Theoretical Backgrounds of Audio & Graphics

Digitization

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Digital Audio

- In order to understand what digital audio data is, we will start with answering the following two question:
  - How is analog sound/audio turned into digital sound/audio?
    - This is achieved by a process referred to as **digitization**
  - How is the signal information stored and processed in a computer?
    - This is usually accomplished by an **audio buffer**

Digitization

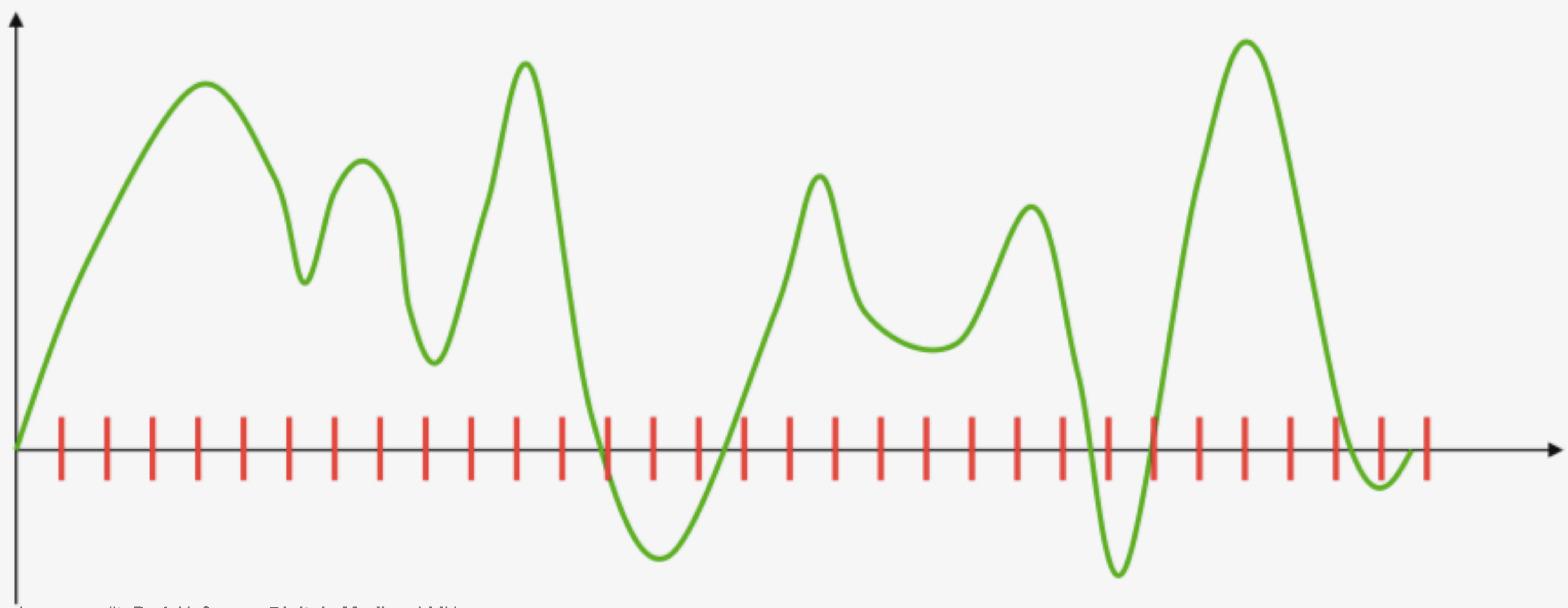
- Digitization consists of two steps: **sampling** & **quantization**
- Most common method: Pulse Code Modulation (PCM)

- **Sampling:**
  - The amplitude of the analog signal is measured (in volts) at fixed time intervals determined by the **sampling rate**

- **Quantization:**
  - The sampled amplitude values are mapped onto discrete values defined by the **bit depth** or **sample size**

Sampling

- Sampling is also called **discretization:**
- The **continuous** audio signal is converted into **discrete** audio samples



