) is about 15 kHz

ضعف الـ ٢ double channel

- FM radio stations are assigned 200 kHz band per channel

كل channel

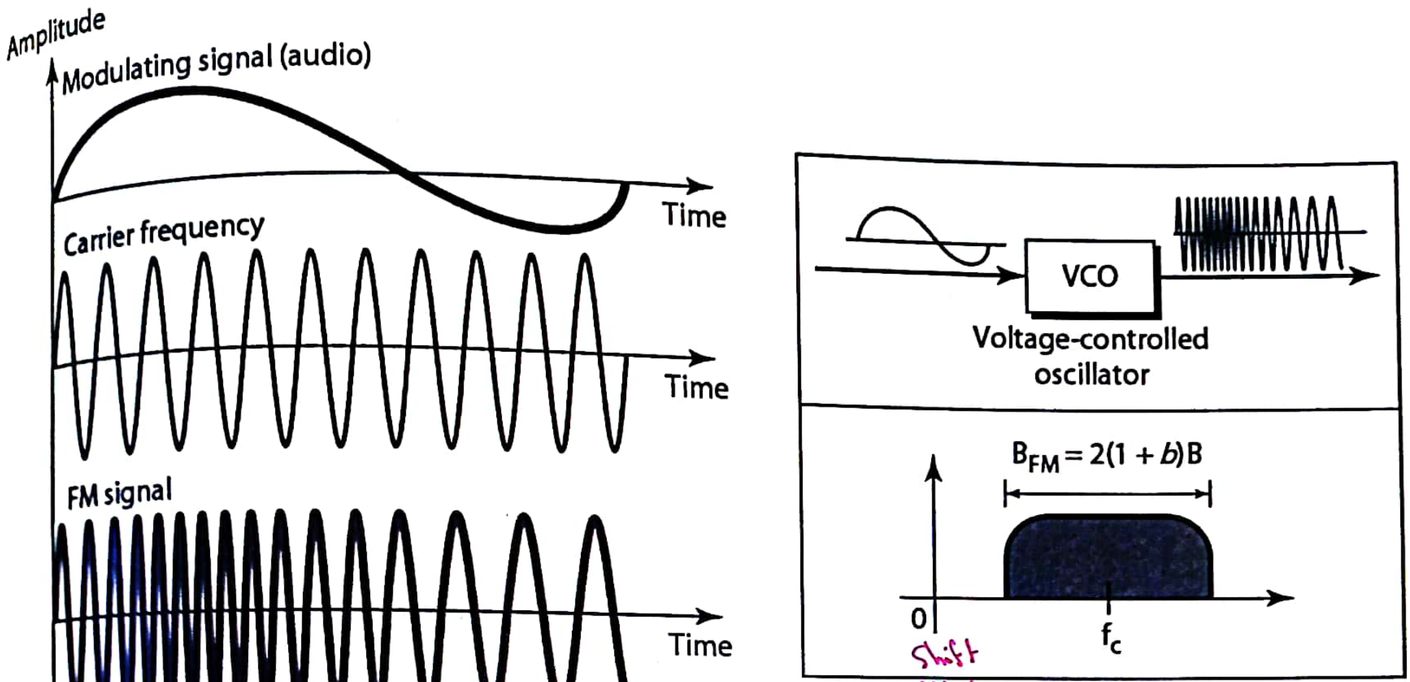
- Every other band is used as a guard band to prevent interference among adjacent channels

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 بالـ Modulation، بالعادة = 4. (FM) معطاه بالسؤال

Figure 5.18 *Frequency modulation*



للbandwidth

الي بيتطلع عندي

و Quality احسن (FM).

بين ال Am ابعد احسن

Figure 5.19 *FM band allocation*

عشان ما يتداخلوا.

* كل ال range الاستخدام

Phase اي بتغير

Phase Modulation (PM)

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

  يعتمد بمقدار 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

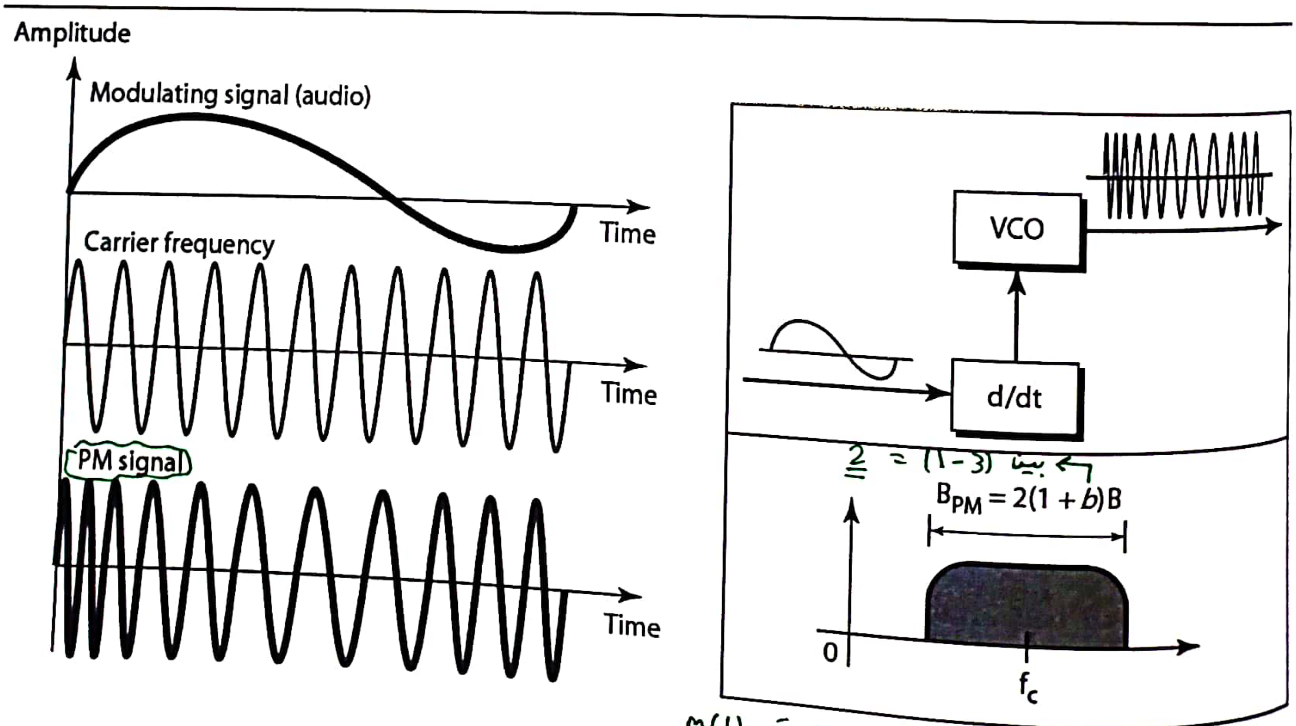
  مشتقة m(t) هي بتغير الـ freq تاع الـ carrier signal

- The bandwidth of the PM signal is usually abut 6 times the bandwidth of the modulating signal

  اقل

  بالوسط بين AM و FM

Figure 5.20 *Phase modulation*



بده نتتبع تغير المشتقة m(t)
كيف بتغير المشتقة للـ Carrier.

Note

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

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

2

عشان هيك 6 times
معطاه بالسؤال.

Chapter 6

Bandwidth Utilization: Multiplexing and Spreading

* bandwidthال link كبير كثير فمش منطق يكون فيها بس user واحد فحيكونوا أكثر من user يعبي ال link وال bandwidth كامل.

عكس بخص

لـ حكون بدي أزيد الـ bandwidth لأغراض مختلفة مثلاً لزيادة ال Security
بدي انو Bandwidth اكثر زي ال Security

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* link ← الها bandwidth كثير كبير.
* Channel ← تحت استخدام ال user (في حالة ال 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

مجموعة من ال users.

Privacy and anti-jamming can be achieved by spreading.

على نفس ال freq
• حد ابعث Noise بـ عالي Power على ال receiver ح يفكر ال Noise هي الاصلية.

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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.*

ما بعطي كل ال Channel ال bandwidth بعطي لا Users التانيين.

Topics discussed in this section:

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

أكثر من Channel واحدة لانه ال Channel خاصة لا تعدل الواحد

- Several recent transmission media such as optical fibers and satellite microwave links have much higher bandwidth than the average bandwidth needed for most applications

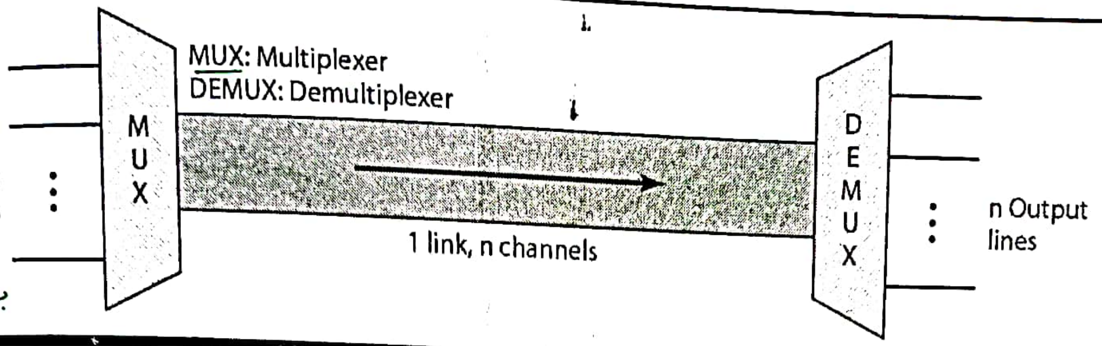
ممكن اطلع اكثر من محطة ع Fiber.

- It is essential to optimize the utilization of a given bandwidth in order to reduce costs and prevent wastage. That is, to improve the cost-effectiveness and utilization of the resources

نستخدمه بشكل كامل. لانه مكلف فنبي استغلها كاملة.

Figure 6.1 *Dividing a link into channels*

كل user
عنده channel
وحدة (أبسط
senario)



users
بدي اطلعهم على link واحد

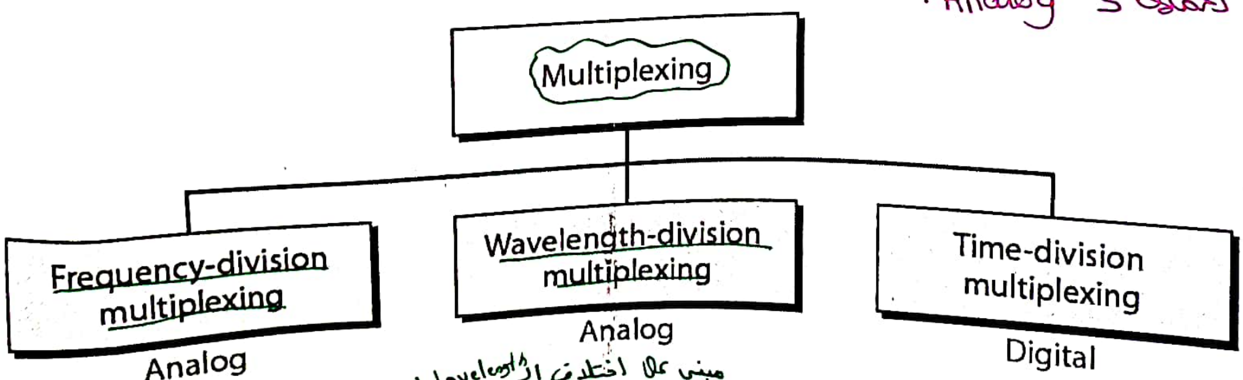
- * The multiplexer (MUX) combines several channels into one stream (many-to-one conversion)
- * The demultiplexer (DEMUX) separates the stream back into the original channels (one-to-many conversion)
- * The link refers to the physical medium
- * The channel refers to the portion of the link that carries the transmission between any pair of lines

حصة الuser من الlink

6.5

Figure 6.2 *Categories of multiplexing*

* حتى الDigital ممكن نحملها ع Analog.



بعطيك جزء منه

مبني على اختلاف الWavelength

كل الSignals
بتكون Digital
بعطيك كله لالك بس
لفترة زمنية محددة.

Frequency Division Multiplexing (FDM)

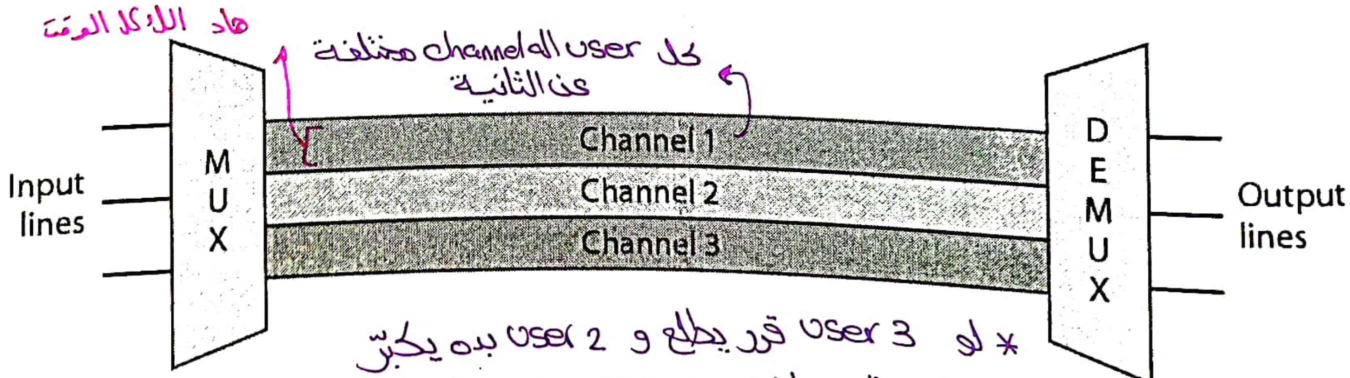
- FDM is an analog technique that can be applied when the bandwidth of the link in Hertz is greater than the combined bandwidths of the individual signals to be transmitted
- Each signal is modulated with a different carrier frequency
- The modulated signals are then combined to form a composite signal, which is transmitted over the link
  كلهم بنجمعوا
- The carrier frequencies should be separated enough using guard bands to prevent overlapping or interfering with neighboring signals

مسافة صغيرة بين الـ Channels الهدف منها ماتتداخل 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*



* لو user 3 قرر يطلع و user 2 بده يكبّر حصته بقدر ياخد حصة user 3 بس أكيد بده يدفع ثمن هاي الزيادة أكيد.

Note

ميزتها مش معقدة

FDM is an analog multiplexing technique that combines analog signals.

* في مشكلة حنكتشفها بالسايمنت الثاني ؟ كيف نحل مشكلة انه اذا اجانا Phase 180 زيادة حيصير استقبال ال bits كله غلط فكيف نحله ؟.

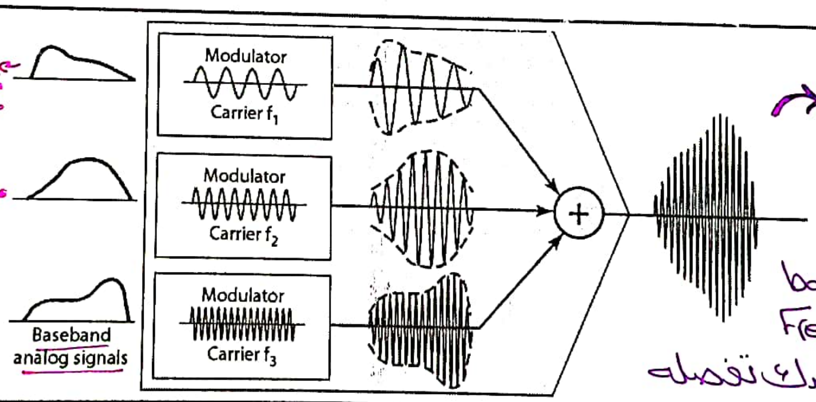
Figure 6.4 *FDM process*

Time domain

ممكن كل وحدة محطة

مثلا محطة قرآن الكريم

محطة -



لو بدي أرجع أفصلهم عن بعض بستخدم low pass Filter أو band-pass Filter centered على Frequency تبع ال 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.

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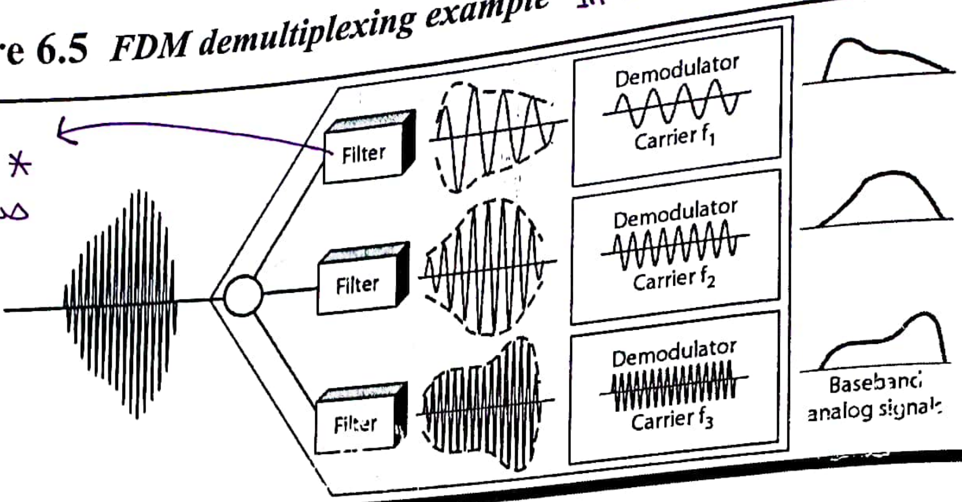
Figure 6.5 *FDM demultiplexing example* In Time Domain

Design Process الـ *
صعبة بس مجرد ماتشف
خلص بحيس 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 كلهم جنب بعضا.

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

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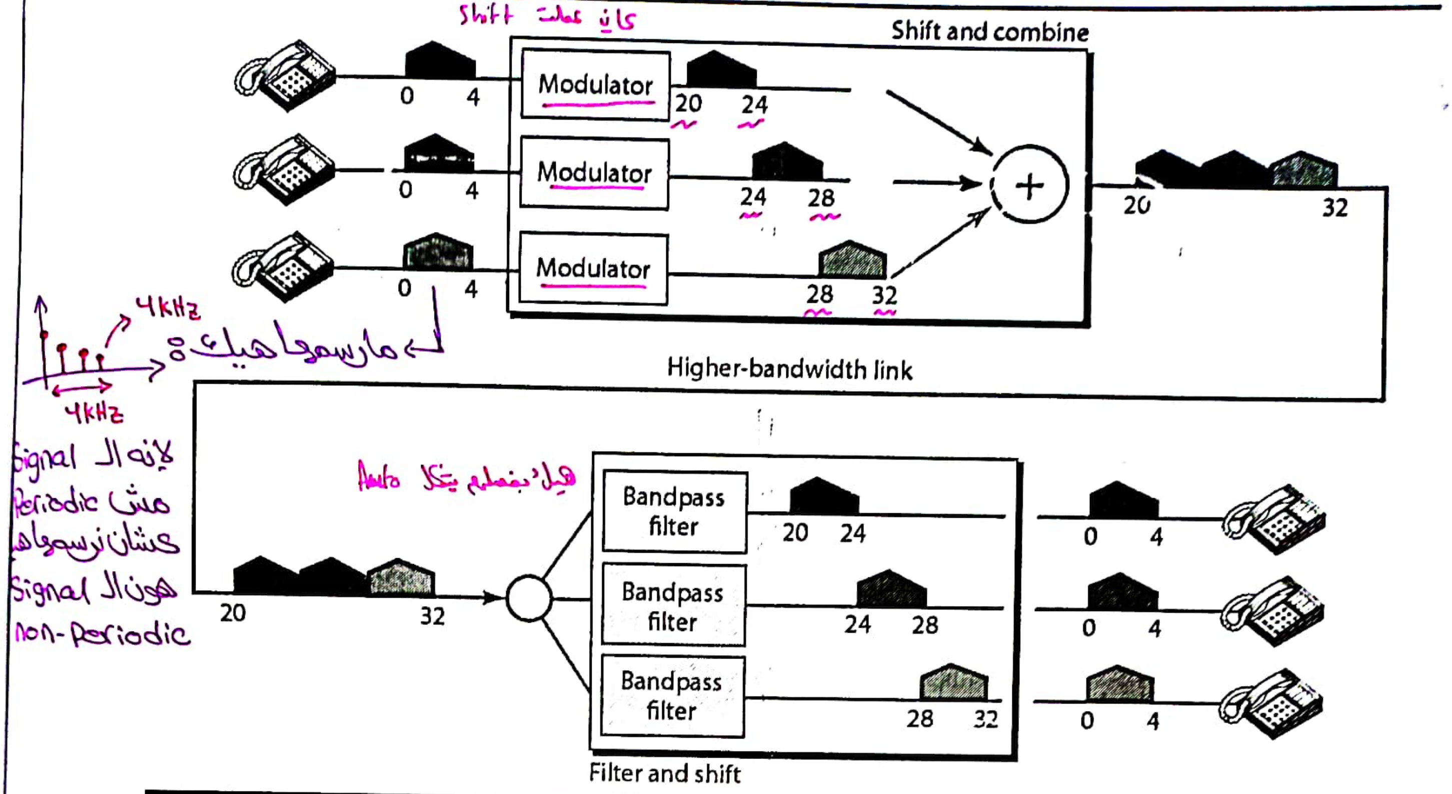
Figure 6.6 *Example 6.1*



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

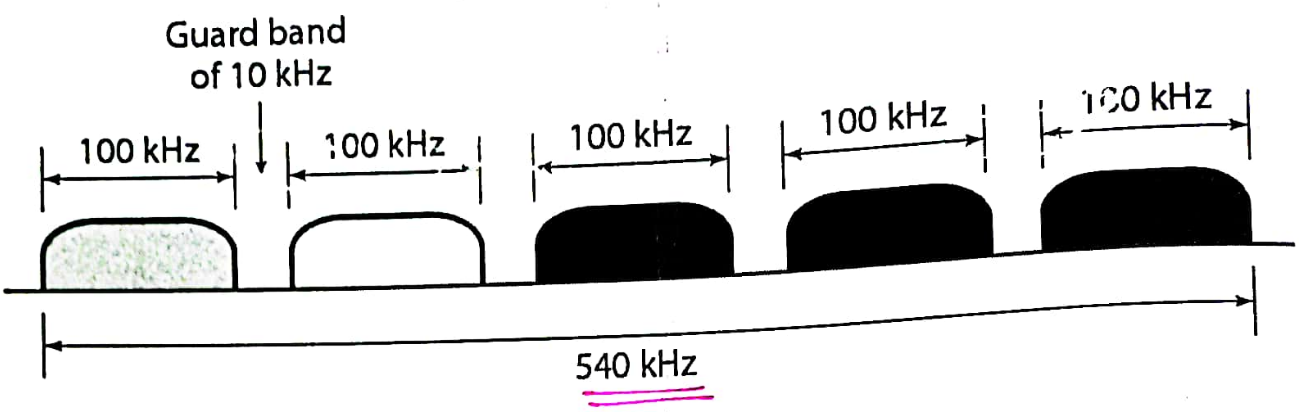
For five channels, we need at least four guard bands. This means that the required bandwidth is at least

$$5 \times 100 + 4 \times 10 = 540 \text{ kHz},$$

as shown in Figure 6.7.

