Audio

sound is an analog signal, which is made up of waves travelling through matter. The computer needs to convert this signal into a digital format to be machine readable.

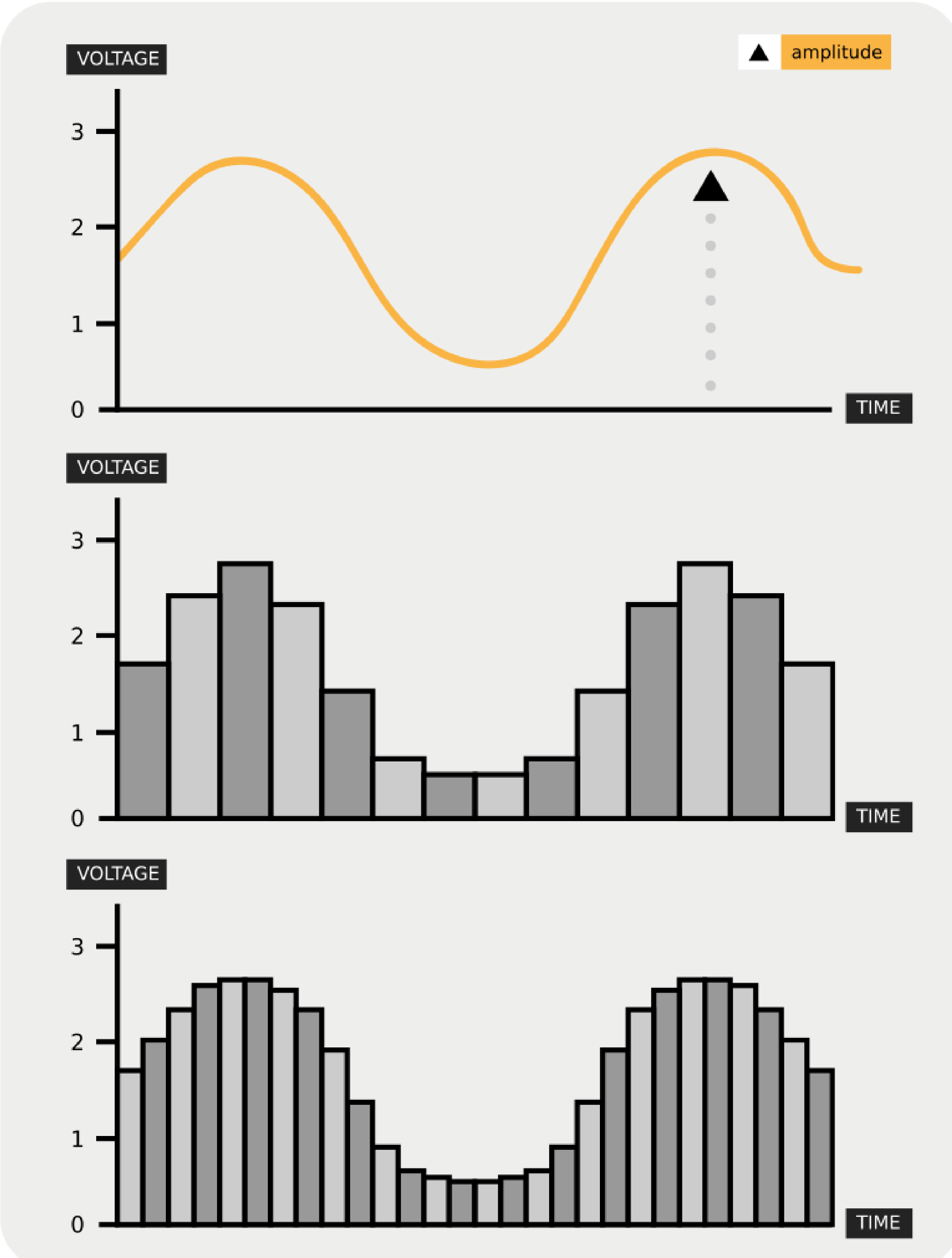
The sound card on your computer performs this process for you:

- The sounds card has **four** main processing components to perform this task.
    - **analog-to-digital converter (ADC)** - to process the incoming analog signal into a digital format, e.g. sound recorded by your microphone.
    - **digital-to-analog converter (DAC)** - converts the digital audio signal on your pc to an analog format so you can listen to it, e.g. converted to analog and output through the speakers.
    - **PCI interface** - to connect the sound card to the motherboard.
    - **Input and output** - for devices such as a microphone or speakers.