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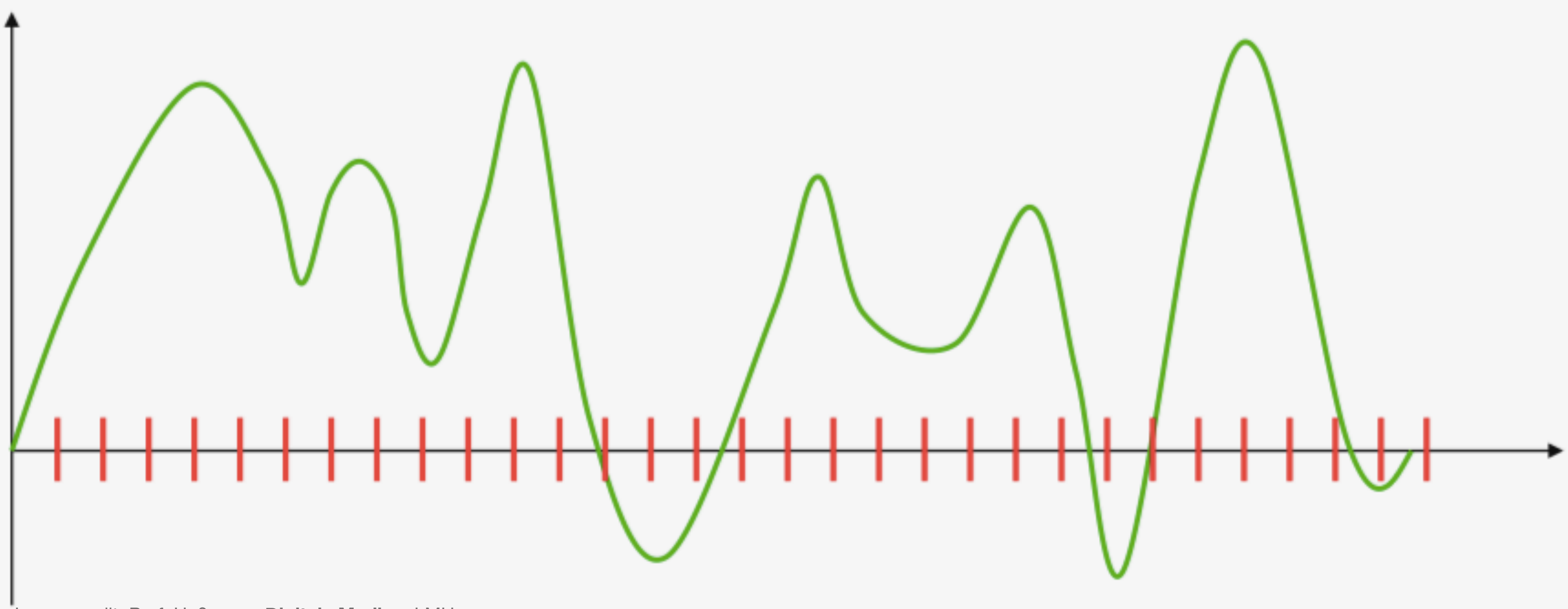
Sampling

- Therefore the amplitude of the analog signal is measured at fixed time intervals — the **sampling rate** or sampling frequency
- The **sampling rate** determines how many samples are taken per second



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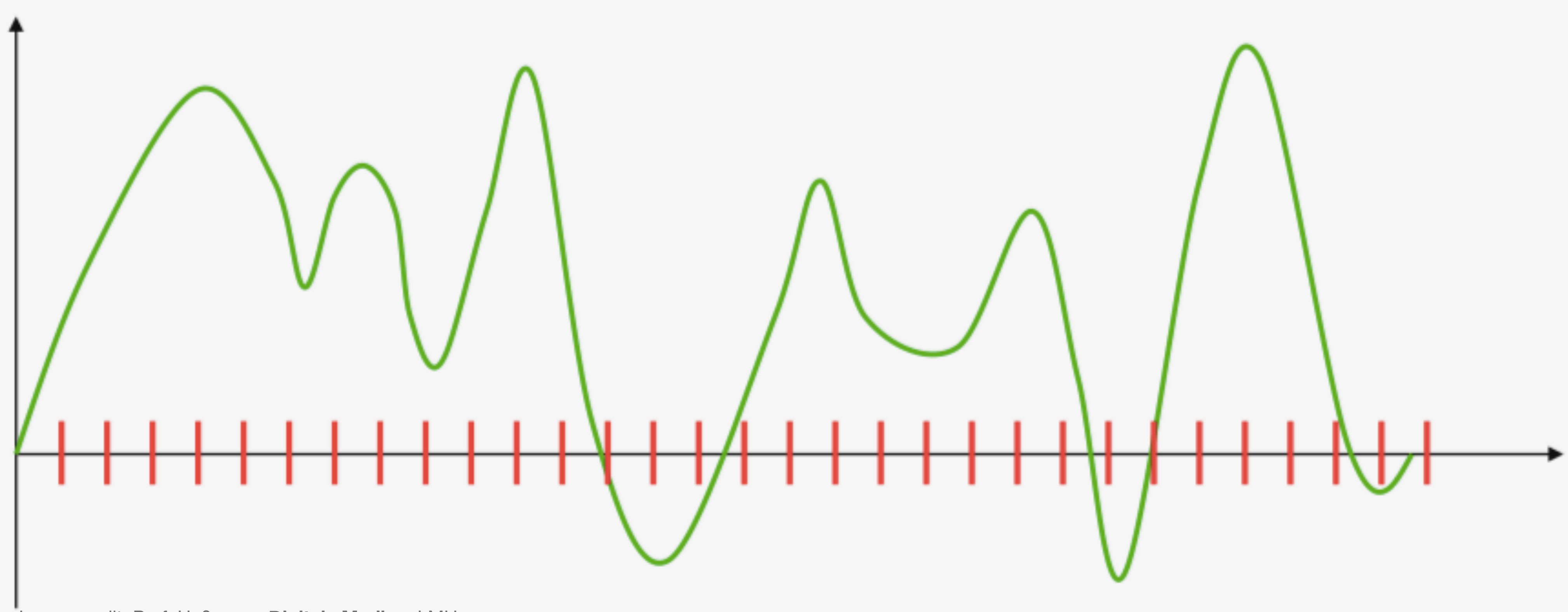
Sampling