5-1

Figure 6.7 *Example 6.2*



16/45 عالدفتر

6.15

Example 6.3

Data Rate ←
لكل يوزر لحاله

Four data channels (digital), each transmitting at 1 Mbps, use a (Main) satellite channel of 1 MHz. Design an appropriate configuration, using FDM.

↳ Bandwidth of the channel (total)

Solution

The satellite channel is analog.

بناخد ال take bandwidth

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

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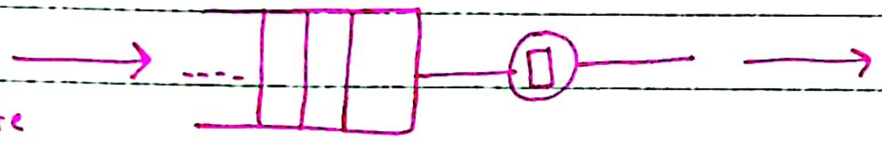
$\boxed{16/5}$

λ input rate pkts/s (avg)



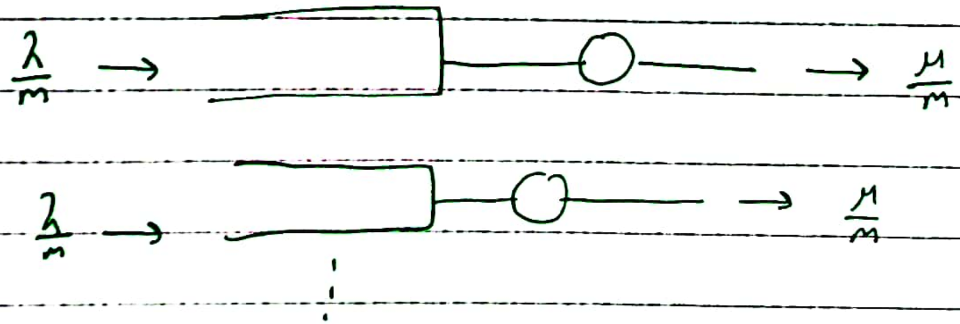
μ output rate pkts/s (avg.)

Assume "m" users

$\frac{\lambda}{m}$ rate for one user:

$\frac{\mu}{m}$ Service (output) rate for one user:

} For FDM



} m users

CamScanner

RV

$E[\cdot]$

avg / expected value / mean

$$E[T_{FDM}] = \frac{1}{\frac{\mu}{m} - \frac{\lambda}{m}} = \frac{m}{\mu - \lambda}$$

avg. expected Value

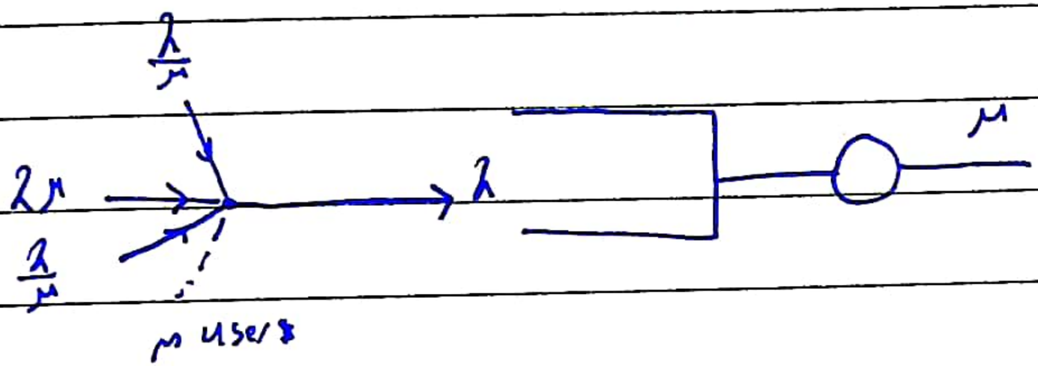
* ال FDM افضل من ناحية

ال Multimedia لانه بيهمه ال Variability

$E[(X-\bar{X})^2]$

$Var[X] = \sqrt{\sigma}$

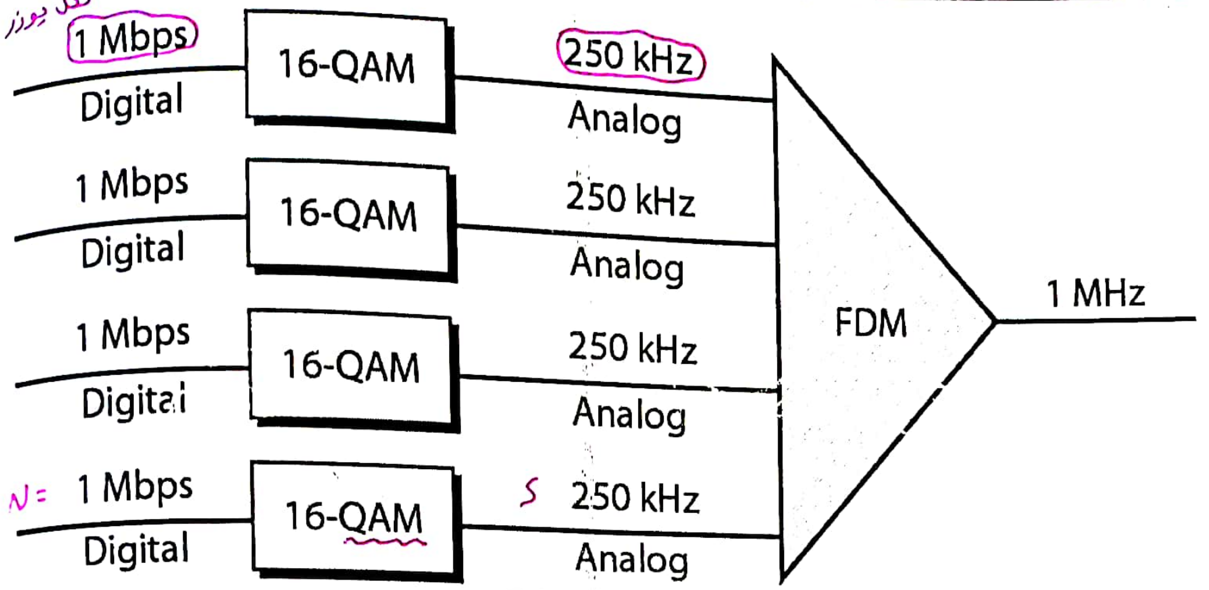
Not FDM ~~[illegible]~~ (SM)



$$\boxed{E[T] = \frac{1}{\mu - \lambda}}$$

Figure 6.8 *Example 6.3*

لكل يوزر

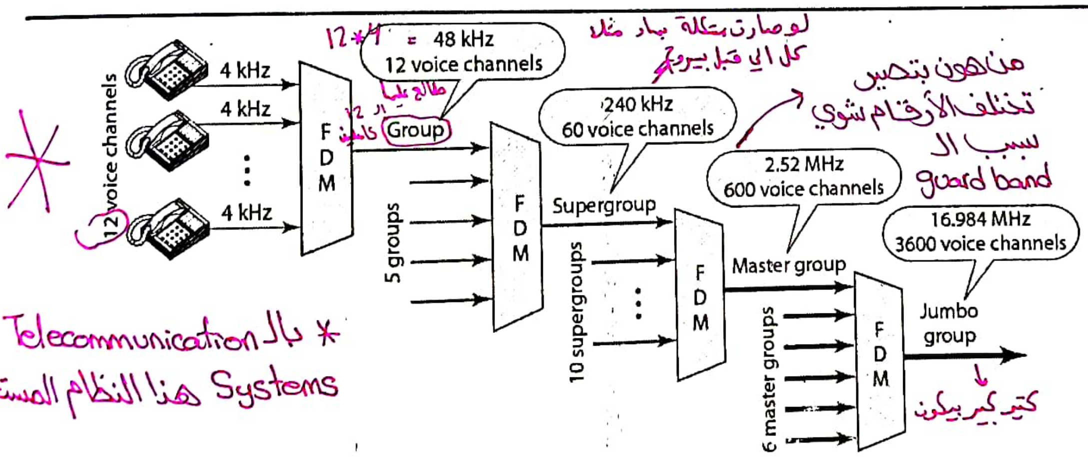


N = S

$S = \frac{N}{r} = \boxed{r=4}$

* النظام المستخدم في الأردن هو ال Standard الأوروبي.

Figure 6.9 *Analog hierarchy* → حاجتك للـ bandwidth بتزيد



12*4

لوصارت مشكلة بهاد مثلا كل الي قبل بيروح

منا هون بتصير تختلف الأرقام شوي بسبب ال guard band

*ما بقدر العب فيه وا غير كثير.

* سهل ا فير البورد

* بال Telecommunication Systems هذا النظام المستخدم

كثير كثير بيكون

- * Used by telcos to optimize the efficiency of the infrastructure
- * Combines multiple low bandwidth lines into higher bandwidth
- * Fewer but larger lines are used: → بدي أقل بس الـ 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.

← بالعادة يستخدم مع الـ Analog بس ممكن يستخدم مع الـ Digital لأنه هون نقدر نحولها لـ 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

*range of AM: 530 → 1700 KHz

* شوفي مثال الكتاب عالريموت (مثال الـ 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?

الجيل الأول

ارسال

استقبال

عشان اعرف كم يوزر عندي.

يوزرز

↳ " كله بده يكون فل "

الـ Users هون random and Independent.

Solution

متى ترفع يكونوا قديمين

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

من السؤال

790 مشترك بنقدر نخدم.

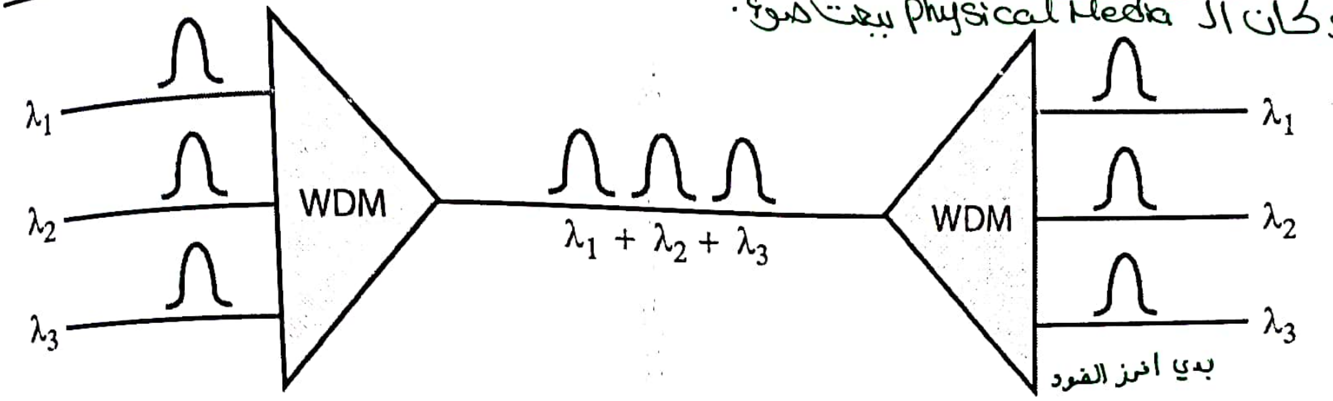
Independent and random ←

التردد عالي جداً م

بعتمد خائص 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

منستخدم فيوم المنشور (Prism) هو الي بتعامل معه بالضوء.

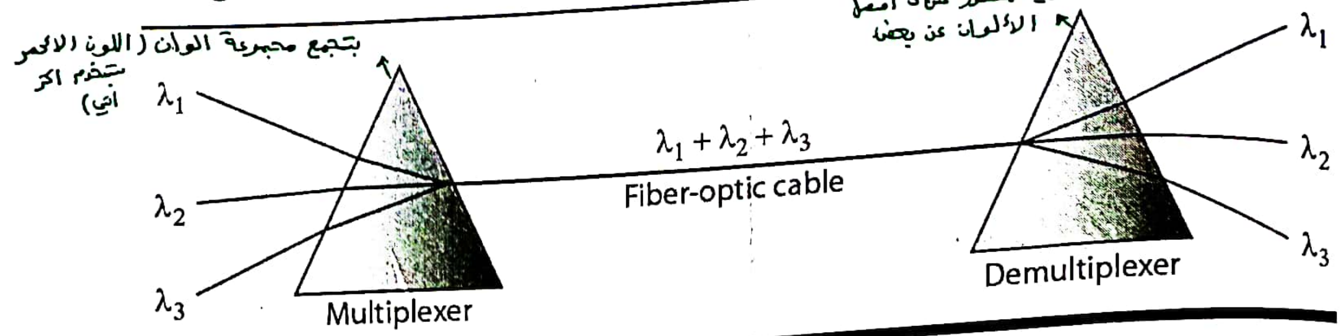
* في losses بالضوء بتتأثر بالانحناء إلا بال Cable.

6.21

Note

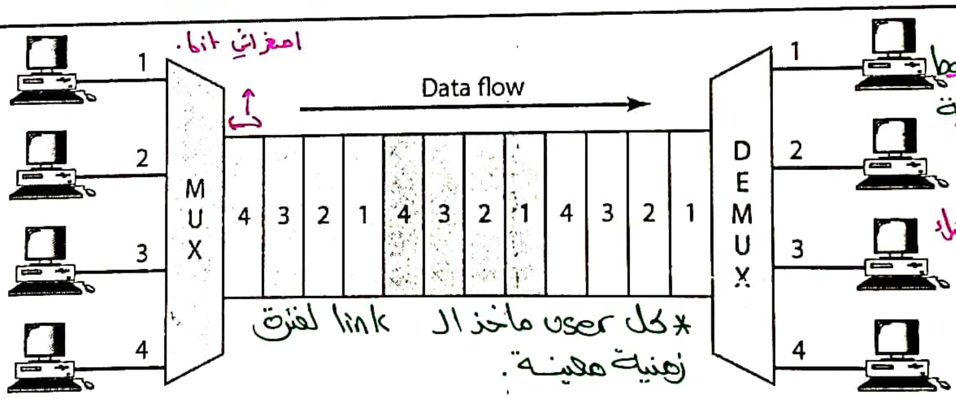
WDM is an analog multiplexing technique to combine optical signals.

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 *

عالدور وبتفضل دمتها دايما ثابتة لو بدي اعطيك اكثر بعطيك اكثر من Slot

- * TDM is a 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

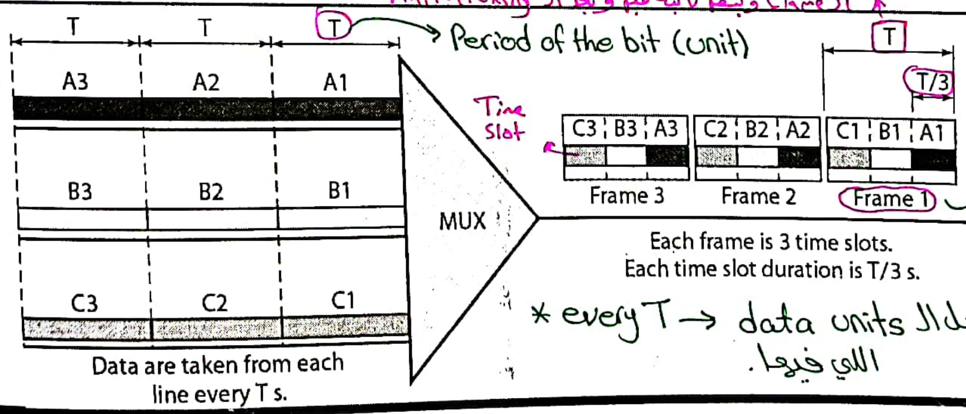
TDM is a digital multiplexing technique for combining several low-rate channels into one high-rate one.

Fiber optic بالغالب

Time sharing على الـ based

6.25

Figure 6.13 ***Synchronous time-division multiplexing***



* لو وحدة من الأجزاء تاعت ال Fram ماكان الها data بضل الجزء من ال Time محجوز الها (الدفش)

ال Period ثابتة ال Frame وبتظل ثابتة قبل وبعد ال multiplexing.

- ▪ 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

حصة اليوزر بال Frame.

* Frame duration تساوي بالضبط ال data unit duration on Input link.

* أحسن ال Performance ل TDM لما يكون عندك light load مش heavy load.

* العلاقة بين ال time وال Rate علاقة عكسية *

6.26

لازم الـ input
→ والـ output
يكونوا متناسقين

لأنه الـ Time slot بكل موجود
لا user حتى لو مافي
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

$\boxed{18/5}$ عالدفتر

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.

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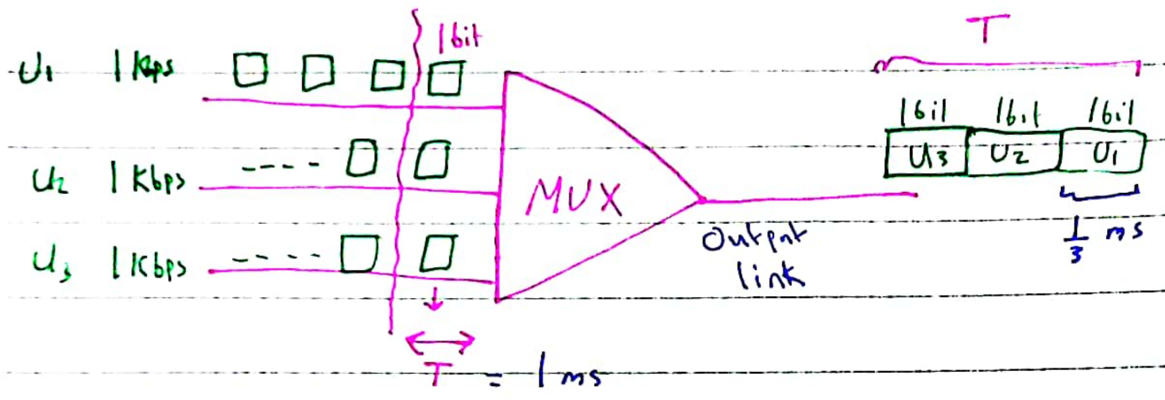
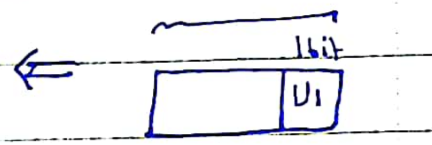
18/5

Example 6.5:





U_1 data rate =

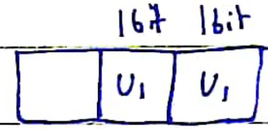
$= \frac{1\ bit}{1\ ms} = 1\ Kbps$

1

* $T = ??$

$= \frac{1}{10^3} = 1\ ms$

لو عندي U_1 ← 2 slots.



U_1 rate $= \frac{2\ bit}{1\ ms}$

$= 2\ Kbps$

T is the frame duration

* Frame rate = ??

Five Maple

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$$= \frac{1 \text{ frame}}{1 \text{ ms}}$$

$$= 1000 \text{ frame/s}$$

☒ Output link data rate = ??

a) $\frac{3 \text{ bits}}{1 \text{ ms}} = 3000 \text{ bps}$

or

b) Output rate = frame rate * frame size

$$= 1000 * 3 = 3000 \text{ bps}$$

CamScanner

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.

$\boxed{*}$ $\frac{2 \text{ bits}}{4 \mu s} = 0.5 \text{ Mbps} = \boxed{U_1 \text{ rate}}$

مثال عن Example 6.6

$T = 4 \mu s$

U_3	U_1	U_2	U_1

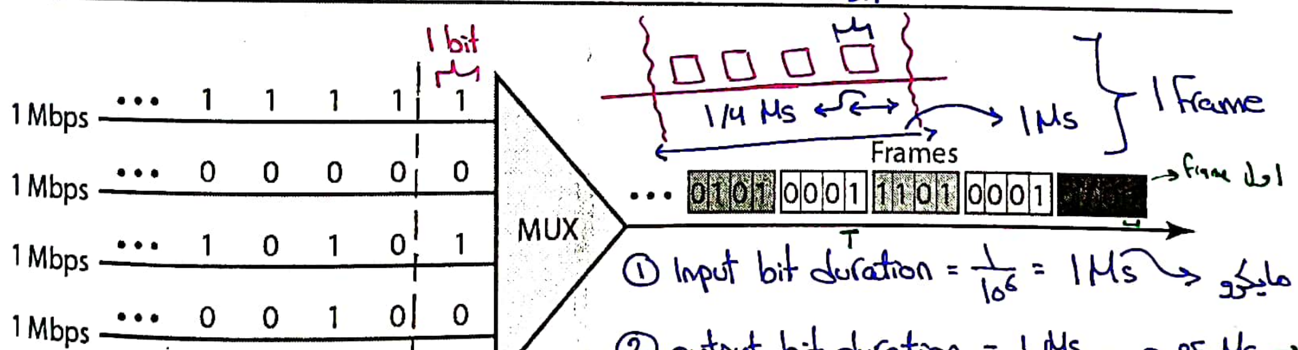
6.29

Slot لوكان وحدة $= \frac{1 \text{ bits}}{4 \mu s} = 0.25 \text{ Mbps}$

* ما دخل U_1 يكونوا جوا ال frame جنب بعض عادي.

Figure 6.14 *Example 6.6*

$U_1 \text{ rate} = \frac{1 \text{bit}}{1 \mu s} = 1 \text{Mbps}$



① Input bit duration $= \frac{1}{10^6} = 1 \mu s$ → مايكرو

② output bit duration $= \frac{1 \mu s}{4} = 0.25 \mu s$ → حصة الواحد

③ Frame rate $= \frac{1 \text{ frame}}{1 \mu s} = 1000{,}000$ frame/s

④ output data rate = frame rate x frame size
$= 10^6 \times 4 = 4 \text{ Mbps}$
لـ ميجا

or

6.30

$\frac{4 \text{ bits}}{1 \mu s} = 4 \text{ Mbps}$

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.

نفس اي قبل.

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.

