



# 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.

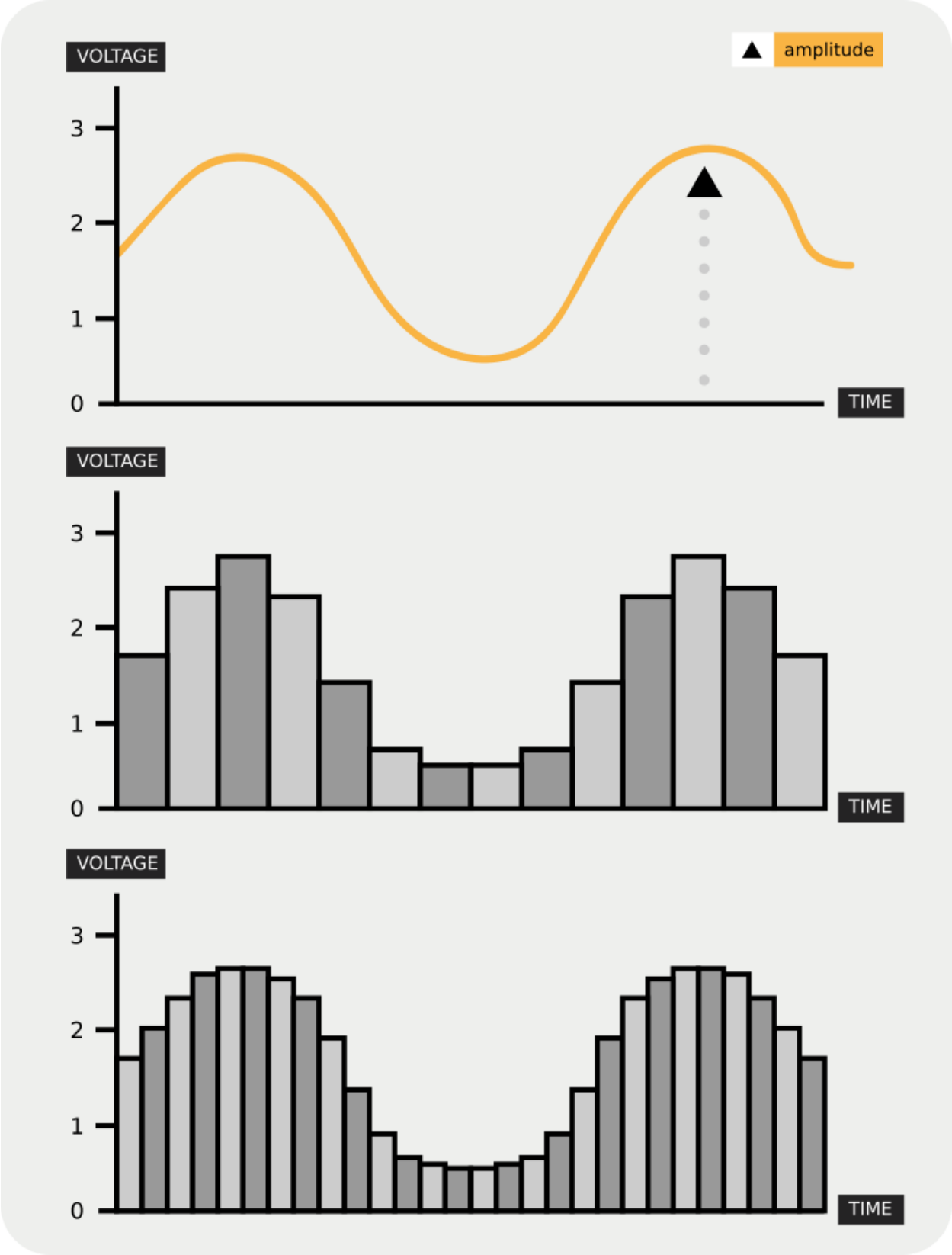
The sound card on your computer performs this process for you:

- The sound 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.