- Sampling rate must be **greater than twice** the highest frequency of the original signal for proper reconstruction (Nyquist-Shannon sampling theorem) — f_sampled > 2 * f_max



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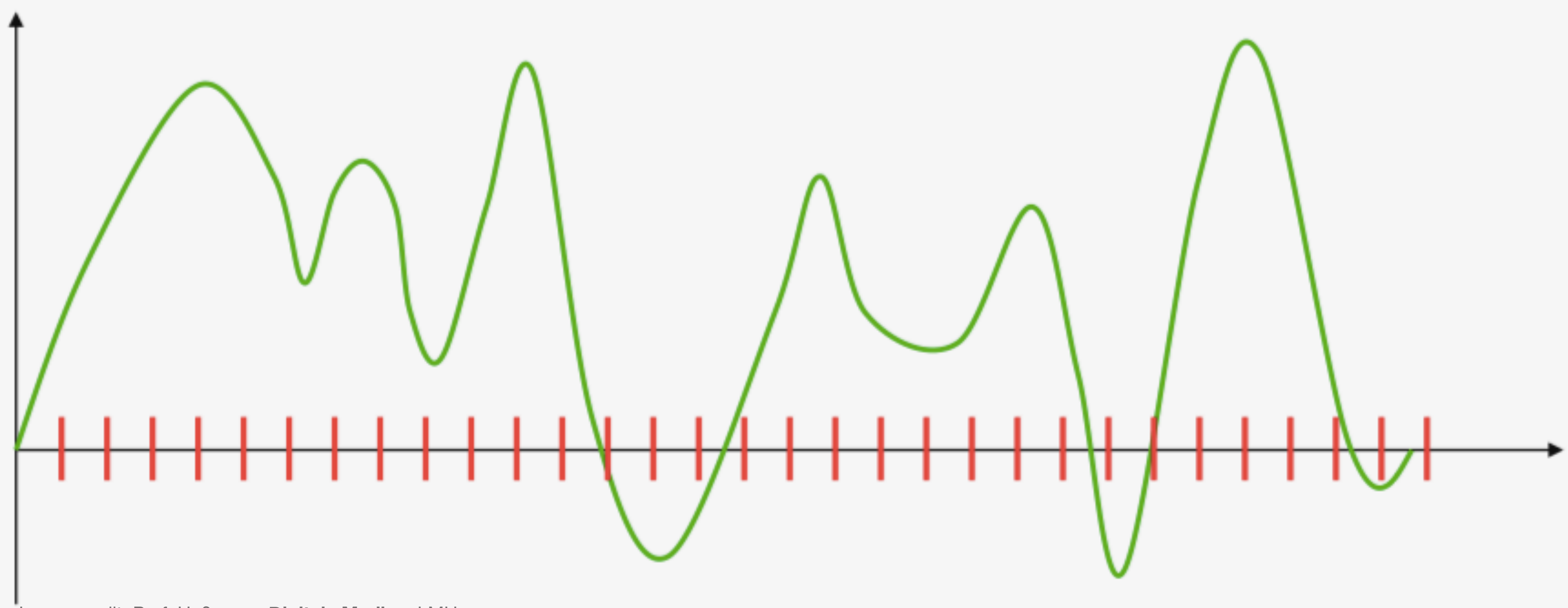
Sampling

- Standard sampling rate for CD quality is 44.1 kHz or 44100 Hz due to the fact that the human hearing range is at 20 kHz maximum
- Hence, the sampling rate for CD quality must be greater than 2 x 20 kHz