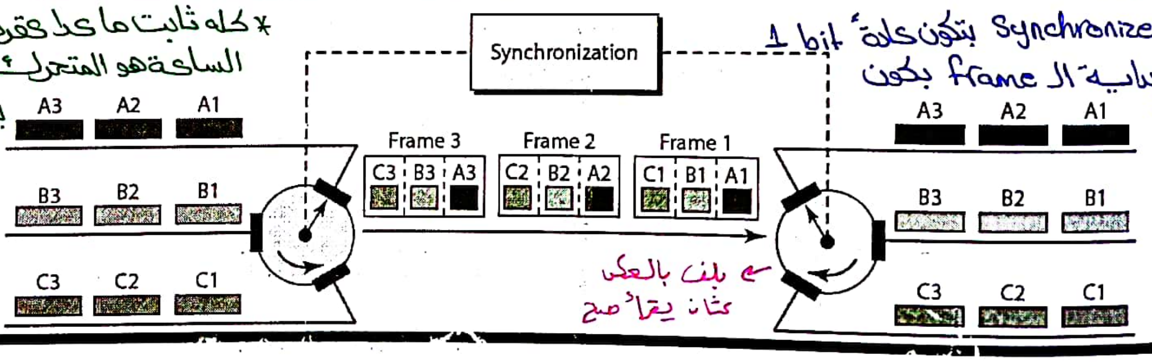
6.31

كيف ابني أميز الـ Frames اللي جاية من الـ users عن بعض؟ ممكن يصير في رخبطة بين اللي انبعت واللي وصل فلتجنب هالاشي بضيفو عادة Synchronize bit.

Figure 6.15 *Interleaving*

* كله ثابت ما عدا عقرب الساعة هو المتحرك فلازم ننسق بين الـ input والـ output لتكون القراءة صحيحة

* الـ Synchronize bit بتكون عادةً 1 bit بنهاية أو بداية الـ frame بكون متفق عليها وقيمتها



← بلف بالعكس عشان يقرأ صح

- * 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 الـ Synchronize بس بنعمل مشكلة الـ overhead.
↓
هو زيادة بعت ما كان لازم نبعت فبالتالي بستهلك من الـ link bandwidth.

6.32

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.

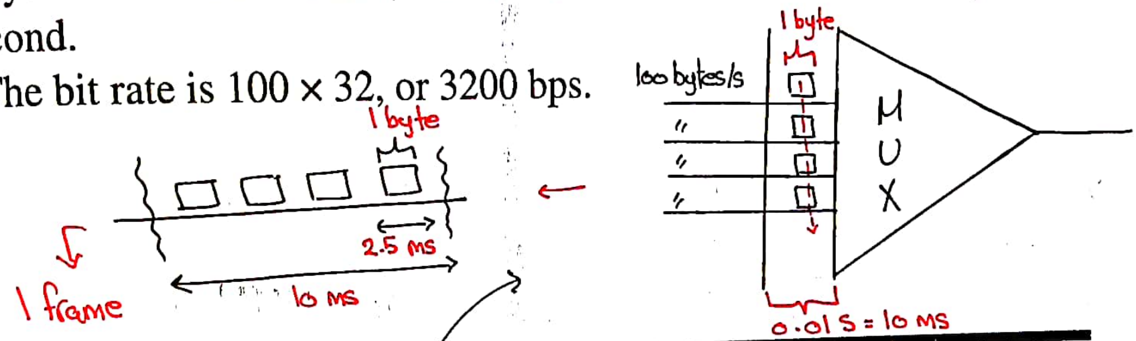
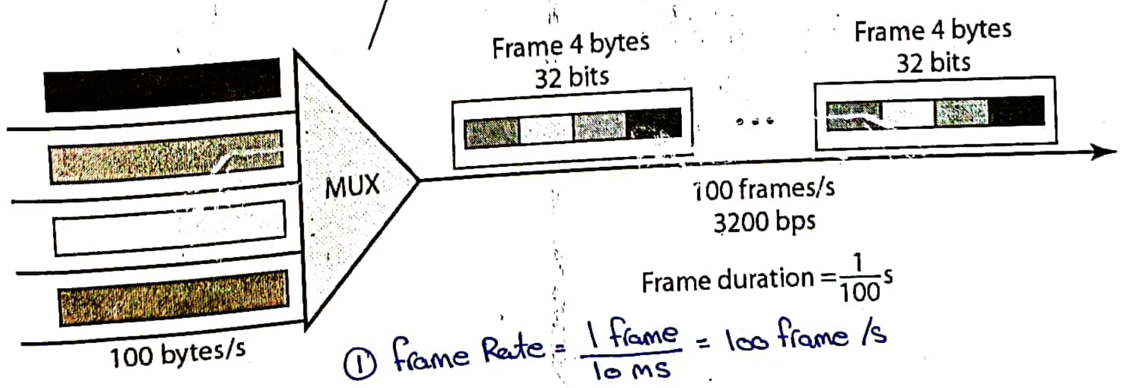


Figure 6.16 *Example 6.8*



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.

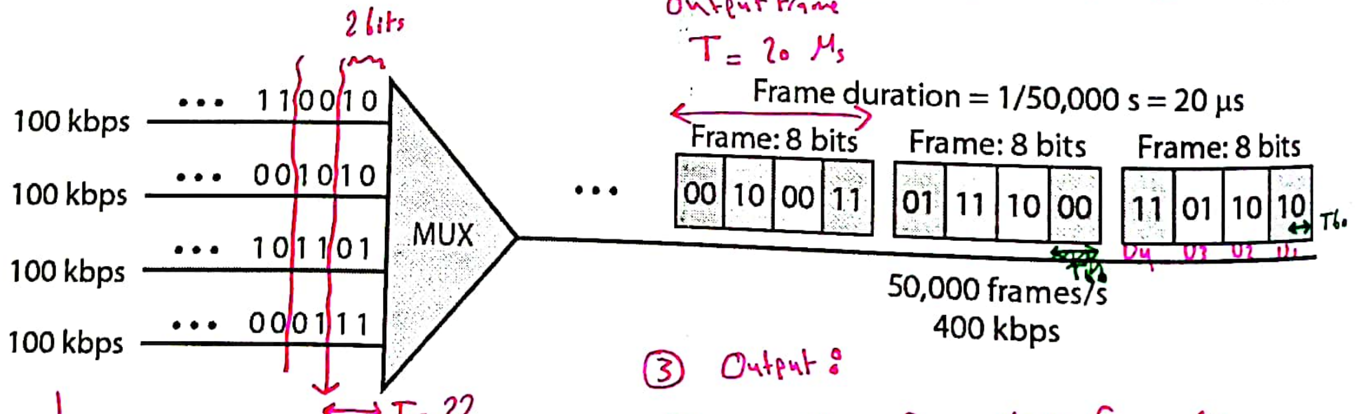
① Input: Bit duration $T_b = \frac{1}{100*10^3} = 0.1\ ms = \boxed{10\ \mu s}$

$T = 2T_b = 2(10\ \mu s)$ $\boxed{T = 20\ \mu s}$ $\boxed{T = nT_b}$

② Frame rate = $\frac{1\ \text{frame}}{20\ \mu s}$ = $\boxed{50,000\ \text{frame/s}}$

frame size = 8 bits

Figure 6.17 *Example 6.9*



لو كانوا مختلفين بعمل اضافة للأقل عشان يصيروا نفس الاشي.

③ Output:

ⓐ rate = frame rate * frame size $= 50*10^3 * 8 = 400\ k\ \text{bit/s}$ bps

or ⓑ rate $= \frac{8\ \text{bits}}{20\ \mu s} = 400\ k\ bps$

④ $T_{bo} = \frac{1}{400*10^3} = 2.5\ \mu s$

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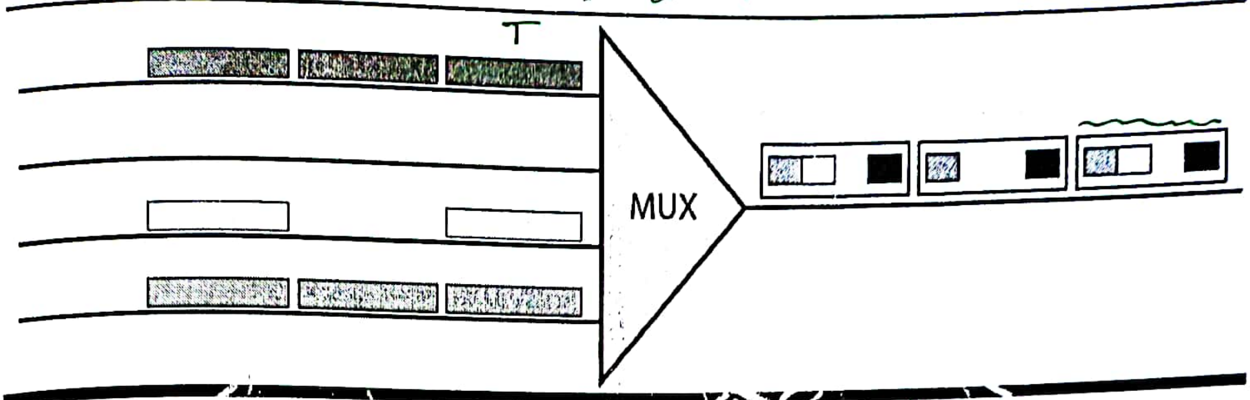
Figure 6.18 ***Empty slots***

* مكان ال frame بيضل فاضي وهاد إشي مش منيح .



- Synchronous TDM may not be very efficient بنستخدم بالاماكن الي فيها Traffic عالي عشان ما يكون فراغ بالداتا.
- 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

ال users ما ضيا rates مختلفين.

Non-Homogeneous TDM

هل ممكن ال rates الي فاينة تكون مختلفة؟

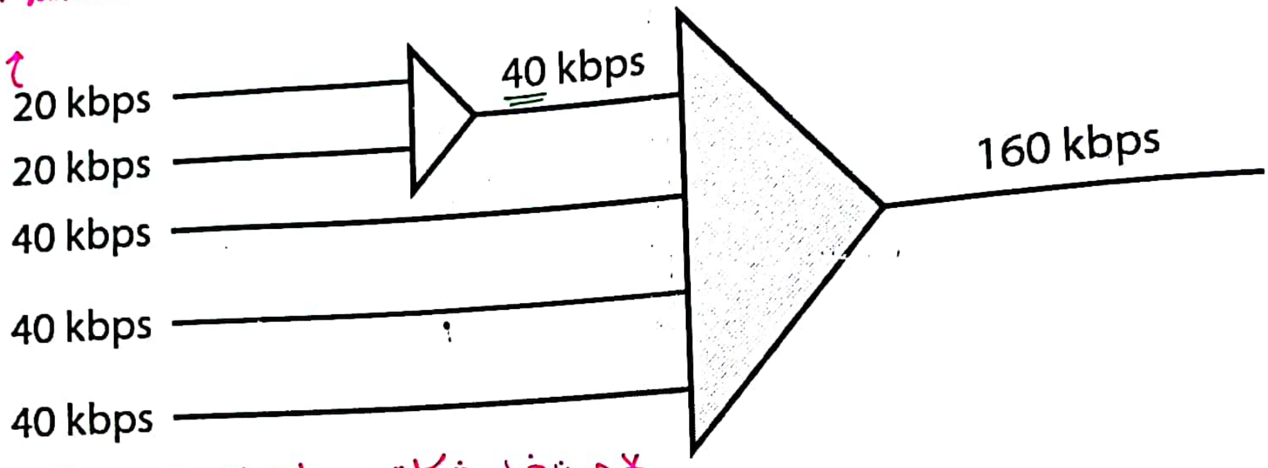
- Can we multiplex devices that have different data rates?
- The answer is yes. How?
  - Faster devices may use more time slots per frame than slower devices بنعطيهم حصة أكبر من ال Frame
  - But remember that in a TDM system:
    - The number of multiplexed devices, the number of time slots per frame, and the time slot duration in the frame are always fixed
  - بحل المسألة بدلالة الثوابت هدول.
  - 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
  - المفروض 9 time slot
  - 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. بشوف عاقل واحد بفرضه X والباقي بكونوا حصص من هذا ال 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

CamScanner

Figure 6.19 *Multilevel multiplexing*

اذا كان مثلا 18 kbps
بخليه 20 بحط 2
. dont care

اليوزر بلزمه
20

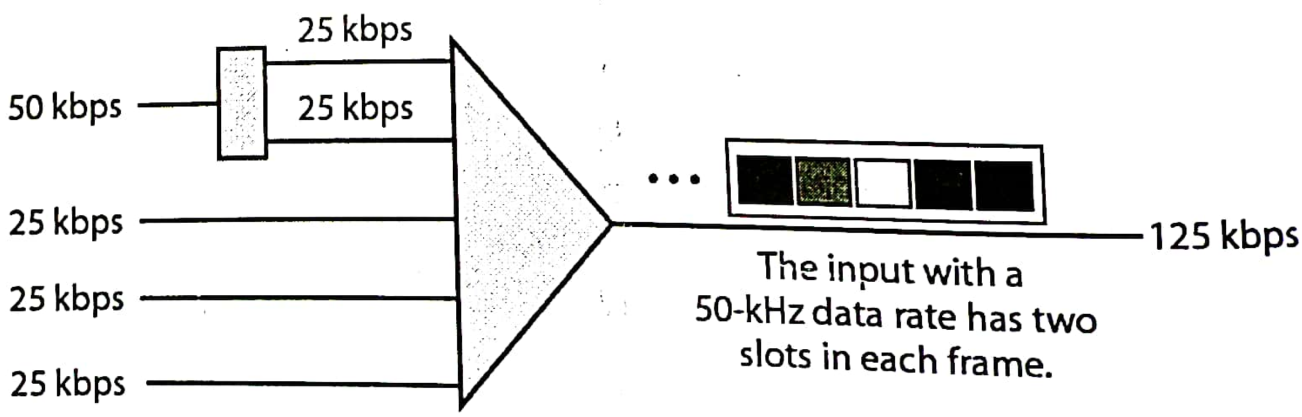


*هون في مشكلة بسيطة بال Propagation Delay
كثير عالية بتكون.

دفتر 21/5

6.39

Figure 6.20 *Multiple-slot multiplexing*



21/5

Example:

Consider five users with the following data rates:

U_1 : X bps   Data unit = 1bit

U_2 : X bps

U_3 : $2X$ bps

U_4 : $2X$ bps

U_5 : $2X$ bps

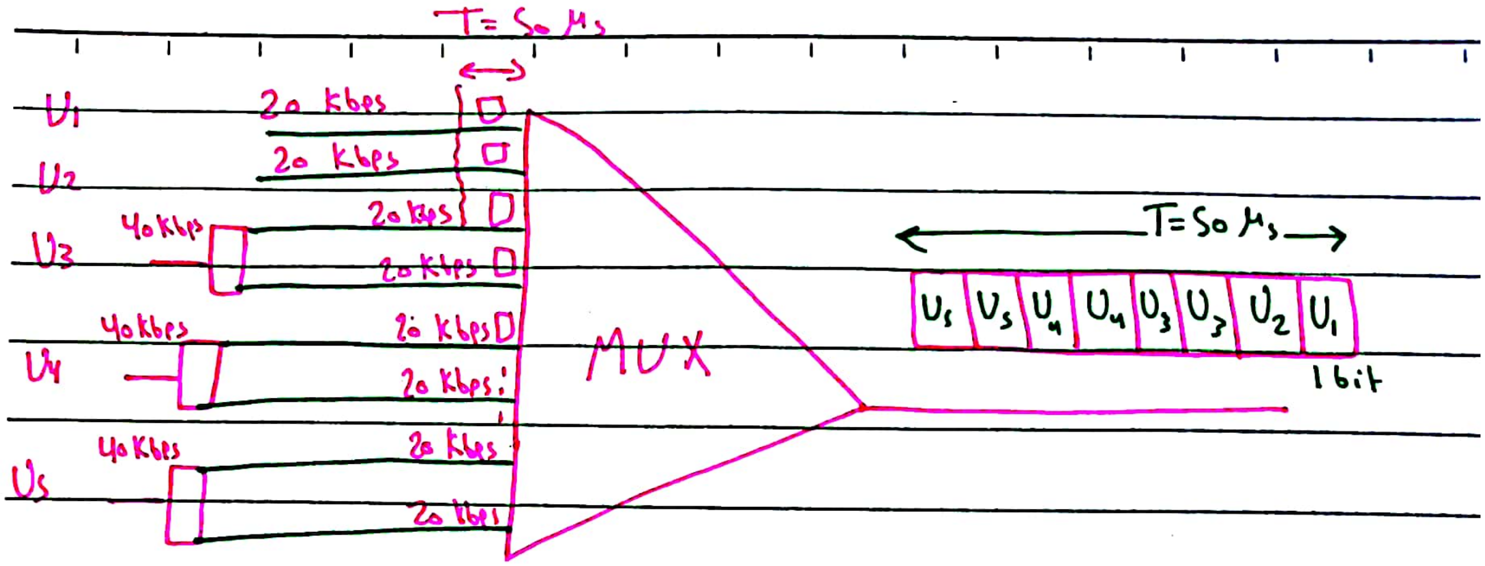
and an output link of 160 kbps.

Sol.

$$X + X + 3(2X) = 160 \text{ kbps}$$

$$8X = 160 \text{ kbps}$$

$$\boxed{X = 20 \text{ kbps}}$$



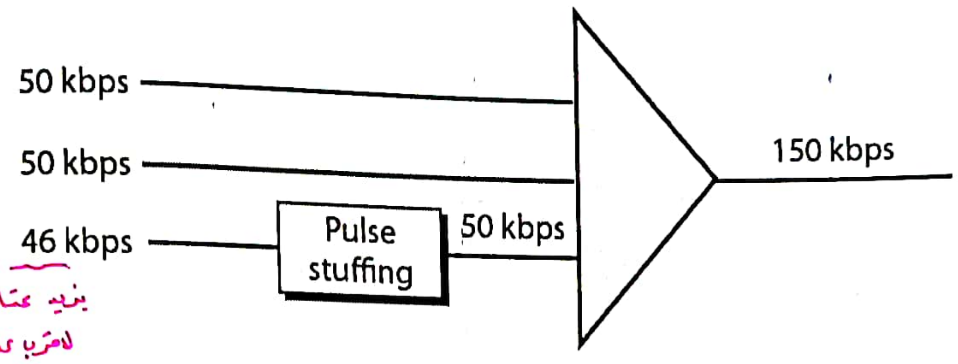
$$T = nT_b$$

$$= T_b \quad (n = 1)$$

$$U_1 = \frac{1\ \text{bit}}{50\ \mu s} = \boxed{20\ \text{Kbps}}$$

$$U_2 = \frac{2\ \text{bits}}{50\ \mu s} = \boxed{40\ \text{Kbps}}$$

Figure 6.21 *Pulse stuffing*

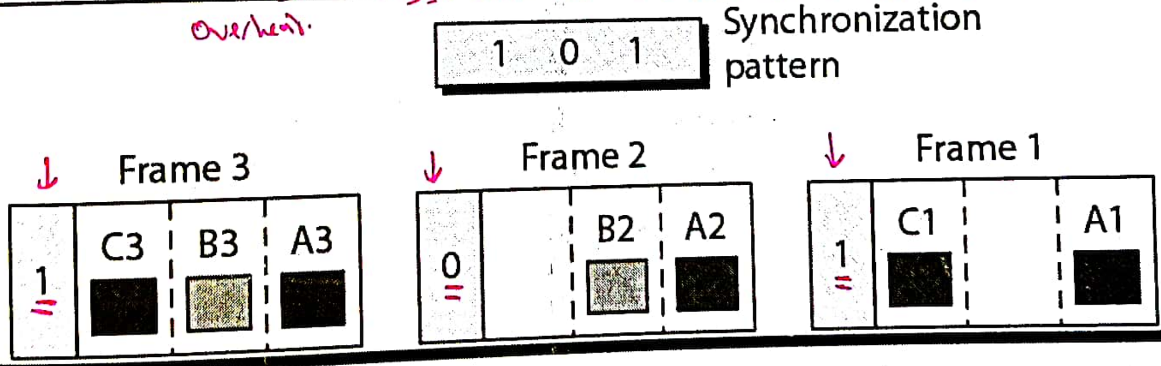


بنزيد عشان يطلع
لاقرب عدد صحيح

Figure 6.22 *Framing bits* → Frame ليقدر يعرف وين بداية ونهاية الـ

(لانه مرات بتداخل (خاصة اذا TD كانت مخزنة.

Overhead.



- 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

CamScanner

Example 6.10

مشاكلة

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.

$\boxed{2/1/S}$ كالدفتر

Solution

We can answer the questions as follows:

a. The data rate of each source is 250 × 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 × 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 × 33 = 8250 bps.

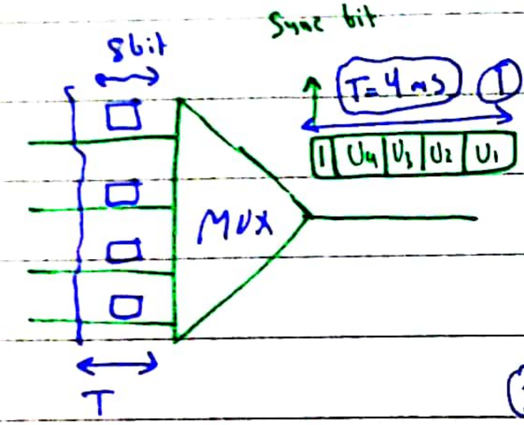
Example 6.10 (continued)

- 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.