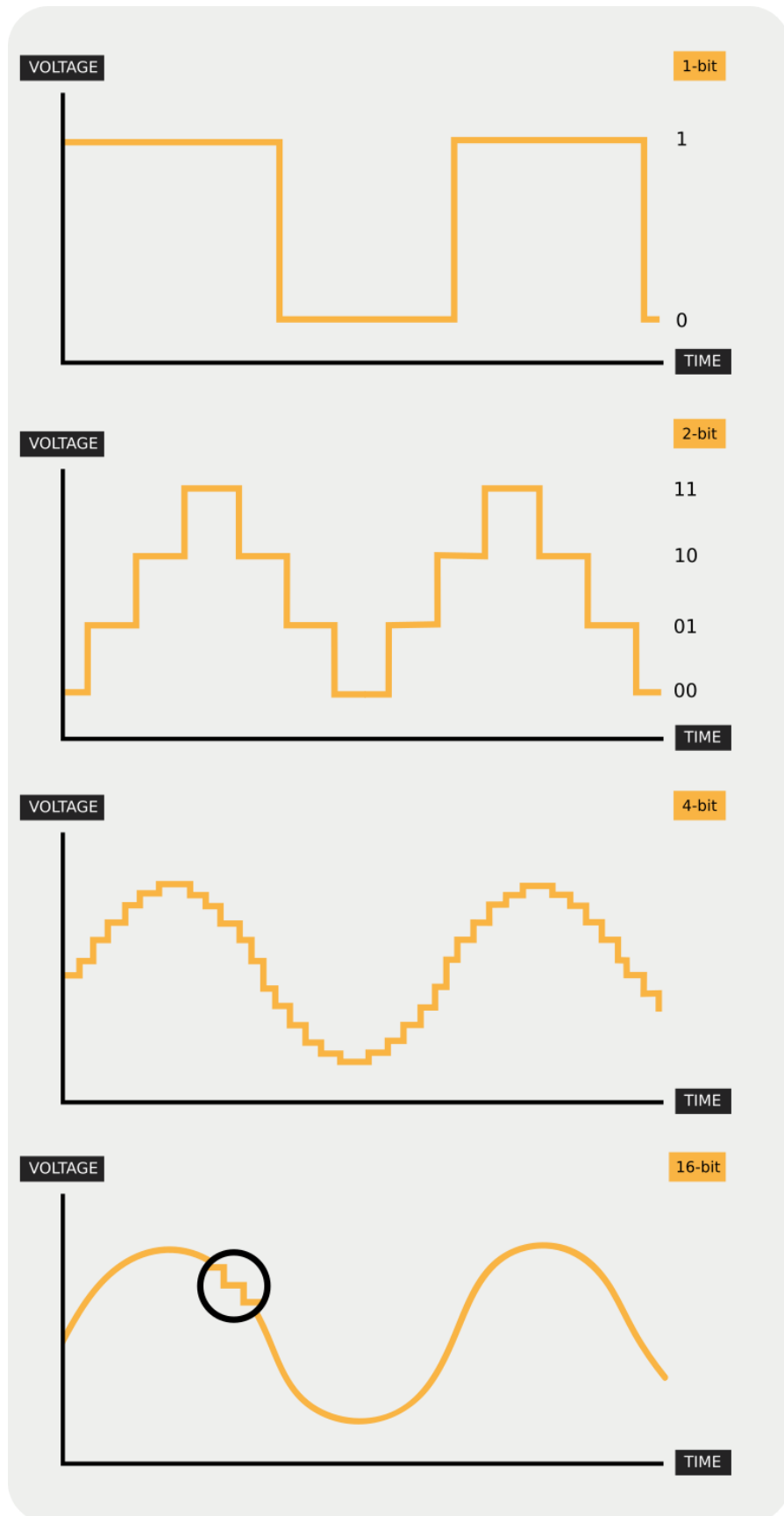
Sound cards represent audio with the following process:

- 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:



Increasing sampling resolution:



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.