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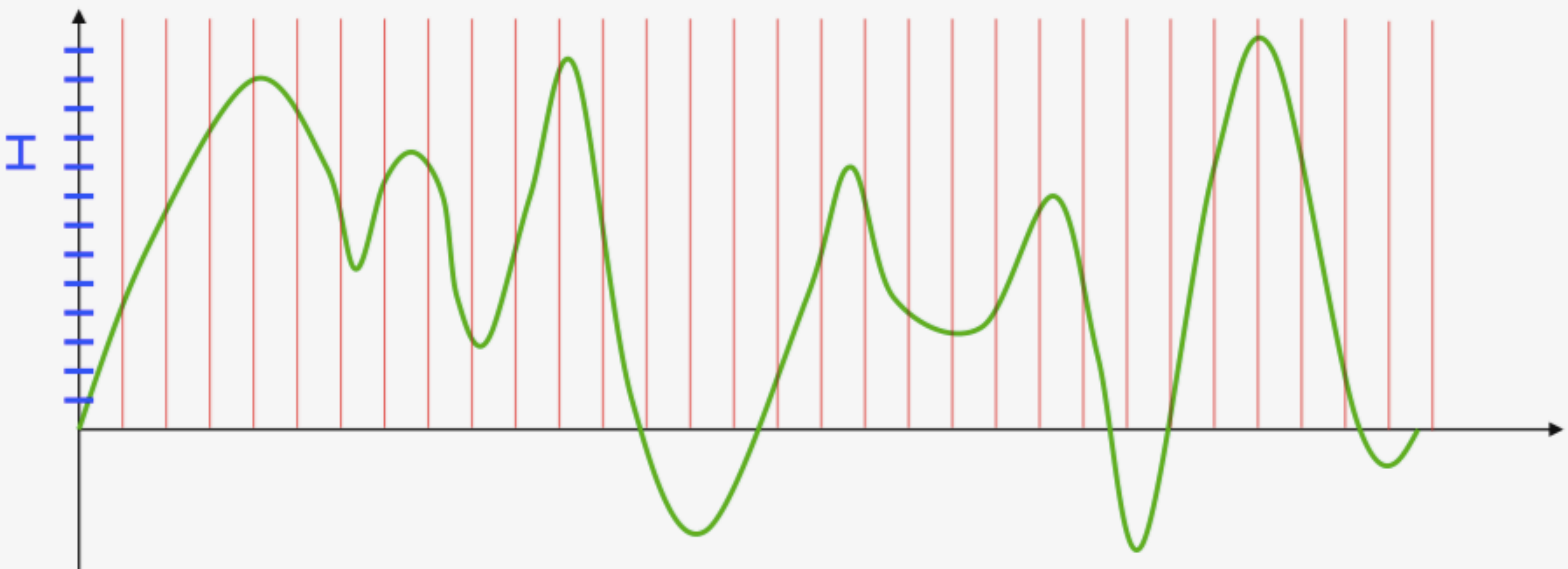
Quantization

- During quantization, the sampled amplitude values are quantized, i.e., mapped onto discrete values defined by the **bit depth** or **sample size**



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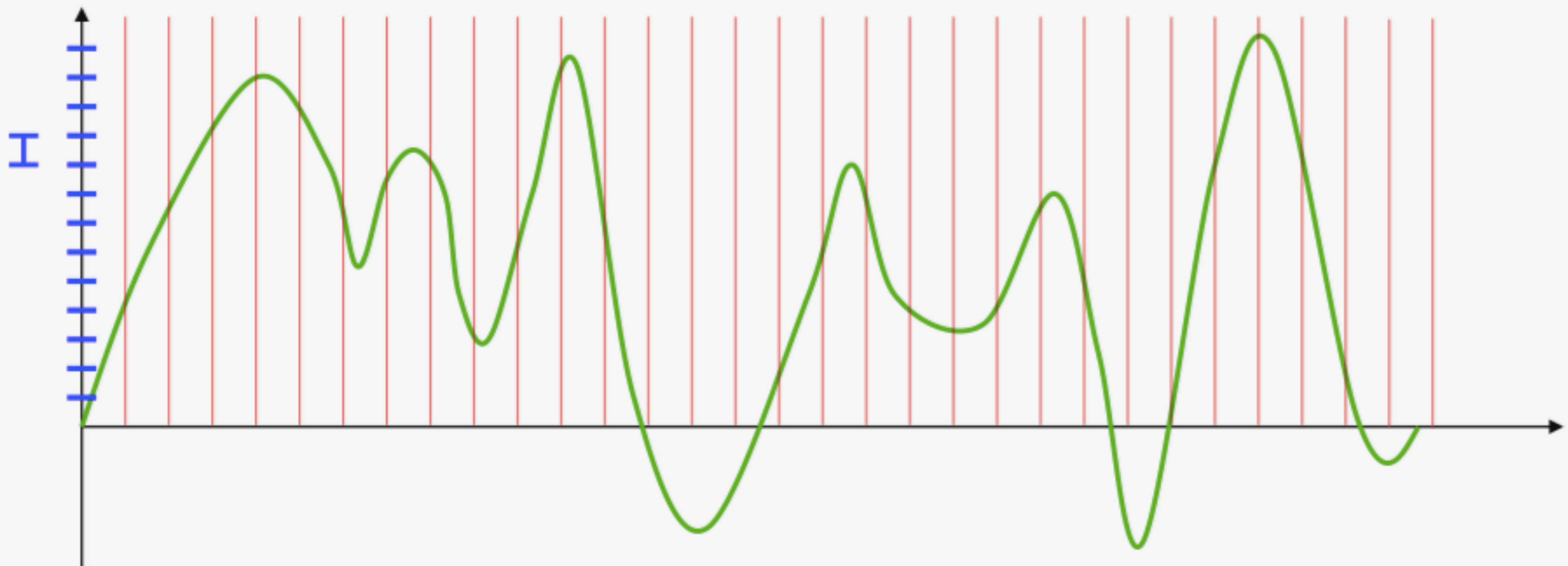
Quantization