CamScanner

Example 6.10

Sol



① Input rate $= 250 \ \frac{\text{char}}{1s} = 250 \ \frac{\text{Byte}}{s}$

$= 250 * 8 \text{ bits} = \boxed{2 \text{ Kbps}}$

② Data unit duration :

$T = nT_b = \boxed{4 \, ms}$

③ Frame rate $= \frac{1 \text{ frame}}{4 \ ms} = \boxed{250 \text{ frame/s}}$

Frame size $= 4(8) + 1 = \boxed{33 \text{ bits}}$

④ Data rate of the output link $=$ Frame size $*$ Frame rate

$= 33 \text{ bits} * 250 \text{ Frame/s} = \boxed{8250 \text{ bps}}$

CS CamScanner

⑤ Without framing bit = $\boxed{8000 \text{ bps}}$ → Frame size = 32 bits

CamScanner

Example 6.11

• 3X

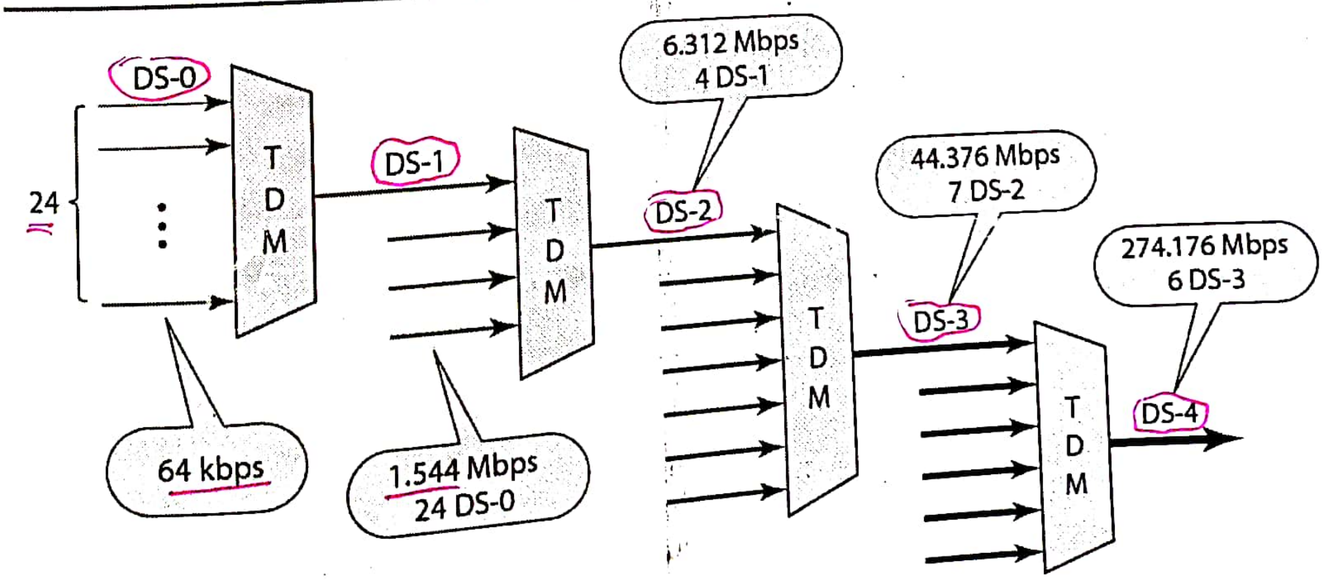
Two channels, one with a bit rate of 100 kbps and another with a bit rate of 200 kbps, are to be multiplexed. How can this be achieved? What is the frame rate? What is the frame duration? What is the bit rate of the link?

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.

6.45

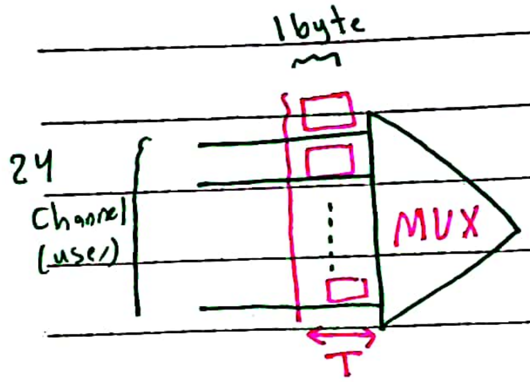
Figure 6.23 *Digital hierarchy*



CamScanner

23/5

T-1 LINE:



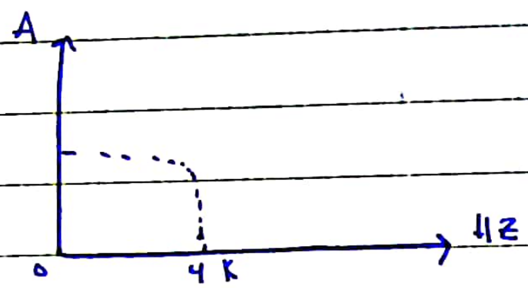
① Each channel has 64 Kbps.

Voice $\longrightarrow$ 0 – 8 K

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

$= 4000 - 0$

$= 4\ K$



$f_s \geq 2 f_{max}$

$f_s = 2(4000) = 8000 \ \frac{\text{Samples}}{s}$

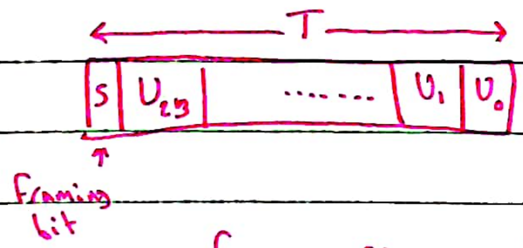
each sample is given an 8-bit word (1byte)

$R = (8000)\, 8 = 64 \ bps$ (PCM)

$T_b = \frac{1}{64 * 10^3} = 15.625 \ \mu s$

$T = nT_b = 8(15.625) = 125 \ \mu s$

②



frame size $= 24(8) + 1 = 193$ bit

CamScanner

③ Frame rate $= \frac{1 \text{ frame}}{125\ \mu s} = 8000$ Frames/s

④ Output rate = Frame rate * frame size

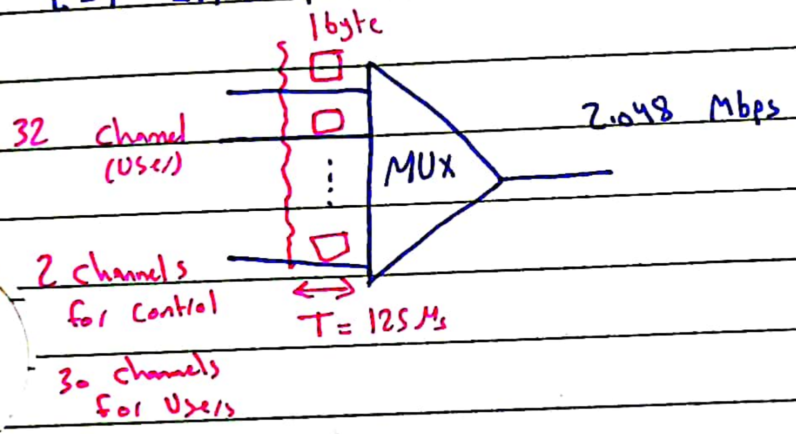
$= 8000 * 193$

$= \boxed{1.544 \text{ Mbps}}$ ← Ⓧ الـ Standard الأمريكي

OR

$= \frac{193 \text{ bits}}{125\ \mu s}$

E-1 LINE : ← الي بيشتغل عليه الأردن

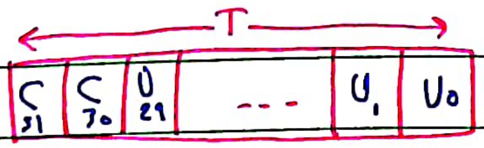


① Each channel has 64 kbps

$T_b = \frac{1}{64 * 10^3} = 15.625\ \mu s$

$T = nT_b = 8\,(15.625\ \mu s) = 125\ \mu s$

②



frame size $= 32(8) + 0$

$= 256$ bits

③ Frame rate $= \frac{1 \text{ frame}}{125\ \mu s} = 8000$ Frames/s

(4) Output rate = Frame rate * frame size

= 8000 * 256

= 2.048 Mbps

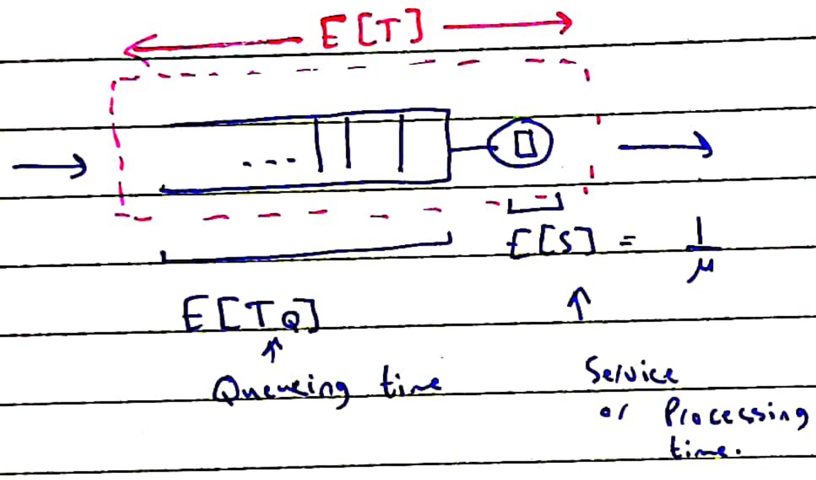
(OR)

$= \dfrac{193 \text{ bits}}{125 \, \mu s}$

☒

SM Vs TDM

E[T]



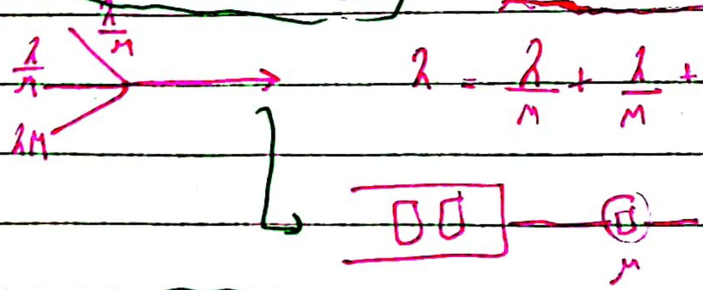
$E[T] = E[TQ] + E[S]$

$= \dfrac{1}{\mu - \lambda}$

$$\boxed{\text{TDM: } E[T] = \frac{M}{\mu - \lambda}}$$

M users

$\lambda = \dfrac{\lambda}{M} + \dfrac{\lambda}{M} + \dots$



$$\text{SM: } \boxed{E[T] = \frac{1}{\mu - \lambda}}$$

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	24*4 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& continous *

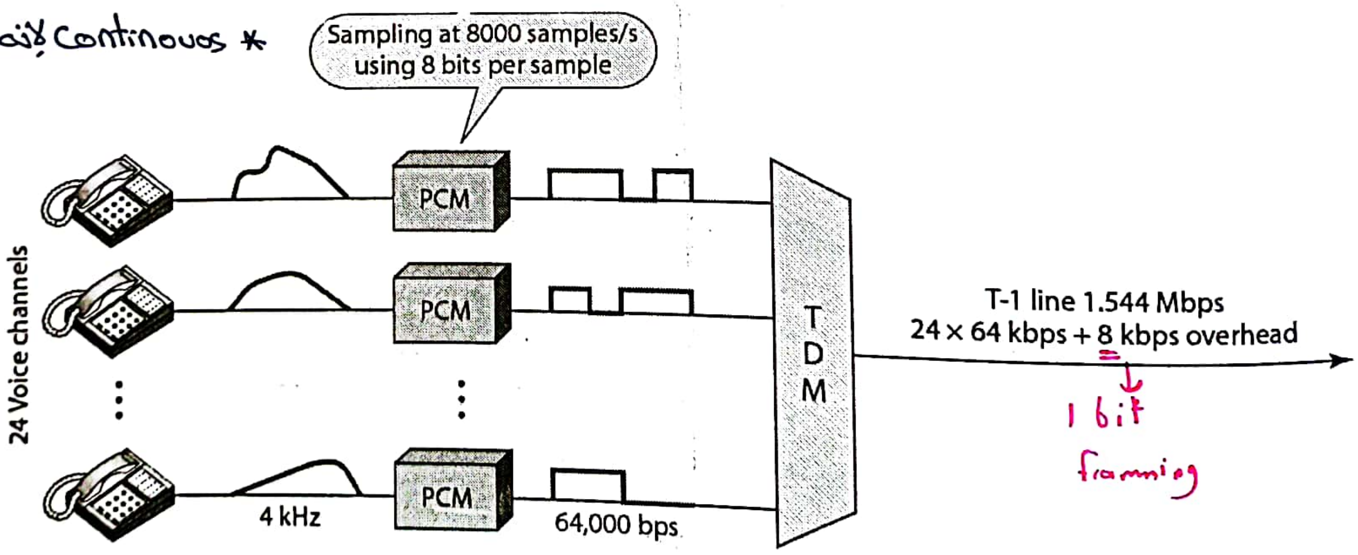
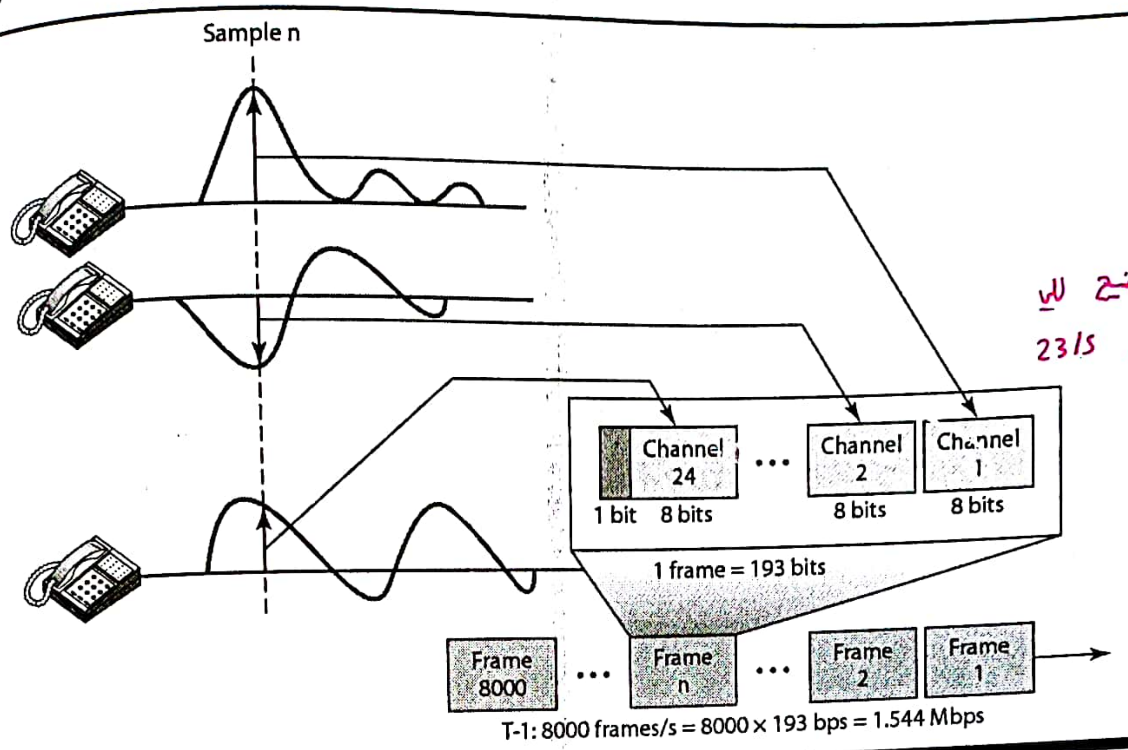


Figure 6.25 *T-1 frame structure*



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

فعليًا هم
32 بس 2
منهم مستخدمين
لأغراض ال
Control

المشكلة فيه إنه لو ماعندك Data بتضل ال Time slot موجودة. اقتصادياً مافي مشكلة بس من ناحية استغلال ال bandwidth في مشكلة،

الحل لهذا الاشي بال Statistical.

اللي ماعنده data مابكون إله Time slot

Synchronous TDM Application

لانه لازم كلهم متزامنا .

- Second-generation cellular phone technology called TDMA (Time-Division Multiple Access):
  - Digital version of the AMPS technology ← الجيل الأول
  - 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

بنلاقي حلول لل TDM لما مايكون في traffic كثير لانه ال TDMA هيك بده يستهلك سي.

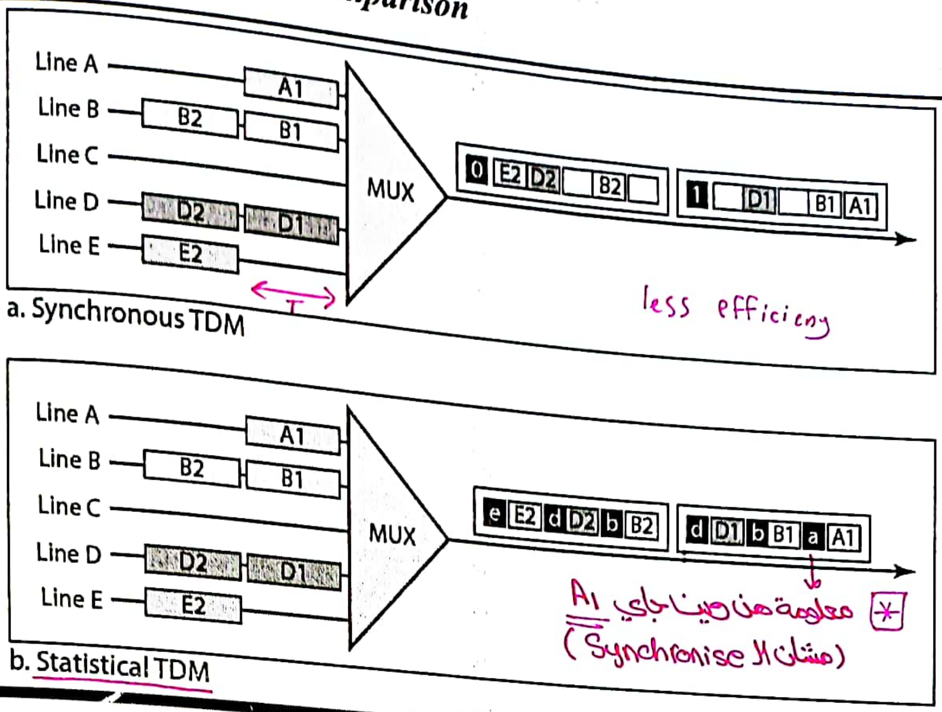
لازم نتأكد فيها إنه ال Sender وال reciever ذهم نفس الاشي لانه بطل يطلع كل شي ع الترتيب → "زيادة" "overhead"

Statistical Time-Division Multiplexing

بحط bit تعريفي وعنوان للداتا تاعت ال user وهيك بعرف مابين الداتا لمين.

- 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*



→ Frame size الـ
بختلف بينهم

6.53

SM Vs TDM
+ دفتر 23/5
من النوتا لا نزلهم

هون مش بنعمل Saving للـ bandwidth هون بنزيد
استخدامنا للـ bandwidth 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

* بنزيد الـ bandwidth مرات لأهداف
مختلفة أهمها الـ Security ← بنضطر لذلك عشان مرات بنحتاج كثير الـ Security
مقابل زيادة الـ bandwidth.

→ التنصت
→ jamming

* ما تخلي المستقبل يوصله الـ Signal الأصلية لإنه بيعت تشويش

Topics discussed in this section:

* **Frequency Hopping Spread Spectrum (FHSS)**
* **Direct Sequence Spread Spectrum (DSSS)**

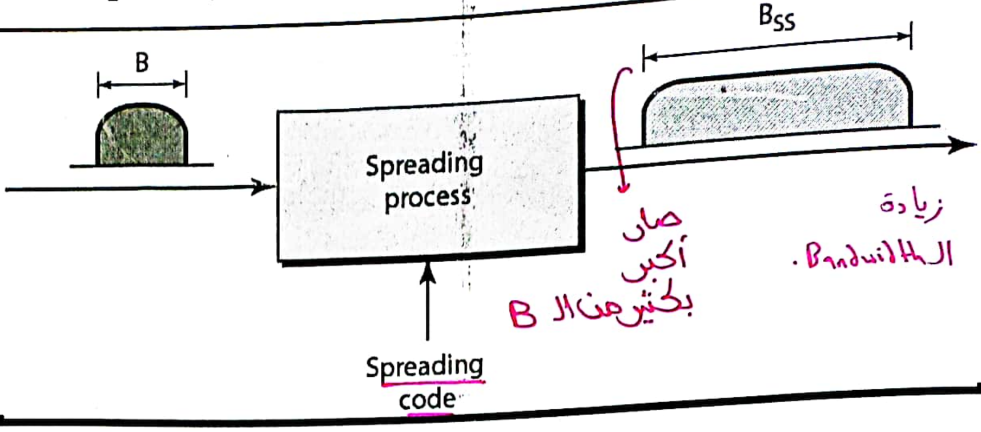
* المفروض أغير الـ frequency باستمرار
لأمنع التنصت فلذلك لازم يكون عندي
bandwidth أعلى.

* الـ Carrier هو النقطة الحساسة اللي
بنشتغل عليها بهذا الموضوع.

6.54

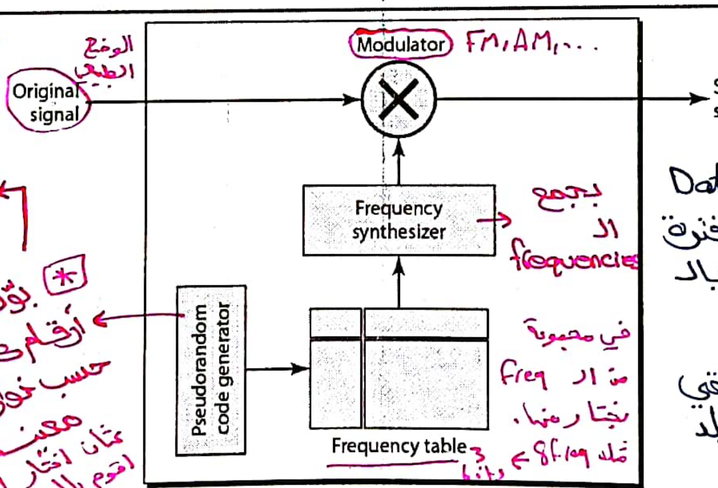
* كل اللي بنعمله إنو نخلي العملية أصعب لأي متخرب بهمه
يوصل الـ data ويخربها مع ذلك بضل في احتمالية إنه يوصلها.

Figure 6.27 *Spread spectrum*



- Principles of spread spectrum:
  1. The bandwidth allocated is much larger than needed allowing redundancy هي اللي بنستدعي إنو نكبر ال bandwidth.
  2. The spreading process occurs after (i.e.; independent of) the signal creation (Signal ← الأصلية) مستقل دائماً عن ال
- The two most well-known spreading techniques used are:
  1. Frequency Hopping Spread Spectrum (FHSS)
  2. Direct Sequence Spread Spectrum (DSSS)

Figure 6.28 *Frequency hopping spread spectrum (FHSS)*



ميزتها لو كطيتها نفس ال int بتطلعلك دايما نفس التسلسل للأعداد العشوائية اللي طلعوا أول مرة

لو بعطيها بذرة (Seed) بتكون int بولد منها أرقام عشوائية.

EX: Seed(int)

5
7
2
1

عشوائية بين 0 و 10

لو عرف المتنصت ال Seed فحيقدر يوصل لل Data الأفضل تغير ال Seed كل فترة (بال real time) صعبة بس بال theory ممكنة)

سمي Pseudo لأنه غير حقيقي (لو كانت نفس البذرة بحل يولد نفس التسلسل موري حقيقي مرة شكل).

