Digital Data Acquisition

الحصول على البيانات الرقمية

What is "Digital Data Acquisition"?

It is the process we use to change signals from our real world (like your voice or a video image) into digital data (zeros and ones) that a computer can understand, store, and process.

Simply: This is the "bridge" that connects the Analog world and the Digital world.
(ببساطة: هذا هو "الجسر" الذي يربط بين العالم التناظري والعالم الرقمي)

1. Analog and Digital Signals

This is the key to understanding everything else. The world around us is "analog" by nature, while a computer is "digital".

Analog Signal (إشارة تناظرية)

It is a **"continuous" (مستمرة)** signal. Think of it as a smooth sound wave or the smooth colors in a sunset. There are no "jumps," just a continuous flow of values.

Digital Signal (إشارة رقمية)

It is a **"discrete" (متقطعة)** signal. The computer doesn't understand waves; it understands specific numbers (0 and 1). A digital signal is a representation of this wave as "steps" or "bars" with specific values.

Why do we prefer Digital? (لماذا نفضل التحويل الرقمي؟)

- **Quality & Stability:** A digital file does not lose quality over time or with many copies, unlike analog.

- **Compression:** We can compress digital data to save a lot of space (like `MP3` and `JPEG` files).
- **Storage:** It's easy to store sound, images, and text (all digital) on the same device (like a hard disk).
- **Interactivity (التفاعلية):** It gives us great flexibility to edit, change, and montage.

2. The Conversion Process (Analog-to-Digital) عملية التحويل

This is the magic process! How do we turn a wave (Analog) into "steps" (Digital)? We use a special device called an `Analog-to-Digital Converter (ADC)`. This device is the heart of a sound card, a digital camera, and all modern input devices.

The process has two main steps:

1. Sampling (أخذ العينات)
2. Quantization (التكميم)

Note :

The reverse process, when playing digital audio on speakers, uses a `Digital-to-Analog Converter (DAC)`, which reconstructs the signal by performing interpolation to return the sampled data into a smooth analog waveform.

3. Sampling (أخذ العينات)

Simply, "sampling" is deciding **"when"** to take a snapshot of the wave. We measure the analog signal's value at regular time intervals.

- **Sampling Rate (f):** This is the number of snapshots (samples) we take in one second. It's measured in "Hertz" (Hz).
- **Example:** A `44.1 KHz` rate in audio files means we take 44,100 snapshots of the sound wave every second.

Nyquist Theorem (نظرية نايكويست)

This is the golden rule that decides "how many" samples we need. The question is: do we take too few samples and risk losing details, or take too many and fill up the hard disk?

> **"To correctly rebuild an analog signal from its digital samples, the `(Sampling Rate)` must be at least twice (ضعف) the highest `(maximum frequency)` in the original signal."**

The Disaster: Aliasing (التشويه)

What happens if we ignore the "Nyquist Theorem" and sample at a very slow rate (less than twice)?
The answer: **Aliasing**.

Aliasing is when the few samples you took create a new "fake wave" with a frequency that is different from the original wave. You are tricking the system into seeing something that isn't there.

These two types are just for looking

Temporal Aliasing (التشويه الزمني)

تخيل إنك بتصور مروحة شغالة بكاميرا بمعدل فريمات قليل. هتلاحظ إن شفرات المروحة أحيانًا بتظهر واقفة أو بتتحرك بالعكس!

Low frame rate in video

Spatial Aliasing (التشويه المكاني)

تخيل إنك بتصوّر صورة بكمية قليلة جدًا من البكسلات. لما تقرب على الصورة، هتبدأ تشوف خطوط غريبة أو مربعات بدل التفاصيل الدقيقة.

Low resolution

4. Quantization (التكميم)

If "sampling" decides "when" to take the snapshot (on the time X-axis), then "quantization" decides the **"precision" (دقة)** of that snapshot (on the strength Y-axis).

We are "rounding" the snapshot's actual value to the nearest predefined digital "Level".

Bit Depth (عمق البت)

This decides the number of available levels.
The rule: `N (Levels) = 2^b (bits)`

- **1 bit:** Gives us 2^1 = 2 levels only (e.g., black / white).
- **8 bits:** Gives us 2^8 = 256 levels (enough for grayscale images and "speech" audio).
- **16 bits:** Gives us 2^16 = 65,536 levels (necessary for "music" to capture tiny differences).

Conclusion: The higher the "Bit Depth," the more "precise" the digital signal is (and closer to the original), but this also increases the "Storage" size.

Practical Example (from the lecture):

We have a sequence: `[550, 600, -100, 150, -300, 900, 0 and 850]` and we want to convert it using `2 bits`.

-300	-->	0	(00)
0	-->	300	(01)
300	-->	600	(10)
600	-->	900	(11)

1. **Number of Levels:** 2^2 = 4 levels. = L
2. **Range:** From -300 to 900 (Difference = 1200). = R
3. **Step (Level size):** 1200 / 4 = 300. = δ
4. **Result:** Any value between 300 and 600 (like 550) will be rounded and assigned the value `(10)`.

Note : Analog signal graph
- Constant line -> "One sample is enough"
- Linear line -> "Two samples are enough"
- Complex wave -> "Many samples are required"

5. Bit Rate (معدل البت)

The "Bit Rate" is the final result that combines the two previous processes. It represents the "total number of bits" that we produce (or use) every second.

Bit Rate = (Sampling Rate) x (Quantization bits per sample)

(معدل البت = معدل العينات * عمق البت)

SIGNAL	SAMPLING RATE	QUANTIZATION	BIT RATE
Speech (كلام)	8 KHz	8 bits	64 kbps
AM Radio	11 KHz	8 bits	88 kbps

6. Filtering (الفلترة)

Once the signal is "digital", we can process it easily using "Digital Filters". A filter simply "allows" certain frequencies to pass and "blocks" other frequencies.

The big advantage of Digital Filters: They are just "software". This makes them flexible, precise, programmable, and not affected by heat or age like analog filters (which are hardware).

Low-pass filter (فلتر تمرير منخفض)

It allows "low" (slow) frequencies to pass and blocks "high" (sharp) ones.
In images: It removes details and "noise", and the result is a "soft" or "blurred" image.

makes the image blurred / soft

High-pass filter (فلتر تمرير عالي)

It allows "high" frequencies (details and edges) to pass and blocks "low" ones.
In images: It finds the "Edges" and removes the base colors, giving an effect similar to a sketch.

makes the image sharp / highlights edges

Band-pass filter (فلتر تمرير حزمة)

It allows a specific "band" of frequencies to pass (not too high and not too low).

keeps a specific range of details