- The sample size determines the difference between the softest sound and the loudest sound, i.e., the **dynamic range** of the audio application
- Numbers are usually stored as integers or floating points at, e.g., 24bit



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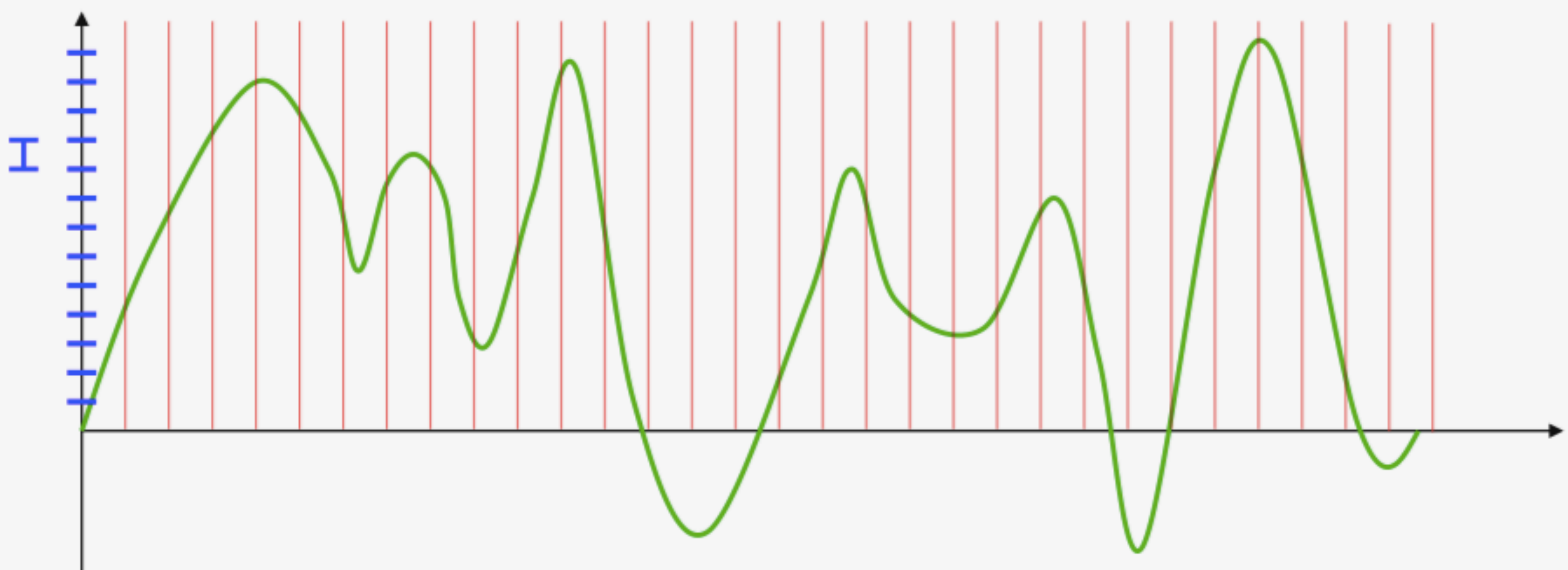
Quantization

- A sample size of 24 bits provides for 16 777 216 amplitude values
- Turned into decibel, a dynamic range of 144 dB can be represented
  - 144 dB = 20 $\log_{10}$(16 777 216)