- 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 → كل T_h بختار freq جديد.
- A frequency synthesizer creates a carrier signal with the corresponding frequency according to a lookup table
- Advantages: privacy and antijamming

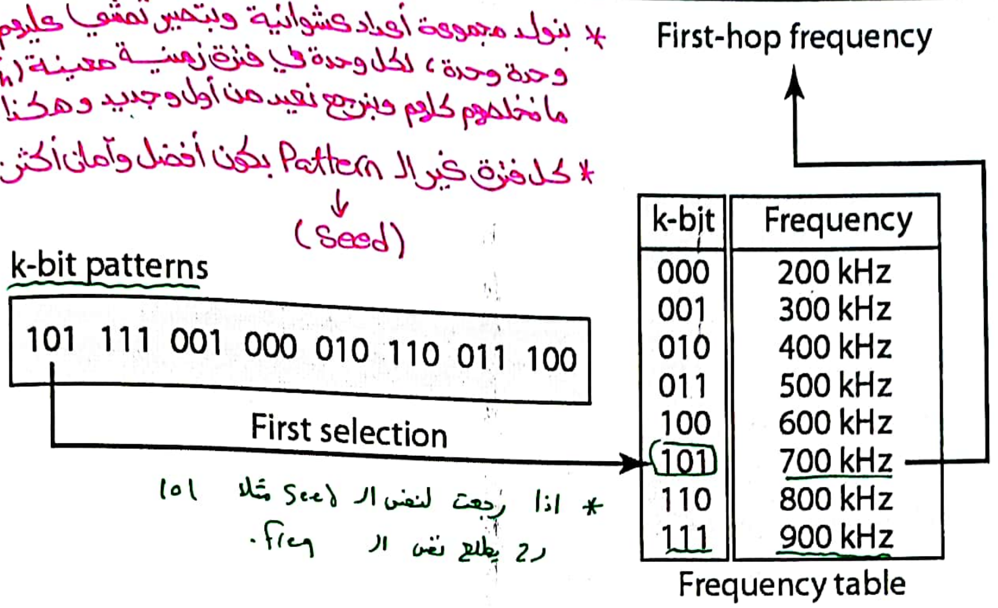
اذا عرفنا ال Freq بنعرف الأصل بس صعب تتعرف لأنه بتظل تتغير.

ال receiver رح يعرف لأنه رح يكون مع ال Signal الأصلية ال Signal كلها noise.

Figure 6.29 *Frequency selection in FHSS*

* بنولد مجموعة أعداد عشوائية وبنجيب تمشي عليهم وحدة وحدة ، لكل وحدة في فترة زمنية معينة (T_h) لحد ما نخلصهم كلهم وبنرجع نعيد من أول وجديد وهكذا.

* كل فترة نغير الـ Pattern بكون أفضل وأمان أكثر
(Seed)

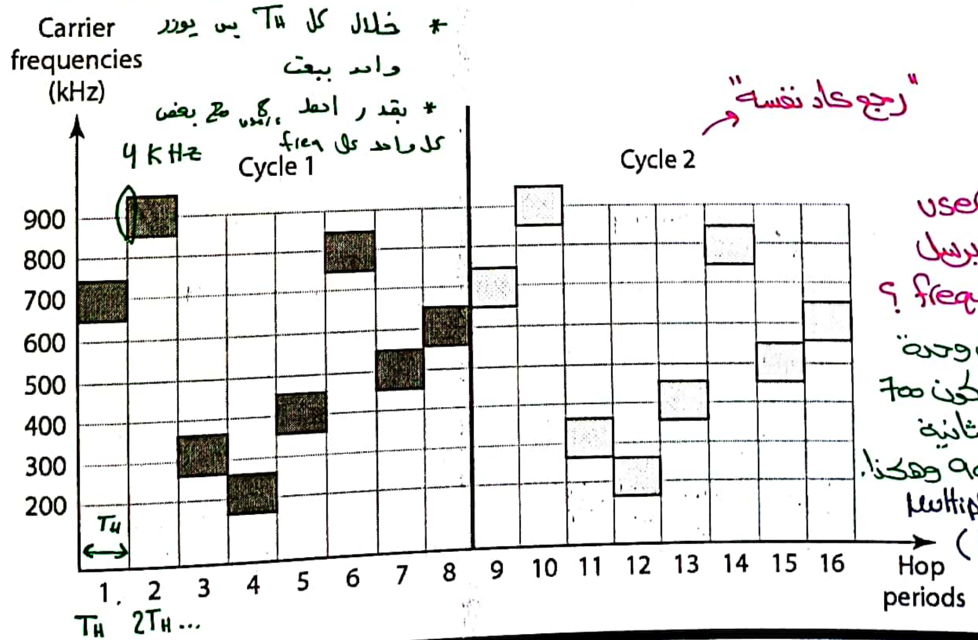


* اذا رجعت لنفس الـ Seed مثلاً 101 رح يطلع نفس الـ freq.

Figure 6.30 *FHSS cycles*

Bandwidth = 100 kHz

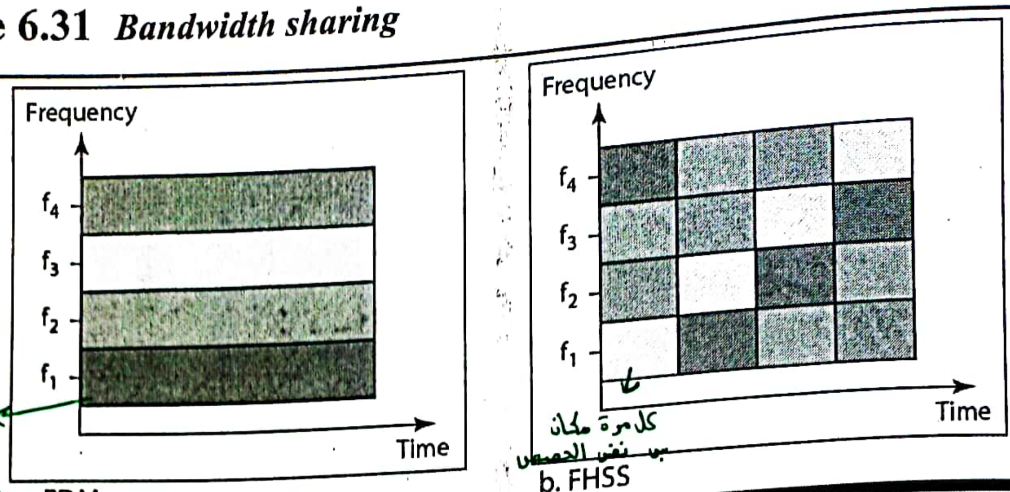
* خلال كل T_H بس يوزر واحد بيبعت
* بقدر احط يوزرين مع بعض كل واحد الو freq



* هلا بينفع أدخل user ثاني وأخليه يرسل نفس الـ frequencies ؟
بنقدر بس بأول وحدة مثلاً ما بزبط تكون 700 الثانية و بالثانية ما بزبط تكون 900 وهكذا.
(هذا اسمه Multiplexing مع الـ hopping)

هون بال hop period الوحدة بنقدر نستخدم 8 users.

Figure 6.31 *Bandwidth sharing*



*ما بصير في Collision بال FHSS

الـ BW كاملة

الطبيعي ← a. FDM

كل مرة مكان بس نفس الحصص

b. FHSS

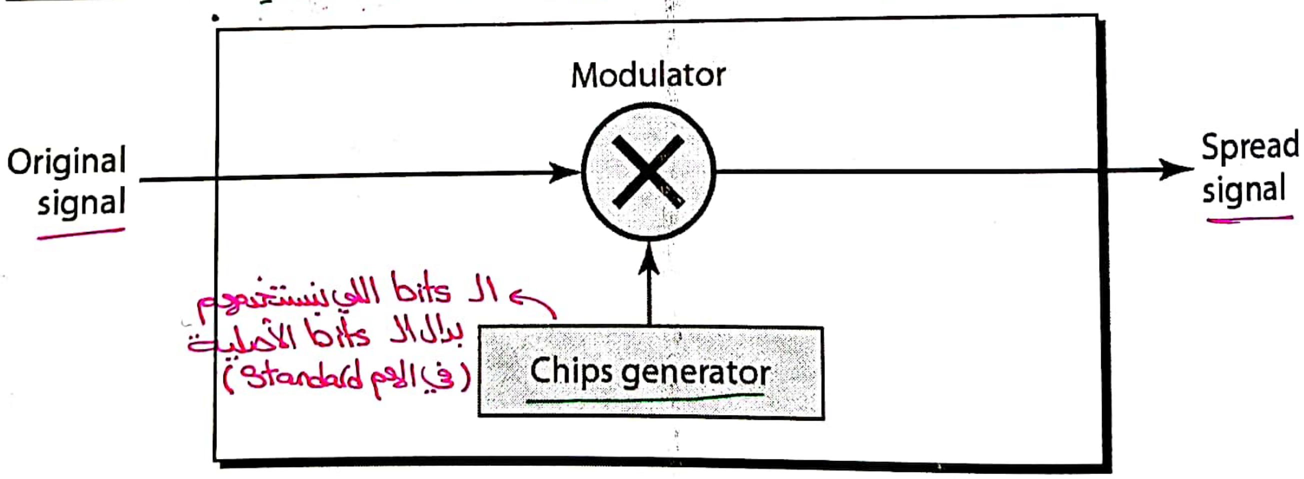
- *For M hopping frequencies, M channels can be multiplexed into 1*
- *In FDM, each channel uses 1/M of the bandwidth with fixed allocation*
- *In FHSS, each channel uses 1/M of the bandwidth but with a dynamic allocation from hop to hop*

* الـ wifi بستخدم الـ frequency hopping *

Figure 6.32 *Direct Sequence Spread Spectrum (DSSS)*

بنشيل الـ bit الواحدة لمجموعة من الـ bits عشان يصير في تمويه.

[x] في bits تضيف زيادة على الـ signal للتمويه



الـ bits اللي بنستخدمهم بدال الـ bits الأصلية (في الـ standard)

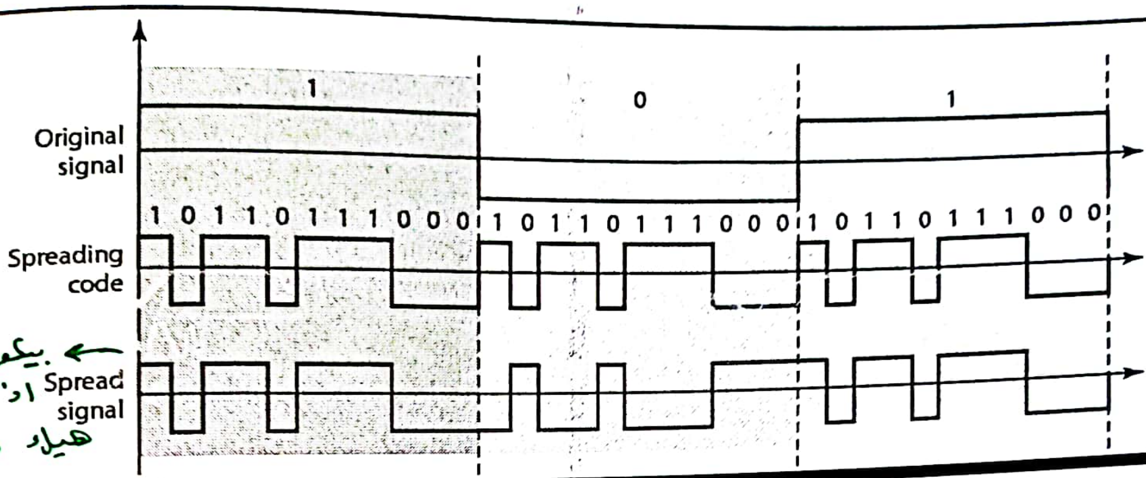
بيدمر الـ bandwidth

- 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 الـ data rate وهذا اشي مش كثير منيح بس بالمقابل بنزيد الحماية للـ data.

Figure 6.33 *DSSS example*



* (ضروب)

← بيكون متعارف اذا اتقبل هيك يعني إ فلازم ماحدا يعرف Spreading . Code

- 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 → بيعتمد 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 الـ 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.

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* كيف الـ receiver يكتشف انه الداتا الي وصلته فيها خطأ ويصلحه.

* ممكن لما اكتشف الخطأ بخليه يرجع يعيد ارسالها الي.

Note

Data can be corrupted during transmission.

مثل الـ noise والـ attenuation. (بتصير بالـ 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 فهون بأثر الـ error.

10-1 INTRODUCTION

Let us first discuss some issues related, directly or indirectly, to error detection and correction.

* ال detection ابسط

Topics discussed in this section:

في errors بتحدث على ال bit وفي ال multiple مجموعة من ال bits.

* بتعتمد على ال data rate اللي بنشتغل عليها وعلى Period of the noise.
* الشائع في ال data comm إنه تضرب مجموعة bits مش وحدة (burst loss)

* Types of Errors
* Redundancy
* Detection Versus Correction
* Forward Error Correction Versus Retransmission
* Coding
* Modular Arithmetic

بتقسم وتأخذ الباقي

$5 \bmod 2 = 1$

$4 \bmod 2 = 0$

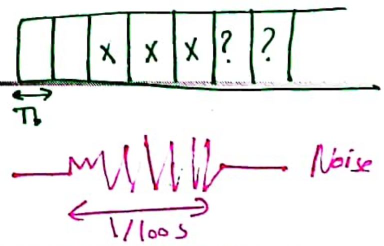
: Automated Repeat Request *

أعمل ال error detection وإذا طلع عندك خطأ في ال Packet اطلب إعادة إرسالها.

$R = 1200$ bits/s

زمن ارسال ال bit الواحدة $T_b = \frac{1}{1200}$ s

10.3



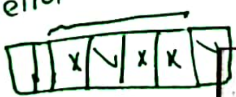
رح يضربوا كلهم بعض 12 bits $= \dfrac{\frac{1}{100}}{\frac{1}{1200}}$

Noise

$1/100$ s

Burst error

ولو اجزاء مضروبة (اول bit و اخر bit) بعتبره كله مضروب



Note

تغيرت حالة bit واحدة فقط من 0 إلى 1 أو العكس ← في أسباب كثير للتغيير مثل ال noise أو ال attenuation

In a single-bit error, only 1 bit in the data unit has changed.

Modular Arithmetic:

$\square \pmod N$

$0 \rightarrow N-1$

mod 2

$\underline{0}$ $\underline{1}$

$\boxed{+}$
$1 \oplus 0 = 1$ ← mod 2
$0 \oplus 0 = 0$
$0 \oplus 1 = 1$
$1 \oplus 1 = 0$

$\boxed{-}$
$1 \oplus 0 = 1$
$0 \oplus 1 = 1$
$1 \oplus 1 = 0$
$0 \oplus 0 = 0$

10.4

ما في فرق بين الجمع والطرح
فبحولها ل $\oplus$ (Xor)

دفتر 30/5

CamScanner

$\boxed{30/5}$ * Single-Bit Error:

X

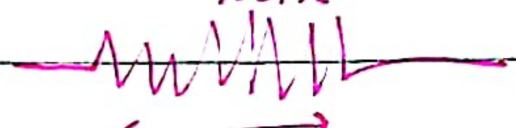
0	1	1	0	1	0	0

* Burst Error: (أكثر)

0	1	1	0	1	1	0	1

Burst Error

Noise



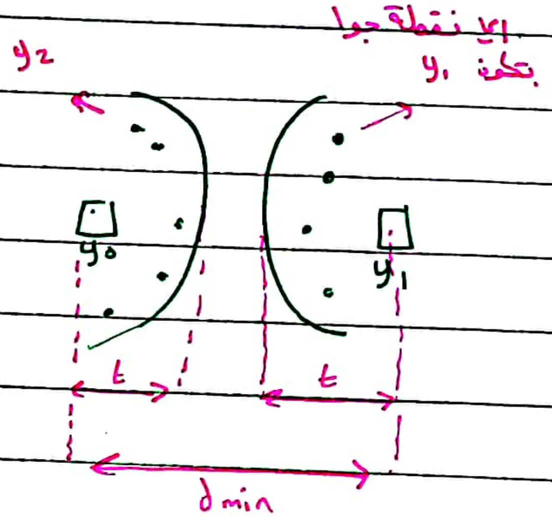
1/100 s

$R = 1200$ bps

$T_b = \frac{1}{1200}$ s/bit

$\frac{1/100}{1/1200} = \boxed{12 \text{ bits}}$

Correction



$d_{min} > 2t$

$\boxed{d_{min} = 2t+1}$

For error Correction

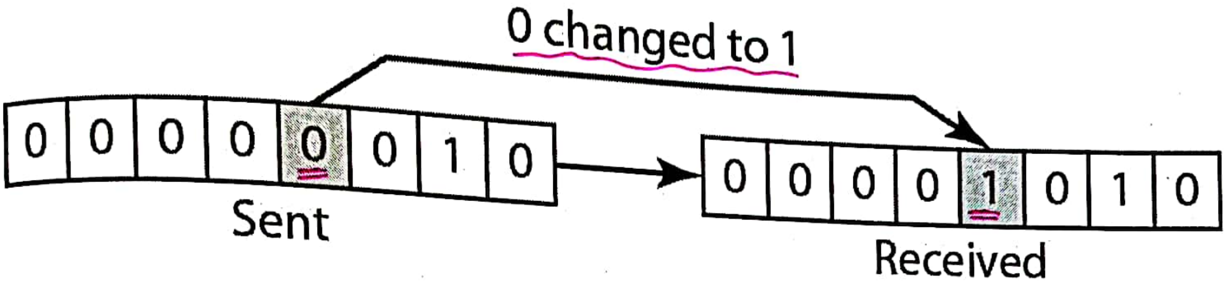
بمثال سلايد 10.19

$d_{min} = 2$

$s = 1$

$t = 0$ فما بيقدر اصلح ايّ bit

Figure 10.1 *Single-bit error*



10.5

سواء كانوا جمب بعض
أو بعد 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.

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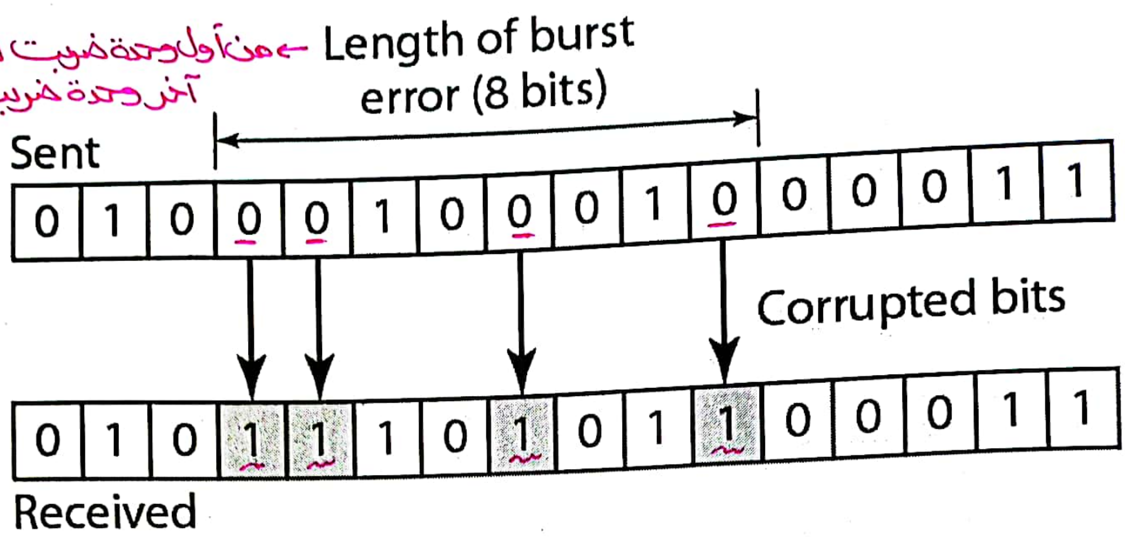
Figure 10.2 *Burst error of length 8*

من أول وحدة خربت لعند آخر وحدة خربت ←



10.7

Note

قاعدة لاكتشاف الخطأ. مثل الـ Parity bit

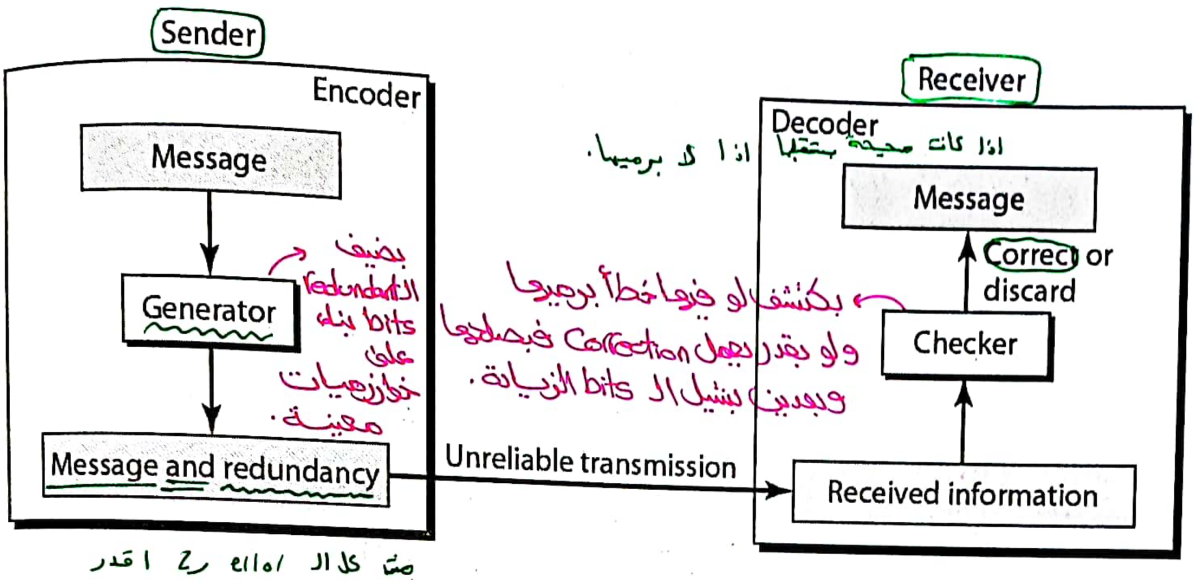
To detect or correct errors, we need to send extra (redundant) bits with data.

"لازم يكون موجود" "عشان نقدر نعمل detection و correction"

لازم تنبعت.