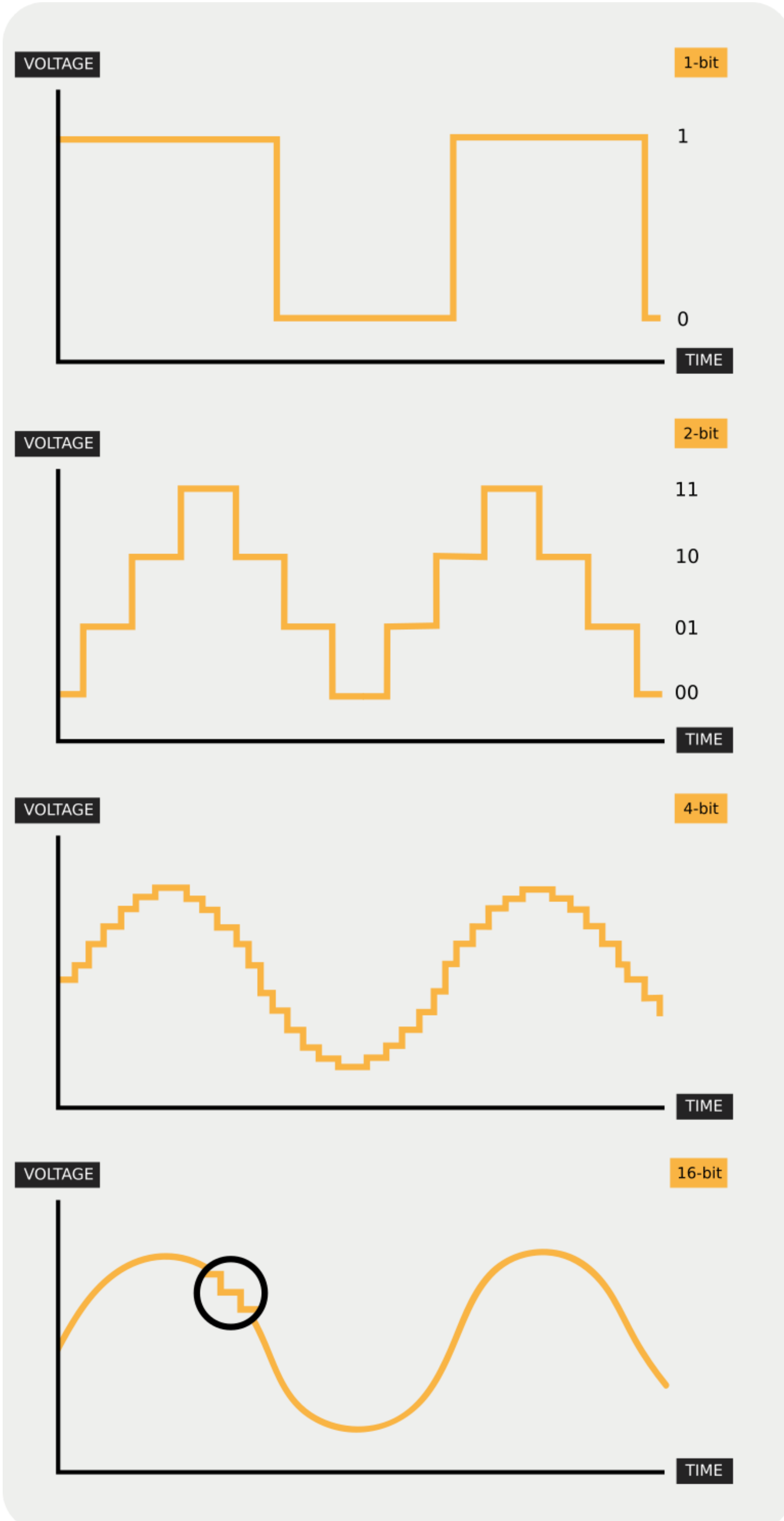
Sound cards represent audio with the following process:

- When an incoming audio is detected by the sound card it will take measurements (samples) of it at regular intervals.
    - **Sampling rate** is defined as the number of samples taken per second of the sound card.
    - Sampling rate is measured in hertz (one hertz is one sample per second).
    - The higher the hertz, the better quality of the sound representation.
- **Sampling resolution** is the numbers of bits used to represent the audio.
    - The higher the resolution, the better the representation of the sound.

True sound (yellow), then visualisations of low and higher sampling rate:


VOLTAGE
amplitude
3
2
1
0
TIME
VOLTAGE
3
2
1
0
TIME
VOLTAGE
3
2
1
0
TIME

Increasing sampling resolution:


VOLTAGE
1-bit
1
0
TIME
VOLTAGE
2-bit
11
10
01
00
TIME
VOLTAGE
4-bit
TIME
VOLTAGE
16-bit
TIME

Commonly, audio data is visualised as a spectrogram. A spectrogram shows time on the horizontal axis, and frequency on the vertical axis with brighter colors where that frequency is present.