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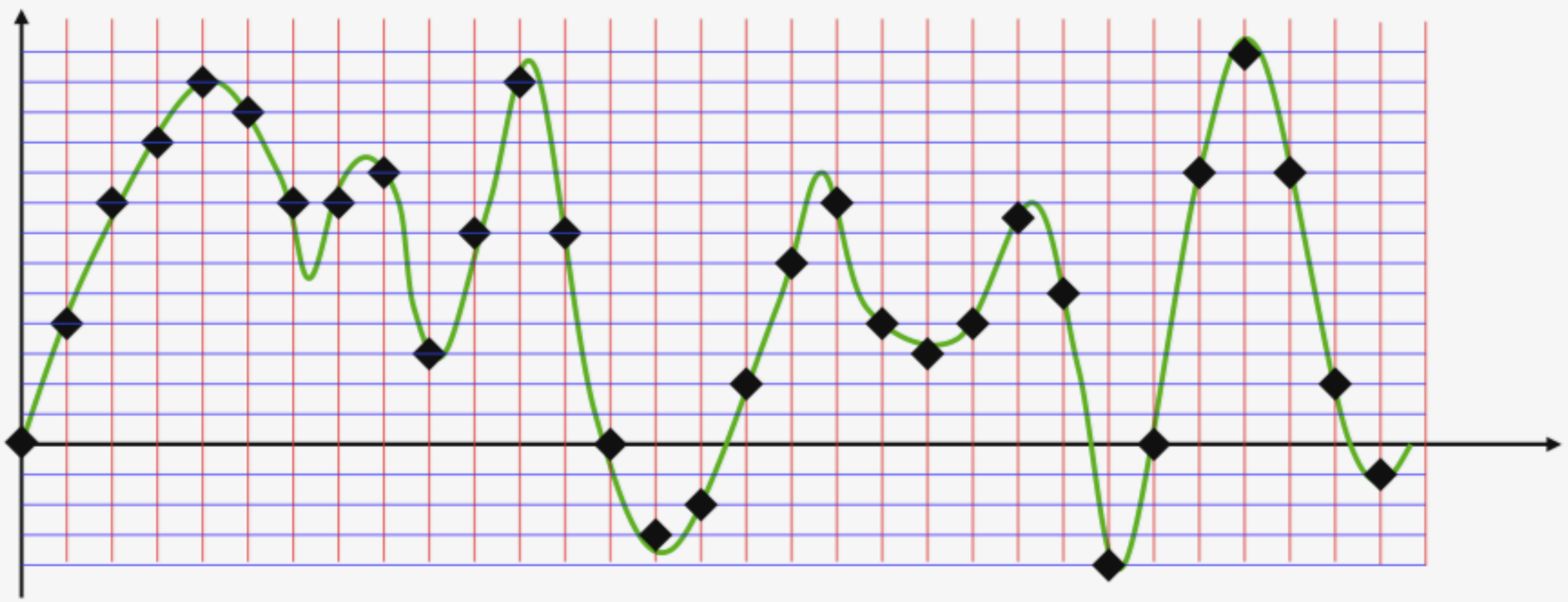
Digitization

- Generally, signal **quality** is controlled by sampling rate and bit depth
- Sampling & quantization always introduce a certain **digitization error**



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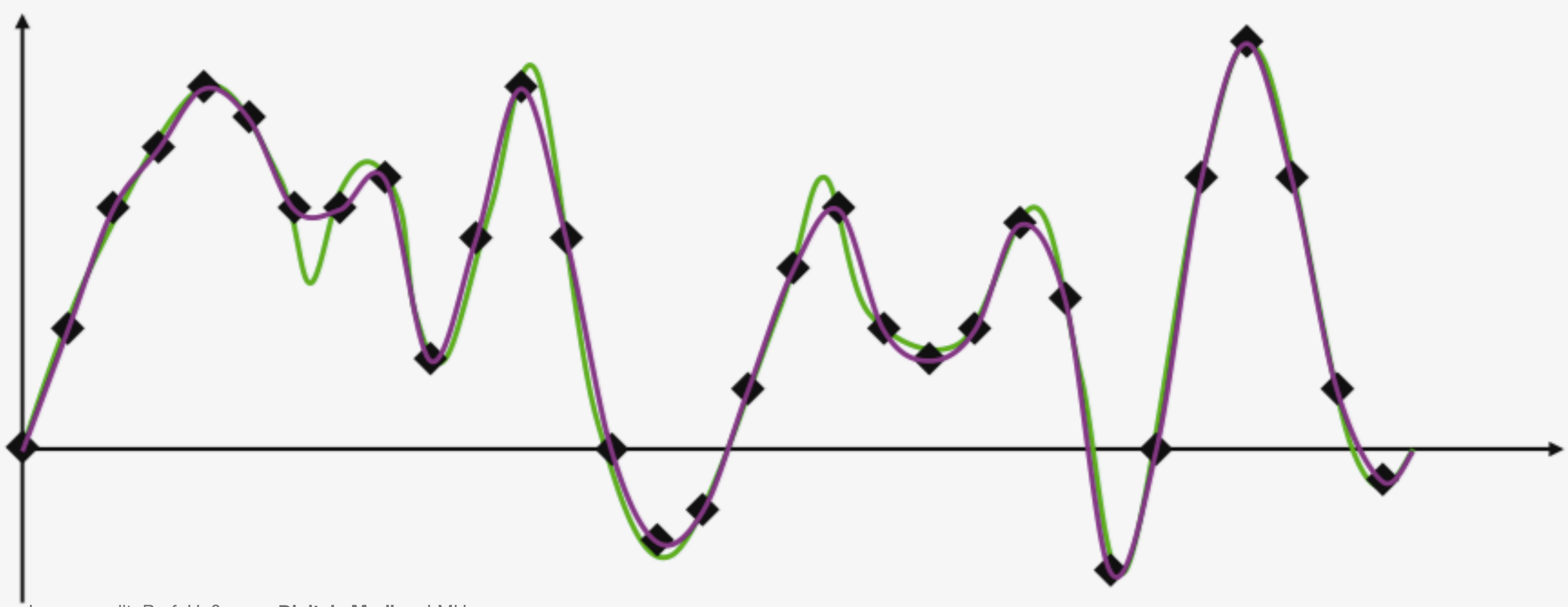
Digitization

- The array of numbers that is stored (or played back) is always an **approximation** of the original analog audio signal only