10.8

Figure 10.3 *The structure of encoder and decoder*



مش كل الـ errors رح اقدر اعملها Detection

10.9 * في تطبيقات بتعتمد فقط عالـ detect لحتى ماترفع الـ Packet للـ layers اللي فوق، بتهمل بعيين مباشرة بترميه لا وتتم إذا سيعاد إرسال الـ Frame ولا لأ عشان توفر وقت وبعضهم لأ 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 ماتشتغل بكل الأعداد الصحيحة
بنشتغل subsets منها.

Note

* أصغر modulo هو Modulo-2. → binary modulo.

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 ← فعملية الجمع والطرح بال binary : (بنهمل ال carry وال borrow) ال mod نفسه

EX : mod 2

$$\begin{array}{r} 101 \\ +\ 011 \\ \hline 110 \end{array} \qquad \begin{array}{r} 101 \\ -\ 011 \\ \hline 110 \end{array} \Longrightarrow \text{XOR}$$

أهملناها

كأنو فاضلت 1 ولاكأنه استلفنا

سهل وكثير مكان إنه نعملها implementation عال hardware.

← 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)

10.11

Figure 10.4 *XORing of two single bits or two words*

mod N: 0 → N-1 mod 2: 0 1

$0 \oplus 0 = 0$ $1 \oplus 1 = 0$

a. Two bits are the same, the result is 0.

$0 \oplus 1 = 1$ $1 \oplus 0 = 1$

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

10.12

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bits redundant فكرته اني بقسم الفايل ال Data Chunk وكل Data Word بضيف عليها

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 ال Codeword وبين ال dataword.

Topics discussed in this section:

Error Detection
Error Correction
Hamming Distance
Minimum Hamming Distance

* k = 2 bits , n = 3 bits

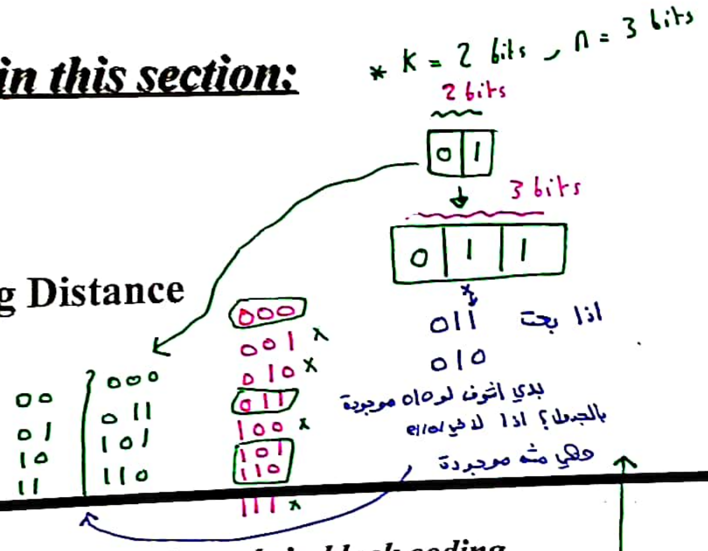
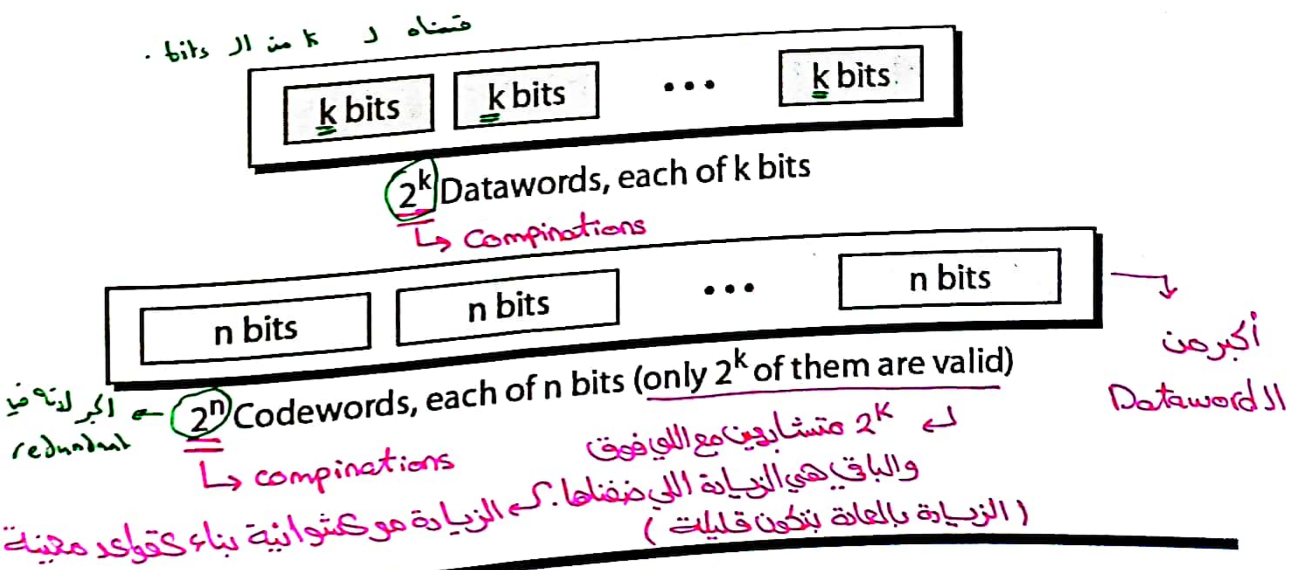


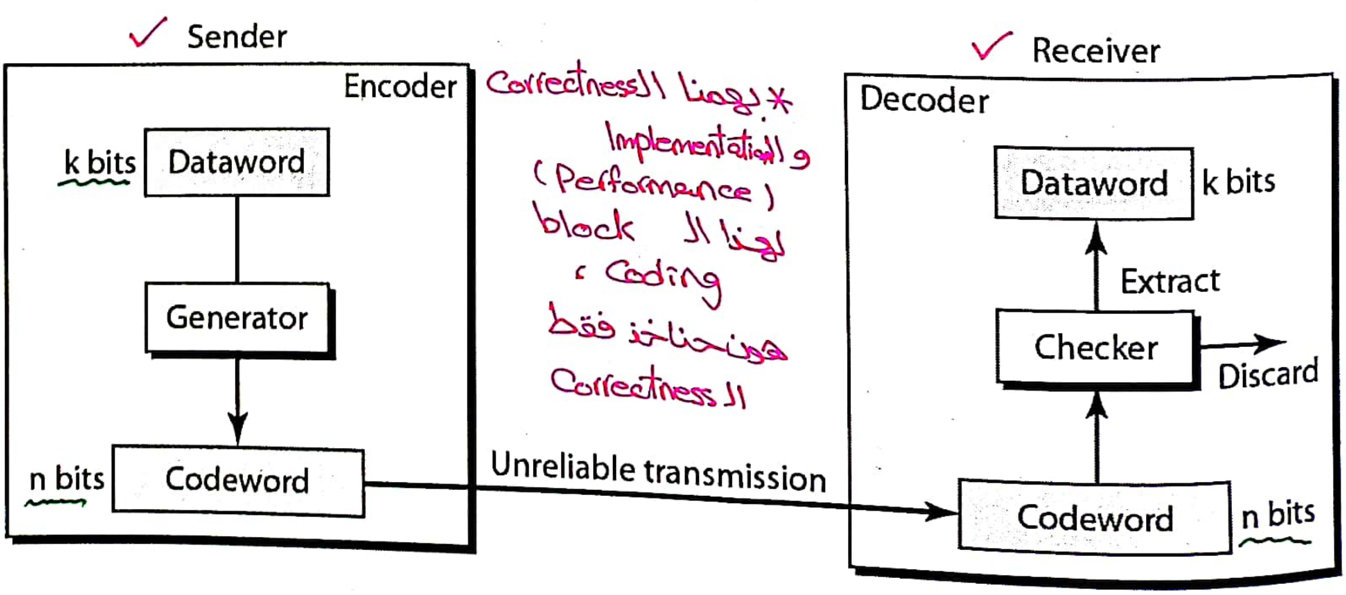
Figure 10.5 *Datawords and codewords in block coding*



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*



* يهمنا الCorrectness
والImplementation
(Performence)
لهذا الblock
Coding ،
هنتاخذ فقط
الCorrectness

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Example 10.2

$\rightarrow 2^2$ datawords $\rightarrow 2^3$ 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.

بنطلعه من 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 ال استقبل 00 لإنه خربطت ب 2 bits
فما قدر يعمل detection.

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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 هيك مارح يعتبرها ك data

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)

* 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 إنو هذا مش موجود بالجدول فبعمله discard فهيك عمل error detection بشكل صحيح.

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.

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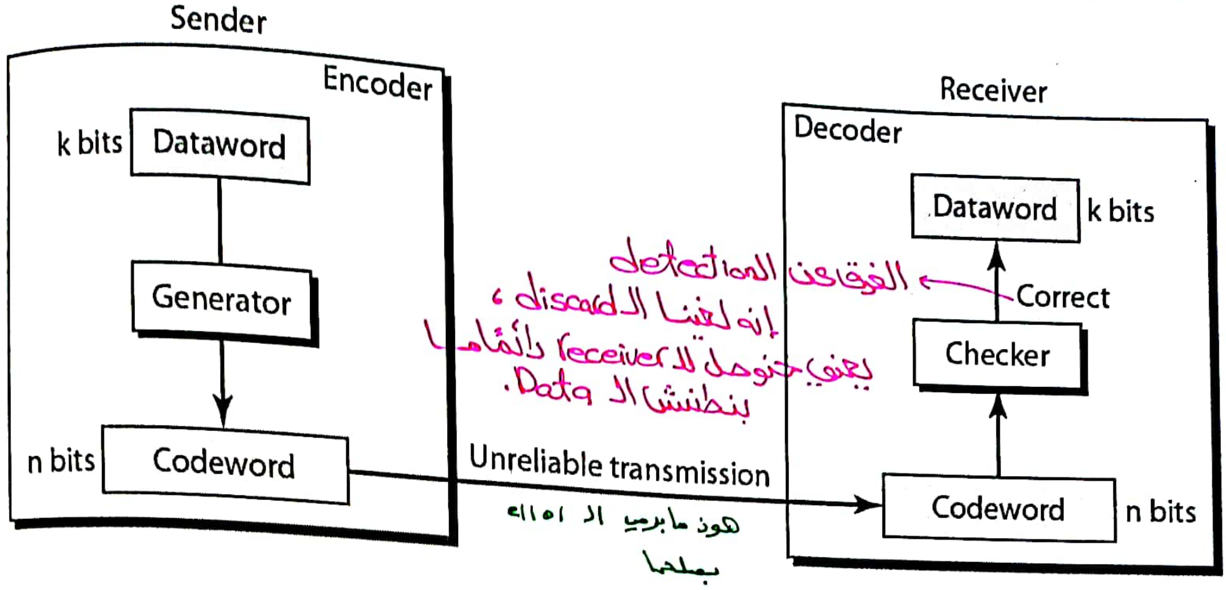
Figure 10.7 *Structure of encoder and decoder in error correction*



الفوقان الـ detected إنه لغينا الـ discard ، يعني ما توصل لل receiver دائماً بنطنش الـ Data.

هون ما برمي الـ data بصلحها

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 (الاصلية)*. The codeword is corrupted during transmission, and 01001* (أي استقبلناها) *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 فيها مشكلة

3of5 عالدفتر

10.22

Example 10.3

Receiver

الي ارسلناها

01

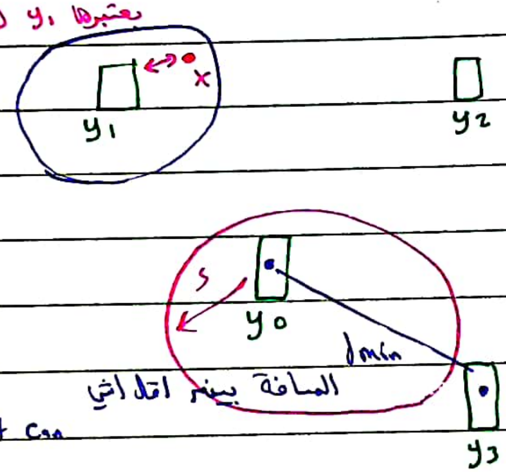
⇓

01011 → 01001

استقبلناها

*هدش موجودة بال table فخيل منا error وبديا اعرف لوكانت لواحدة من الجدول ال قريبة منها

يعتبرها y_1 لانه هي الاقرب الها لعدد ال bits المختلفة بأخذ اقل اشي



اي اشي بيقع جوا الدائرة بينتمي لل y_1

□ : Valid code word

Hamming Distance

$d(1010, 1001) = 2$

$$\begin{array}{r} 1010 \\ \oplus\ 1001 \\ \hline 0011 \end{array} \rightarrow \text{عدد ال 1}$$

# of errors that can be detected

اكبر قيمة لل s

$s + 1 = d_{min}$

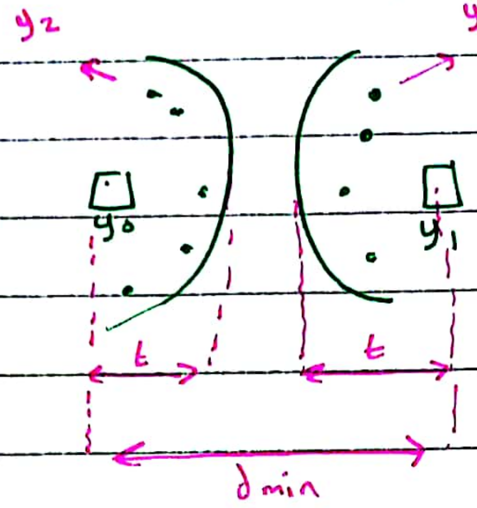
$s < d_{min}$

$1 = d_{min} - s$

Five Apple

Correction

إذا نقطة جوا
بتكون y_1



$d_{min} > 2t$

$\boxed{d_{min} = 2t+1}$

for error correction

بمثال سلايد 10.19

$d_{min} = 2$

$s = 1$

$t = 0$ فما بتقدر اصلح اي bit

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 1 bit 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 1 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 اقل d min
10	10101
11	11110

* 01001
00000
01001 = 2

* 01001
01011
00010 = 1 ✓

* 01001
10101
11100 = 3

* 01001
11110
10111 = 4

10.24

* بال Computer بحتاج ال xor لأعمل شغلة المقارنة، الناتج بمثل الاختلافات.
(عدد ال ones بمثل عدد الاختلافات).

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Note

عدد الاختلافات بال bits دائما بين 2words.

* **The Hamming distance between two words is the number of differences between corresponding bits.**

والاختلافات بديه لما يكون صاير error بال data اللي وصلت مقارنة بالي انبعتت.

Ex: * $d(C_1, C_2) = 0 \rightarrow$ This mean that $C_1 = C_2$ (no error)
↳ 10110 ↳ 10110

* $d(C_1, C_2) = 1 \rightarrow$ This mean that $C_1 \neq C_2$ (with error)
↳ 10110 ↳ 11110

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

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أقل hamming بين set of words

Note

بأخذ كل الاحتمالات وبأخذ اقل ما بعد (d)

* **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 .

* *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.

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*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 $\underline{\underline{3}}$.

عشان اضمن عملية detection

لـ S من الـ error ← $d_{min} = S+1$

لانها لو كانت بس S فممكن الـ Codeword الخاطئة تمثل
عاينا صحيحة وهاي مشكلة. (حنشوف بالمثال الجاي)

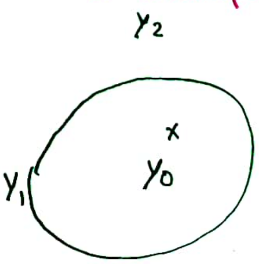
Note

* القطر بدي اياه
نصف قطر يكون بـ
d_{min} والفرق بدا
اياه = منيلة بغير
ما ادخل على الـ...

*To guarantee the detection of up to s errors in all cases, the minimum Hamming distance in a block code must be $d_{min} = s + 1$.

نصف قطر

example : 1 bit error ⇒ $S=1 \rightarrow d_{min} = 1+1 = 2$



هدول Valid Word والفرق
بينا كل وحدة والتانية = الـ hamming.
اي نقطة تقطع جوا الدائرة ,2
تطلع hamming معين اقل
من المسافة بين Y_0 و Y_1 او Y_0 و Y_2

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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 الـtable لإنه كل الـ table أقل شي كان d=2

$d_{min} = 2$

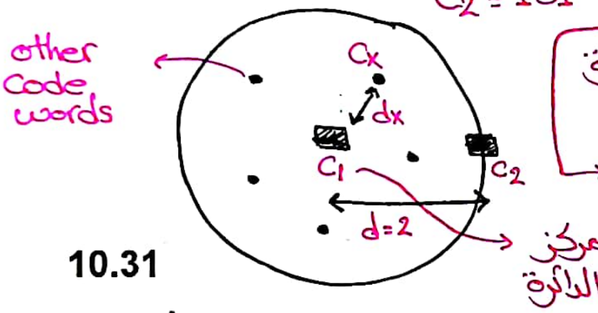
$C_1 = 011 \rightarrow C_x = 001 \rightarrow d(C_x, C_1) = 1 \rightarrow d_x < d_{min}$

$C_2 = 101$

$C_y = 101$

فلو صار خطأ بـ 2 bits ممكن أدخل داخل محيط الدائرة وتنحسب الـ code word إنها صح وهذا لا يعتبر error detection.

$\rightarrow d(C_y, C_1) = 2$



مركز الدائرة → (إحدى الـ code words اللي بالجدول)

Example 10.8

في خطأ بـ 2 bits بس ما التقط فالـ error بكون داخل الـ system فعشان هيك الـ min خلوه بيساوي 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.

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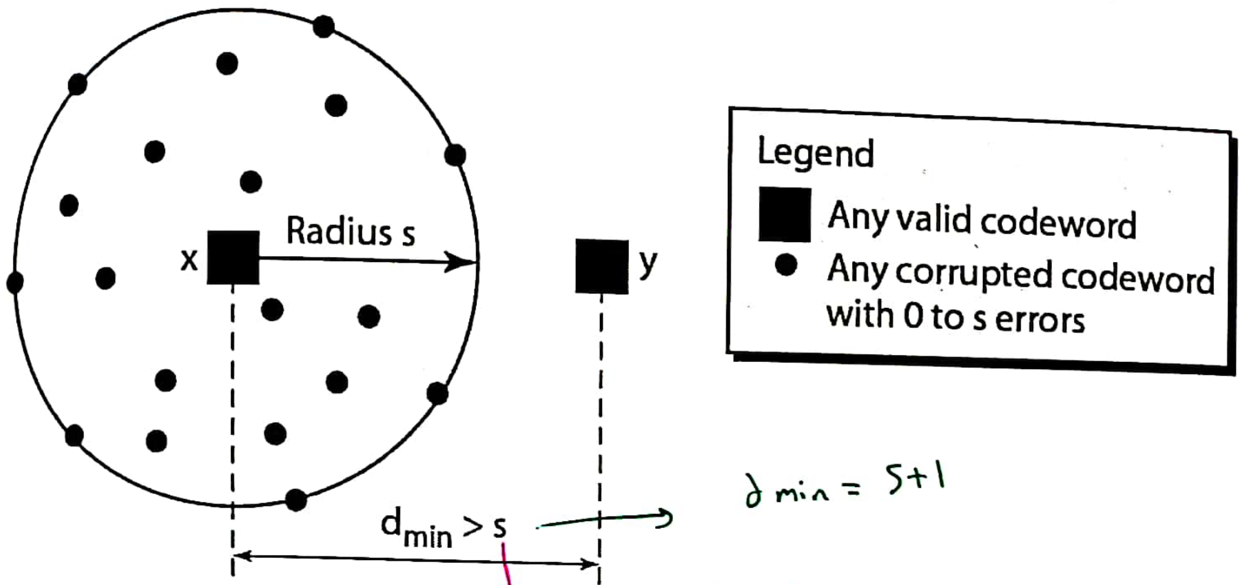
Figure 10.8 *Geometric concept for finding d_{min} in error detection*



أي نقطة تقع على الـ Circle.

* for Error Detection : $d_{min} = s+1$ and $(d_{min} > s)$

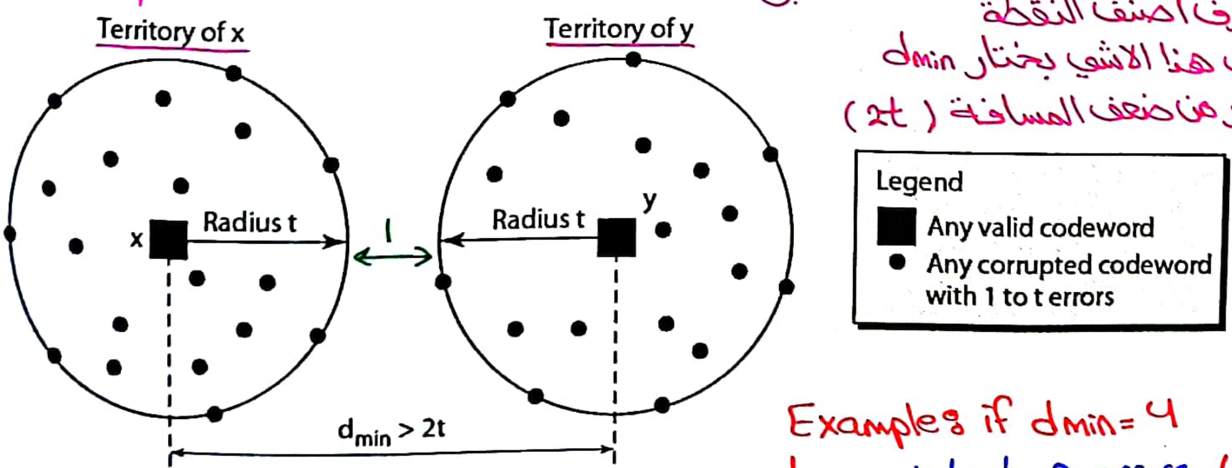
* for Error Correction : $d_{min} = 2t+1$ and $(d_{min} > 2t)$

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 حيبطل أعرف أصنف النقطة فلأتجنب هذا الاشي بختار dmin تكون أكبر من ضعف المسافة (2t)



ما بدي الدائرتين يتقاطعوا

$d_{min} = 2t + 1$

Examples: if $d_{min} = 4$

↳ can detect 3 errors $(s+1)$

↳ can correct 1 error $(2t+1)$

Note

≡* To guarantee correction of up to t errors in all cases, the minimum Hamming distance in a block code must be $d_{min} = 2t + 1$.