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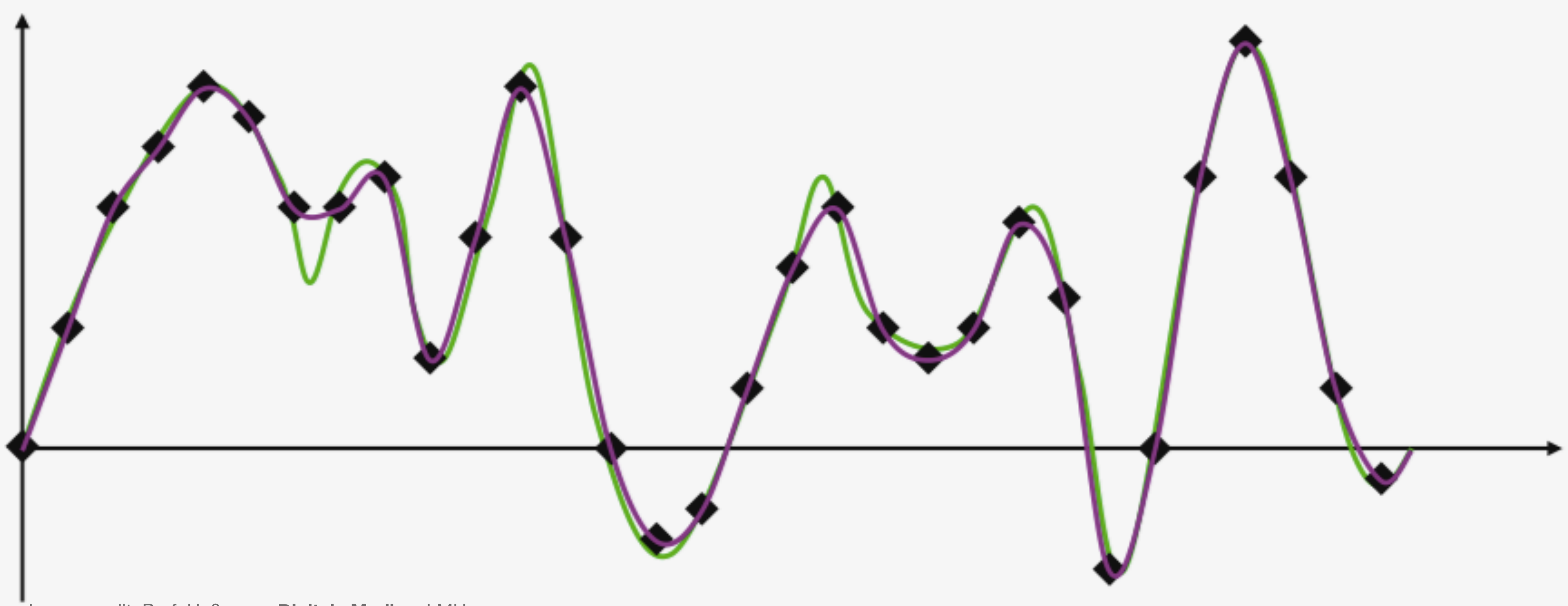
Digitization

- This array is called the **audio buffer**



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http://www.medien.ifi.lmu.de/fileadmin/mimuc/dm_ss04/dm2b.pdf

Audio Buffer

- An audio buffer is a list of discrete numbers that represent **amplitude against time** in a buffer (array) of **value and index**

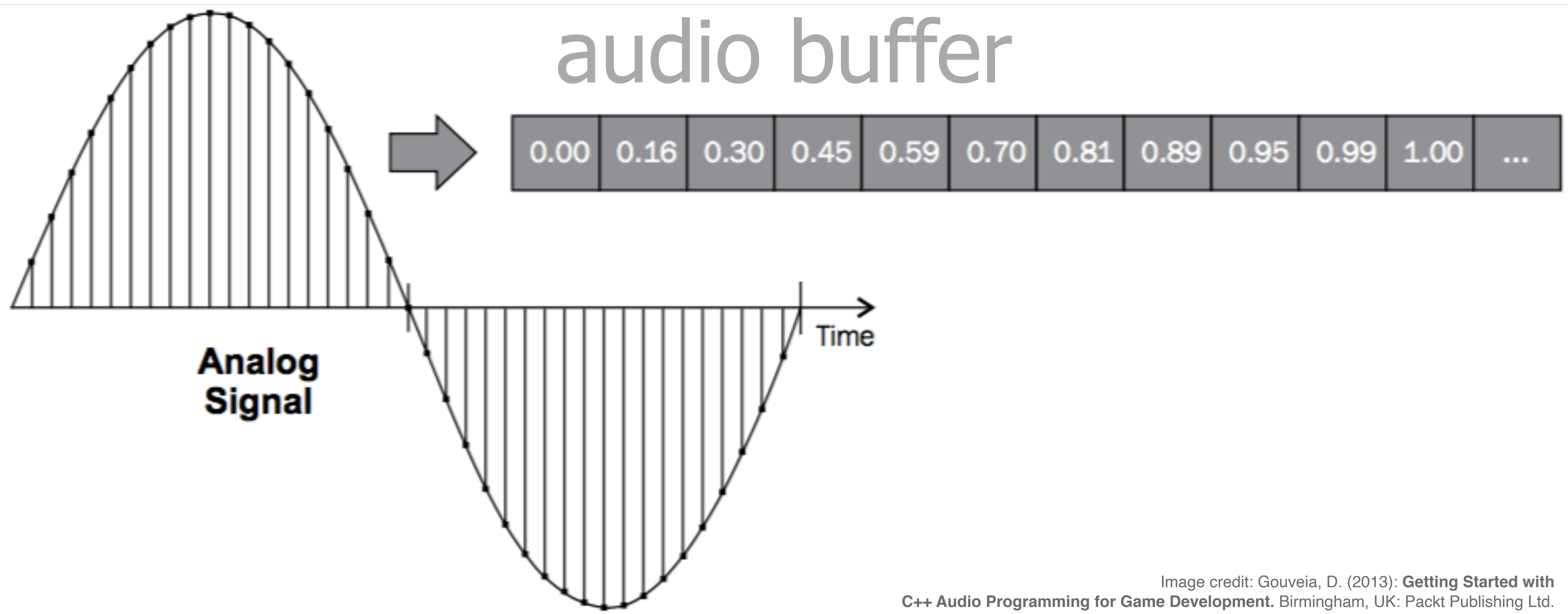
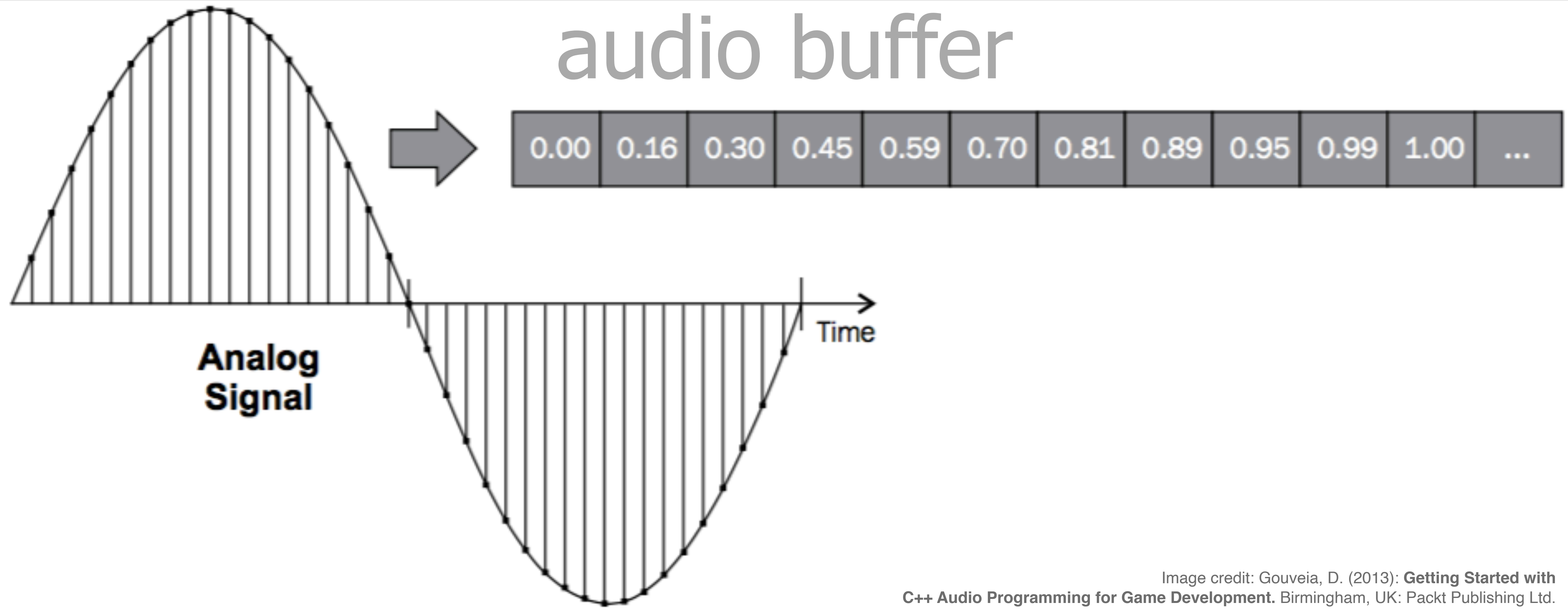


Image credit: Gouveia, D. (2013): **Getting Started with C++ Audio Programming for Game Development.** Birmingham, UK: Packt Publishing Ltd.

Audio Buffer

- When sound is recorded & digitized, the audio buffer is filled
- When digital sound is played back, the audio buffer is read



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