10.35

Example 10.9

A code scheme has a Hamming distance $d_{min} = 4$. What is the error detection and correction capability of this scheme?

detection
$d_{min} = s + 1$
$4 - 1 = s$
$\boxed{s = 3}$

correction
$d_{min} = 2t + 1$
$4 - 1 = 2t$
$t = \frac{3}{2}$
$\boxed{t = 1}$

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 ال bit رايح wasted

10.36

Valid code
* مجموع اي 2 word
بده يساوي Valid word

"حنتطلع لوجهة 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 الـ Codewords (Table 10.2)

Topics discussed in this section:

Minimum Distance for Linear Block Codes
Some Linear Block Codes

* 10101 ⊕ 00000 = 10101 ← Valid ومتاحة بالجدول

* فإذاً أول شرط من شروط الـ linear code انه إذا أخذت 2 Codewords من الجدول وعملت بينهم XOR حيطلعولك Codeword ثانية موجودة بالجدول برضه

* In general:

Dataword (k bits)	1 bit (additional bits)

n bits code word (k+1)

↳ dmin = Single bit error (s=1) → dmin = s+1 = 2

* n = k + 1 ← لانه بس زدنا bit وحدة ، ممكن تكون الزيادة أكثر من 1.

* **Note**

In a linear block code, the exclusive OR (XOR) of any two valid codewords creates another valid codeword.

*

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مثلا

$$\begin{array}{r} 000 \\ \oplus\ 011 \\ \hline 011 \end{array} \qquad \begin{array}{r} 011 \\ \oplus\ 101 \\ \hline 110 \end{array}$$

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.*** → Example: $\begin{array}{r} 101 \\ \oplus\ 110 \\ \hline 011 \end{array}$ → 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.***

10.39

*العمليات الحسابية عال linear أسهل عشان هيك ما بنروح عال Non-linear.

Example 10.11

بنقدر نحدد dmin انه ال Codeword اللي فيها اقل عدد من ال 1 (بدون اي كلها 0) في حالة linear.

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 المهمة في ال linear إني قدرت أحصر ال Set لإنه بعرف إنه كل 2 Valid codewords بينعمل Xor بتطلعلي Valid codeword بال table.

فكرتها إنك تخلي عدد ال 1's في ال word
دائمًا يا even يا odd ، احنا حنتعامل مع ال even

Note

detection بـ

*** A simple parity-check code is a single-bit error-detecting code in which $n = k + 1$ with $d_{min} = 2$.**

1 redundant

$d_{min} = S + 1$

$d_{min} = 2 \rightarrow S =$

If $d_{min} = 2$

$S = 1$ detection

$t = 0$ Correction

ما بيقدر اعدل اي

Data word

$k = 3$

001	1

total # of bits = 3+1 = 4

بضيفها بحيث ال total number of 1's تكون even .

10.41

* حنلاحظ بـ Table 10.1 : إنه آخر bit كنا نضيفها هي Parity-check .

* Table 10.2 مش Parity check ، لأنه أضفنا أكثر من 1 bit *

Codeword data word

Table 10.3 *Simple parity-check code C(5, 4)* min d = 2 ← ثابتة زي قاعدة

dmin

Datawords	*Codewords* (ضفنا 1 bit)	*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

ولازم عدد ال 1 يطلع even -

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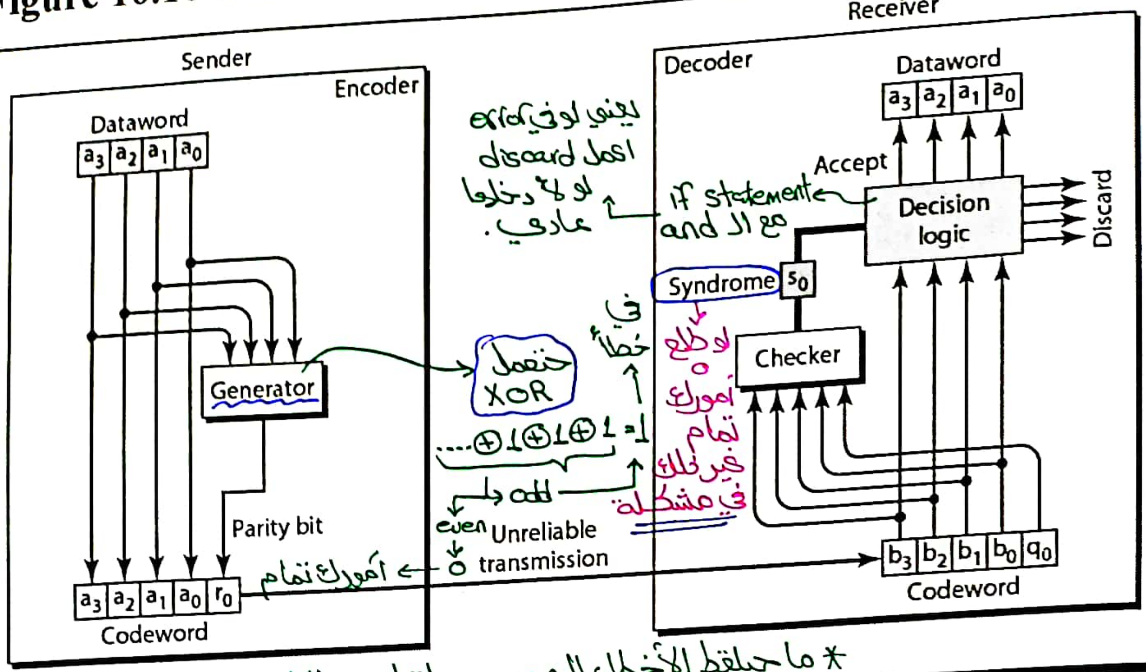
بيعمل detect فقط لل error 1.

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 بشرح الآلية لهاي 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.* → برفعوها 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.* Syndrome = 1
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

لـ رغم إنه ال data فعليًا ما فيها خطأ

(هاي من الأضرار الجانبية لموضوع ال Coding مع ال Parity)

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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. → مع إنه صار في خطئين واحد في ال data وواحد بال Parity

ما رح اكتشف الخطأ
لأنه الكود بعرف خطأ واحد بس

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 ال detect لل error مع Probability.

* من أهم ال error detect 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.** *

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Note

* All Hamming codes discussed in this book have $d_{min} = 3$.

2 errors ←

Can detect up to 2 errors ↲
↳ s=2

* The relationship between m and n in these codes is $n = 2m - 1$.

$d_{min} = s+1 \rightarrow s=2$ detect 2 errors

$d_{min} = 2t+1 \rightarrow t=1$ Correct 1 error

$\xleftrightarrow{k}\xleftrightarrow{m}$

$\xleftrightarrow{\quad n \quad}$

① $k = n-m \rightarrow m = n-k$

② $n = (2^m)-1$

10.47

طريقة أفضل للـ error detection لأنه بـ Parity check وحدة كنا نقدر نحمل بس للـ odd error الـ even error ما نقدر

Figure 10.11 *Two-dimensional parity-check code*

((بحسب الـ Parity على الـ rows و الـ Columns))

* الـ data بترتيبها بـ rows ، الأصل نبعتها 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

إضافتهم عشان نخلي عدد الـ 1's even

بحط كل الـ word بجدول بحسب Parity لكل row بعدين لكل Column بحسب الـ Parity.

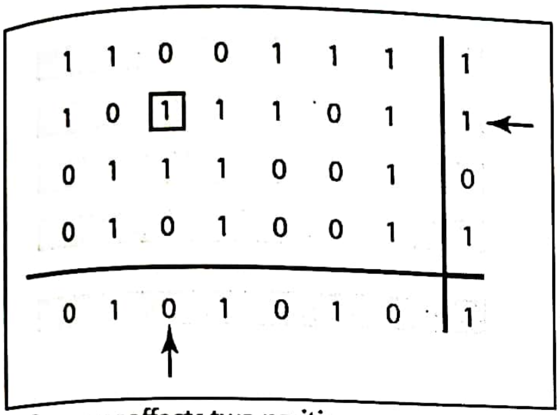
a. Design of row and column parities

* لو ضربت bit بتأثر على 2 parity *

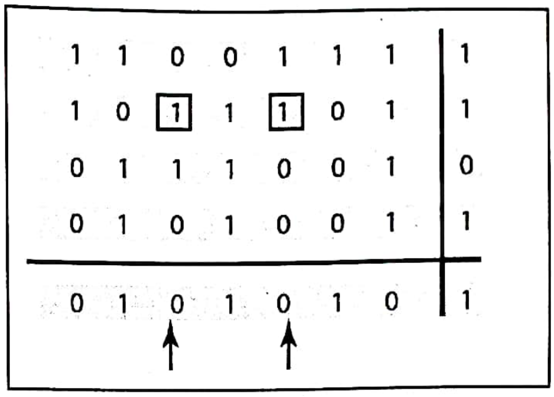
10.48

بس زيادة overhead / "أفضل من ال one"

Figure 10.11 *Two-dimensional parity-check code*



b. One error affects two parities

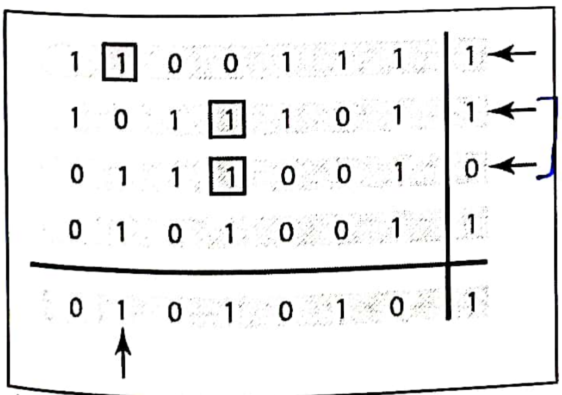


c. Two errors affect two parities

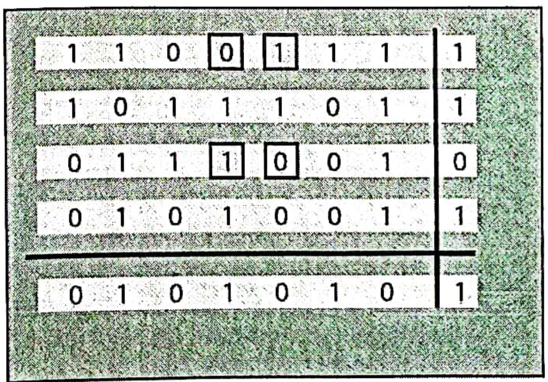
* لو كانت كل ال data موجودة بـ row واحد ما كانت حتقدر اعمل detect زي هون *

Figure 10.11 *Two-dimensional parity-check code*

ما يقدر اكتشفا لانه رح
ما يلغوا تأثير بعض.



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.

. Parity طريقة ثانية غير ال

اختيارهم بناءً على طريقة معينة

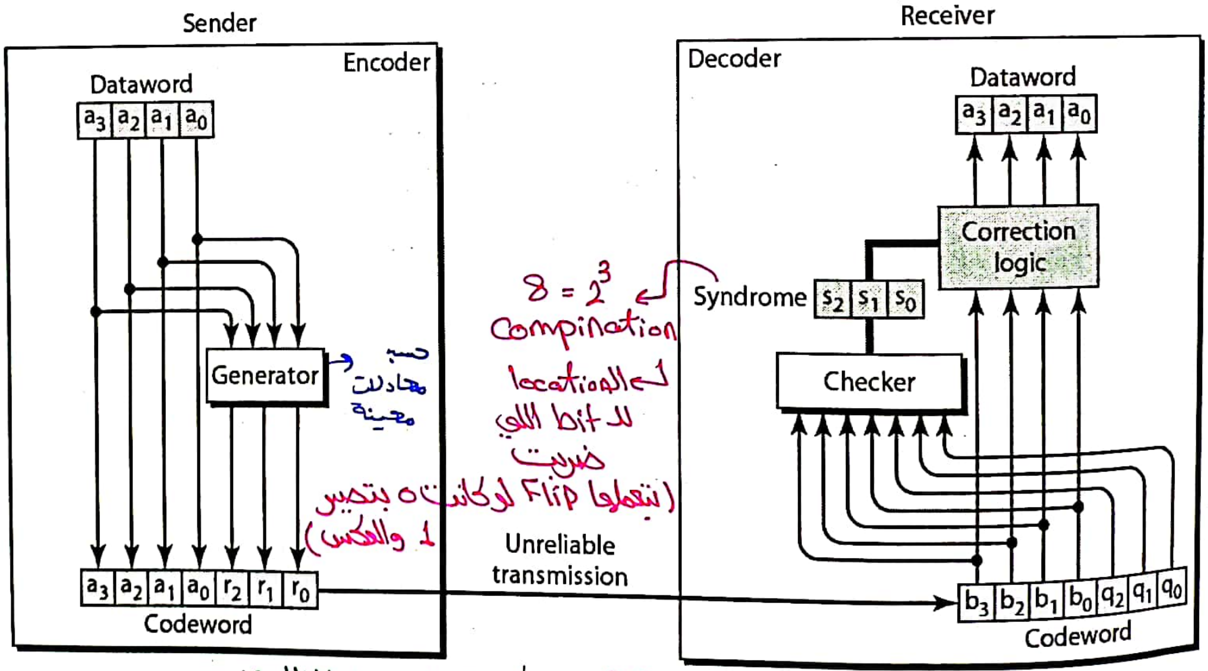
3 bits
مش بس 1 .

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*



لو b0 فيها مشكلة b1 ما بعرف وين الغلط عندي .

$r_0 = a_2 + a_1 + a_0$ 1 1 $S_0 = b_2 + b_1 + b_0 + q_0$

$r_1 = a_3 + a_2 + a_1$ 1 0 $S_1 = b_3 + b_2 + b_1 + q_1$

$r_2 = a_3 + a_1 + a_0$ 1 1 $S_2 = b_1 + b_0 + b_3 + q_2$

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

لـ(إستنتج موقع ال error وبعمله Flip لل bit)

10.53

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.* صحيح
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.* تغير داتا 2bits.

10.54

K ≥ 7 → (n−m) ≥ 7 → (2^m − 1 − m) ≥ 7
→ we try 3 → m=3 → gives 4 → 4 ≥ 7 (wrong)
↳ m=4 → gives 11 → 11 ≥ 7 (right)

Example 10.14

dataword (K)

We need a dataword of at least 7 bits. 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 ≥ 7.

2^m

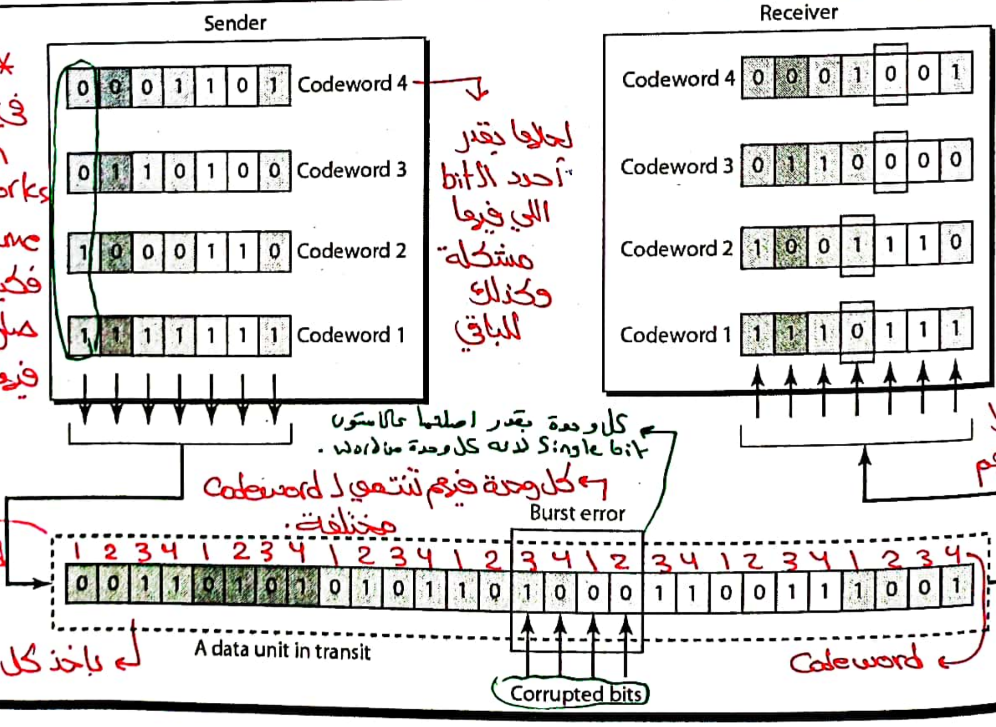
1. *If we set m = 3, the result is n = 23 – 1 and k = 7 – 3, or 4, which is not acceptable.* → أكبر من ال range بطلع
2. *If we set m = 4, then n = 24 – 1 = 15 and k = 15 – 4 = 11, which satisfies the condition. So the code is*

$$\boxed{C(15, 11)}$$

لو مجموعة من ال bits ضربت كيف ممكن تكتشف مواقعها وتصلحها

Figure 10.13 *Burst error correction using Hamming code*

* حسب ال Protocols في ال Computer and Communication networks غالبا بنبعت ككل Packet او Frame فكيف بدي اكتشف لو صار مجموعة من ال bits فيها فيرم خطأ.



لحالها بيقدر أحدد ال bit الي فيها مشكلة وكذلك للباقي

كل وحدة بيقدر اصلحها عالمستوى لانه كل وحدة من ال Single bit words.

كل وحدة فيهم تنتمي ل Codeword مختلفة

حط أكثر من Codeword داخلها ال Packet

بأخذ كل row ورا الثاني

Codeword

بتعرف مكان كل error لحال فبتصلحهم

بشبه ال Time Division Multiplexing

* بتقدر تحلها كمان لو عند 2 Burst error

* المشاكل ممكن تصير لو عندك 5 Burst error فممكن يطلعلك 2 errors بال Codeword الوحدة فممكن ينفذ هذا ال error.

أذا عملت Shift لل bits لإلي بأرادل Codewords

بيطلع Valid 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 بلف بشكل دائري كأنها cycle. *

لو عملت Shift لـ Codeword بتعطيك 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:

- Cyclic Redundancy Check — ال bits الزيادة — CRC ← نغمة — مستخدم بال wifi حاليا (IEEE 802.11)
- Hardware Implementation
- Polynomials
- Cyclic Code Analysis
- Advantages of Cyclic Codes
- Other Cyclic Codes

10.57

*حنشوف هون إنه في errors ممكن تمرق وما ينعملها 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

* ال Codeword بقسمها على ال generator (معروف بين الجهتين) لازم الباقي يكون صفر ومجهود وبعدها انتقلها هيك بتأكد لوضيحا شاكل.

$d = nq + r$

Codeword $\boxed{D.2^r \oplus R}$

10.58

$G \,\overline{)\, D.2^r}$ ... R

G: generator of divisor

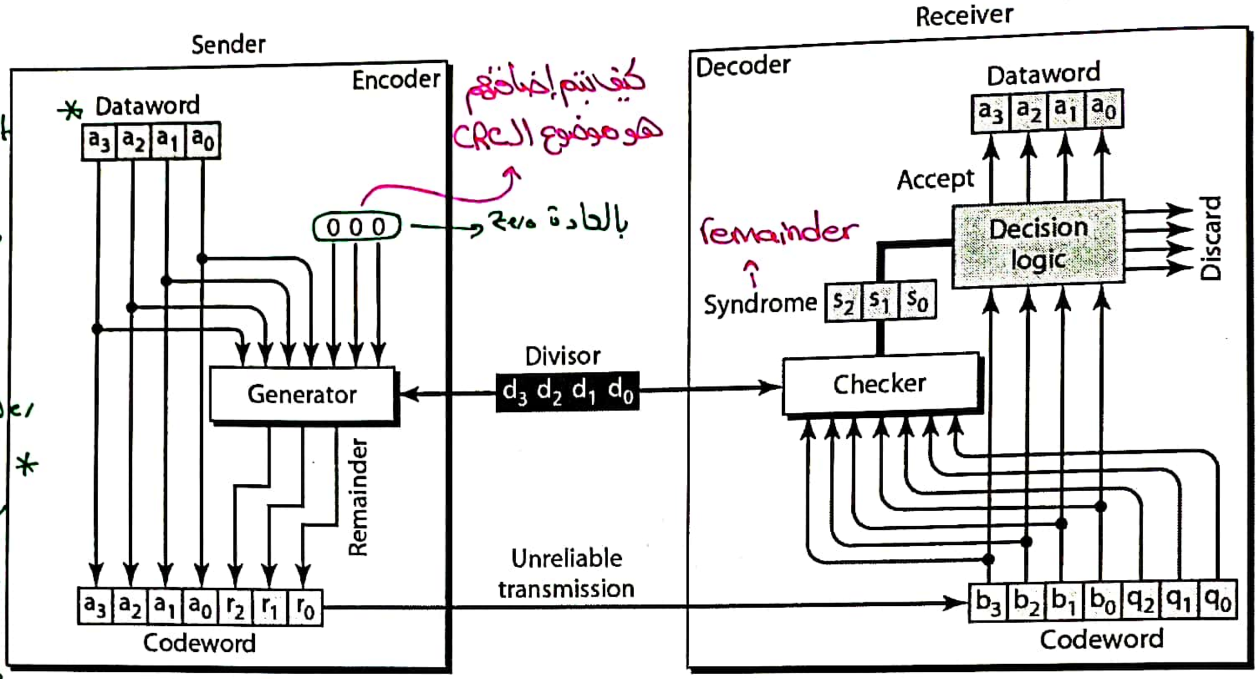
تاع ال Sender / receiver.

② — $\boxed{D^*.2^r = nG \oplus R}$ ← $\boxed{D.2^r \oplus R = n.G}$ --- ①

Figure 10.14 *CRC encoder and decoder*

Shift left بمقدار r بعد ال Shift بنعمل ال Divisor وال بطلع remainder.

* ال Checker ح يستخدم ال Divisor ذات نفسه اذا طلع ال remainder كله zeros فتمام اما غير ذلك في غلط



كيف يتم إضافتهم هو موضوع ال CRC

بالعادة zero

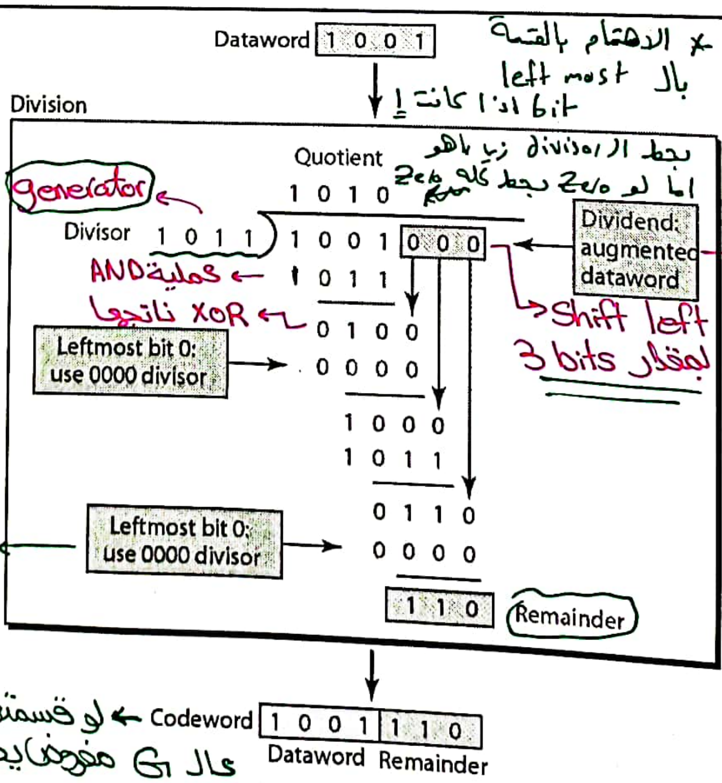
remainder

10.59 * بالعادة بنعمل لل data word إزاحة بمقدار معين (Shift left) لنترك مساحة لل remainder bits بدل ما نعمل عملية XOR . (طريقة إضافة ال remainder bits)

Figure 10.15 *Division in CRC encoder*

* الاهتمام بالقسمة بال left most bit اذا كانت 1

* تبدأ عملية القسمة من ال Most sig. bit (لو كانت 1 بتحط 1 ولو 0 بتحط 0) وبعد كل عملية بطنش ال MSb وبزحف لليمين



بحط ال divisor زي ما هو اما لو zero بحط كله zero

generator

AND عملية

XOR ناتجها

Shift left بمقدار 3 bits

$D.2^r$

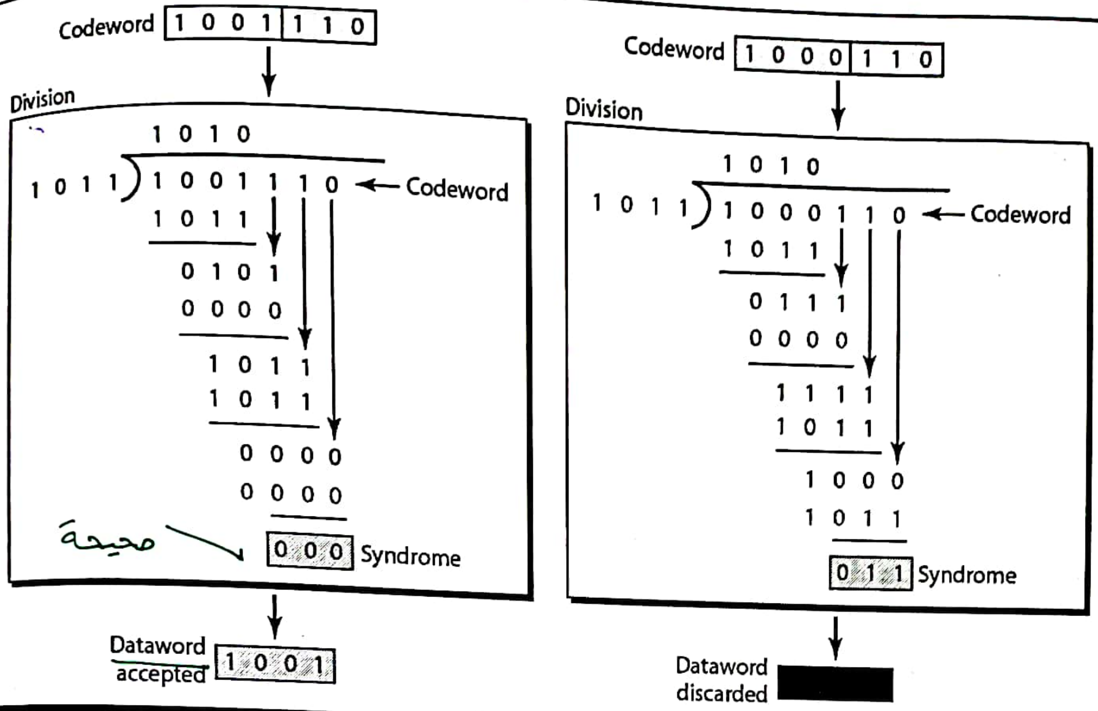
فعليا عملية القسمة بلشت عند T=4 فلهيك احنا نعمل Shift و XOR

لو قسمتها ← Codeword

لل G مفروض يطلع الناتج = صفر ما عدا ذلك أخطاء بال bits

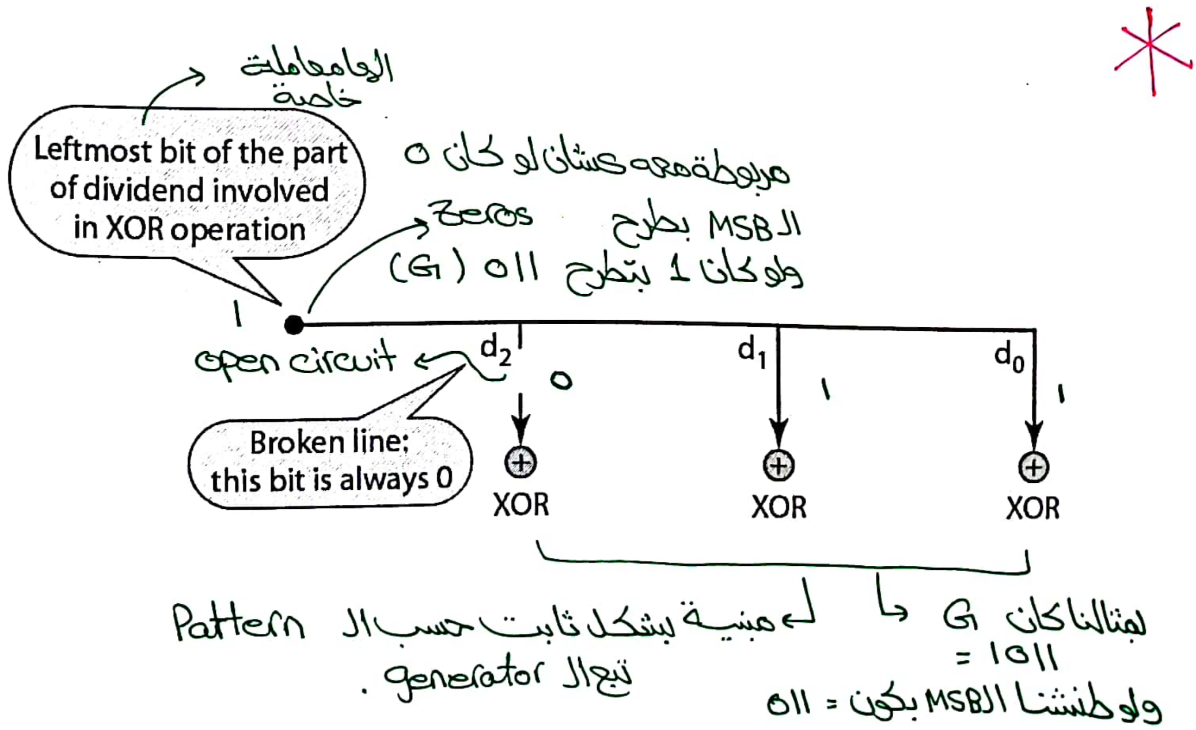
10.60

Figure 10.16 *Division in the CRC decoder for two cases*



10.61

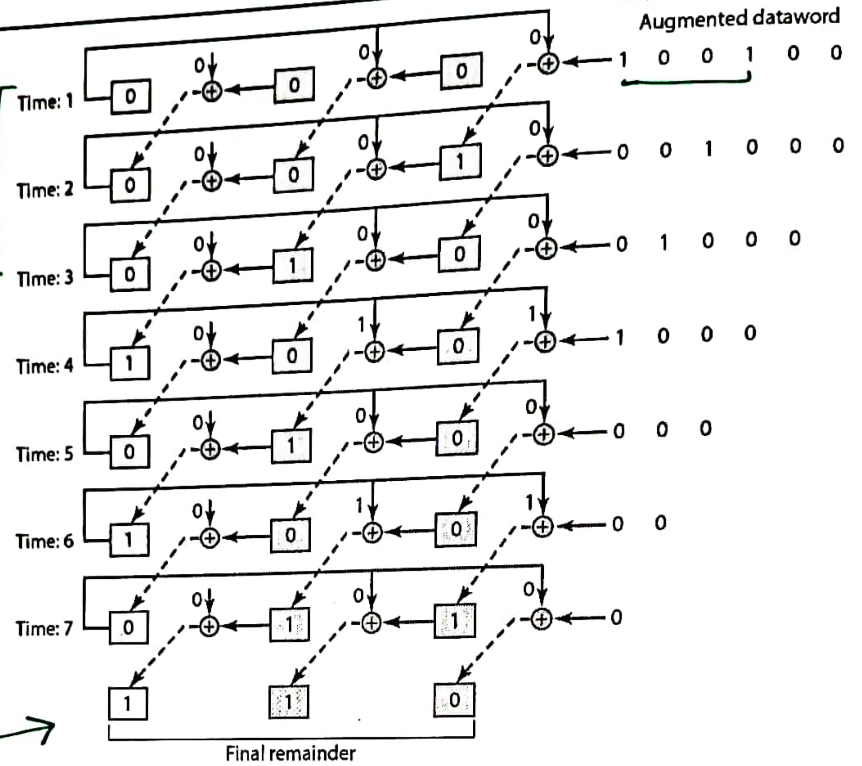
Figure 10.17 *Hardwired design of the divisor in CRC*



10.62

*كلماكان ال generator فيه 0's أكثر كلماكانت تكلفته أقل، وانت بتختاره حسب نوع ال error الي بدك تعمله detect، ولازم نفس ال generator المستخدم عند ال Sender يكون مستخدم عند ال reciever.

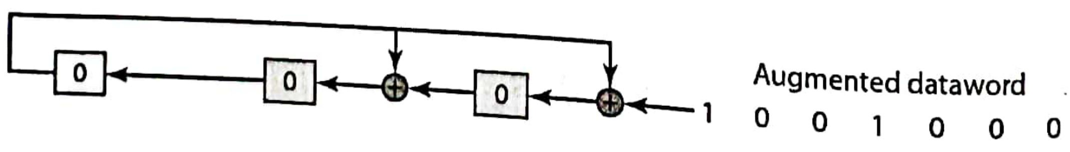
Figure 10.18 *Simulation of division in CRC encoder*



10.63

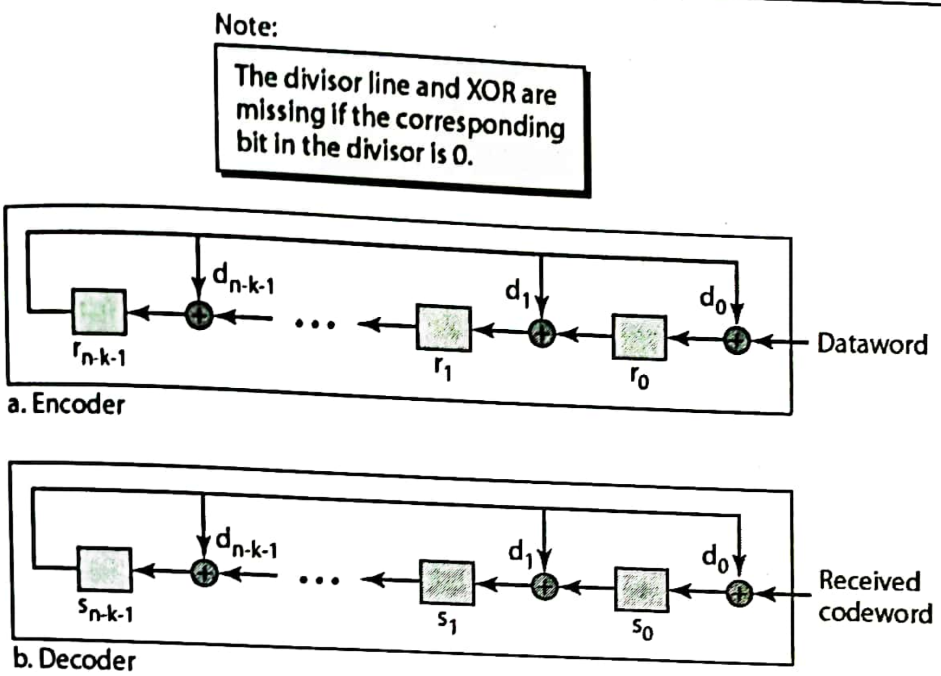
* مو ضروري احتفظ بجميع ال registers السابقة بيكفي بس آخر وحدة فبختصر عدد ال register

Figure 10.19 *The CRC encoder design using shift registers*



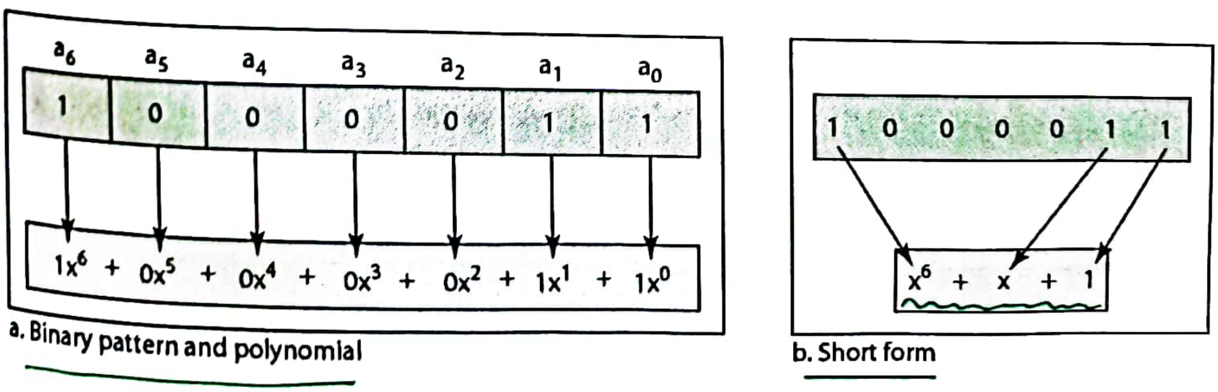
10.64

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

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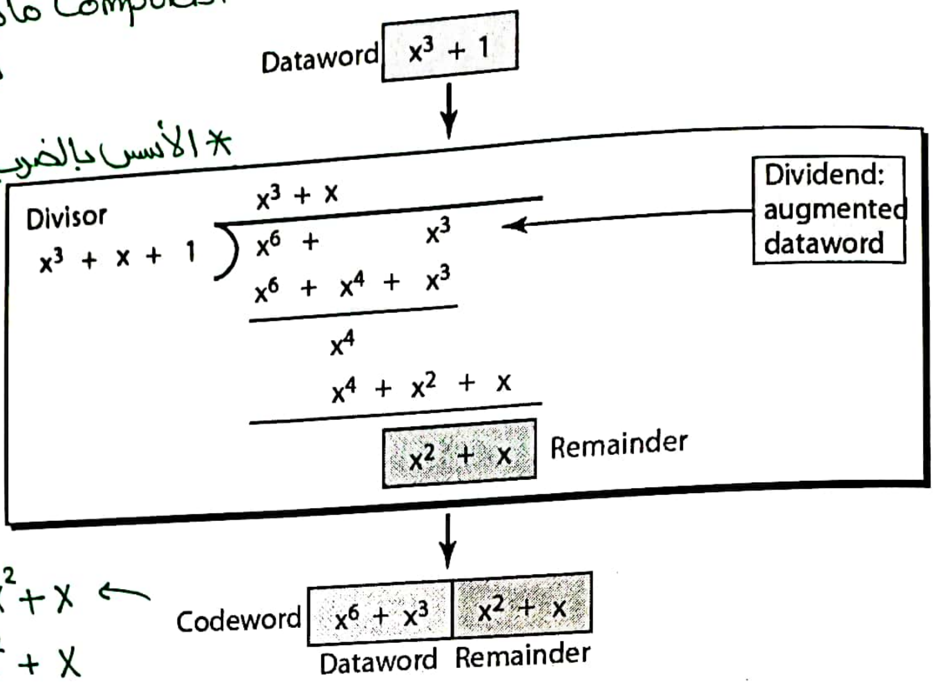
Figure 10.22 *CRC division using polynomials*

ال Computer مابقدر يعمل هيك بعملها بال binary زي قبل .

* الأسس بالضرب تجمع .



$x^3(x^3+1)+x^2+x$

$= x^6+x^3+x^2+x$

10.67

ما يتم اختياره عشوائي ← بناء على تحليلات

بنختاره بذكاء ليلقط عدد معين من ال error رياضية معينة

Note

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.

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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 طلعتنا ١٠٠٪، المهم يكون at least two terms.

T = 1000101
T' 1100101
$E[x]$ = 0100100 $x^6 + x^3 + x^2$

بدي لما اقسم $E[x]$ على $g[x]$ يكون دايما في باقي
وهاد الحكي لما يكون الgenerator فيه على الاقل 2 non zero terms. وح يكون detected (100%)

10.71

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

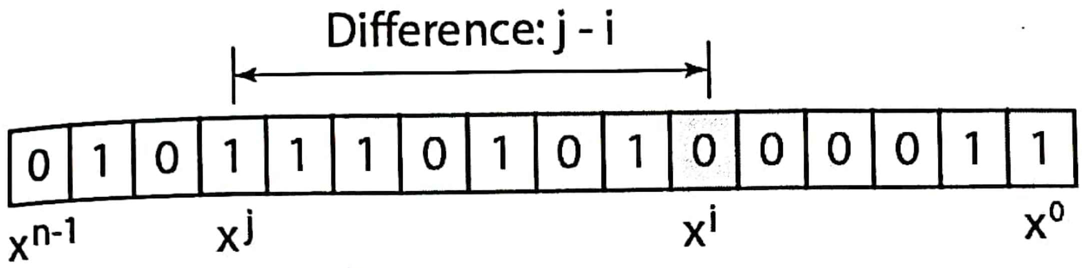
a. No x^i can be divisible by $x + 1$. Any single-bit error can be caught.

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.

10.72

Figure 10.23 *Representation of two isolated single-bit errors using polynomials*



$x^j + x^i$

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.

اذا كان واحد double

بدي ال g(x) ← ما يكون x^i فبدها

على الأقل 3 terms

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.**

عدد ال errors ← فردي فبدي X+1

Note

عاليمين

- All burst errors with $L \leq r$ will be detected. 100%.

بالنص

- All burst errors with $L = r + 1$ will be detected with probability $1 - (1/2)^{r-1}$.

عالشمال

- 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

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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 $x + 1$.**

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Table 10.7 ***Standard polynomials*** مستخدمة بال CRC

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 ويمكن نستخدمه بال Correction

* في تشابه بينه وبين ال TCP.

* في طريقتين لتعديل الأخطاء بال data communication وال Computer Networks

← الأولى إذا اكتشفت الخطأ ووين ال bit موجودة بتعمل Correct لل bit (forward error correction).

← الثانية إذا اكتشفت الخطأ ابعث للمرسل طلب بإعادة الإرسال (Automated repeat request).

10.81

10-5 CHECKSUM

→ بتشابه مع ال CRC بالبساطة.

فكرته انه اول اشي بدي اقسم ال data ل words وبعدين اجمعهم وبأخذ ال 1's Complement بقلب ال 1 ← 0 و 0 ← 1 وال Receiver بنفس الشي هو ال sum وال Complement اذا كله 1 check الا 0 تمام هل

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

4 words → 8 bits

10011001	11100010	00100100	10000100

Sender

10011001
11100010
(1) 01111011
1
01111100
00100100
(0) 10100000
10000100
(1) 00100100
1
Sum 00100101
Checksum 11011010

Receiver

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(1) 00100100
1
00100101
Checksum 11011010
11111111
Complement 00000000
Accept

Topics discussed in this section:

Idea
One's Complement
Internet Checksum

10.82

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.

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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

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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.

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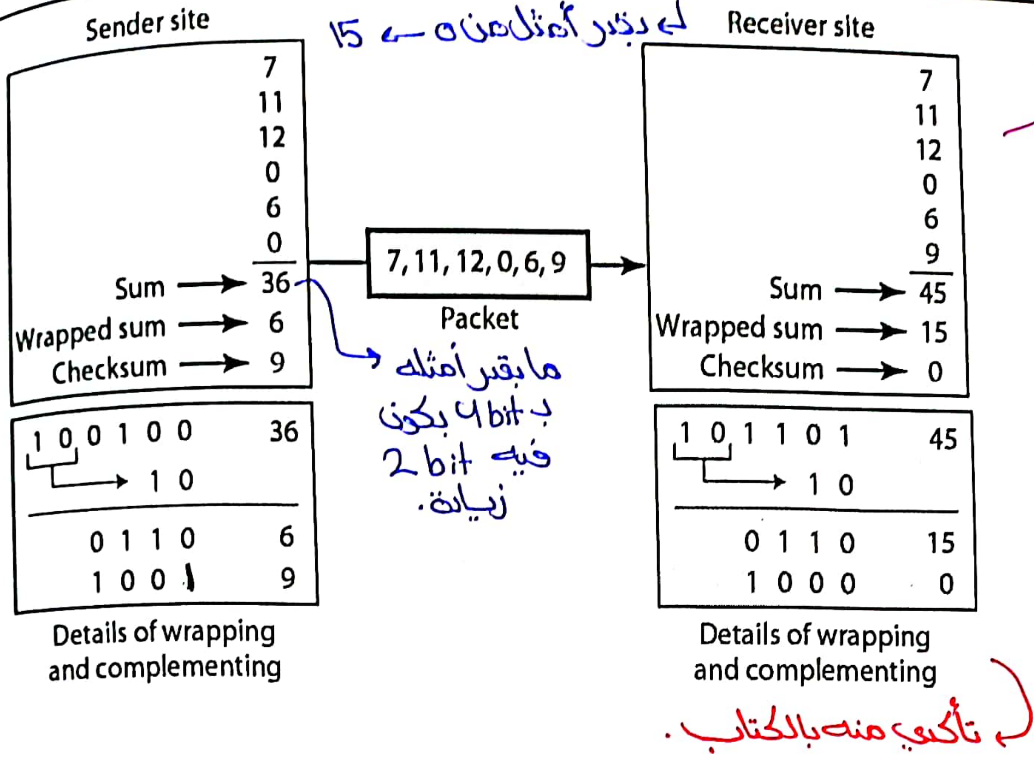
Figure 10.24 *Example 10.22*

4 bit arithmetic



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.

جمع كل شي بما فيها الChecksum ← * 2. All words are added using one's complement addition.

* 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.

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.

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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 »