Multimedia Systems

Part 14

Mahdi Vasighi
www.iasbs.ac.ir/~vasighi

Department of Computer Science and Information Technology,
Institute for Advanced Studies in Basic Sciences, Zanjan, Iran

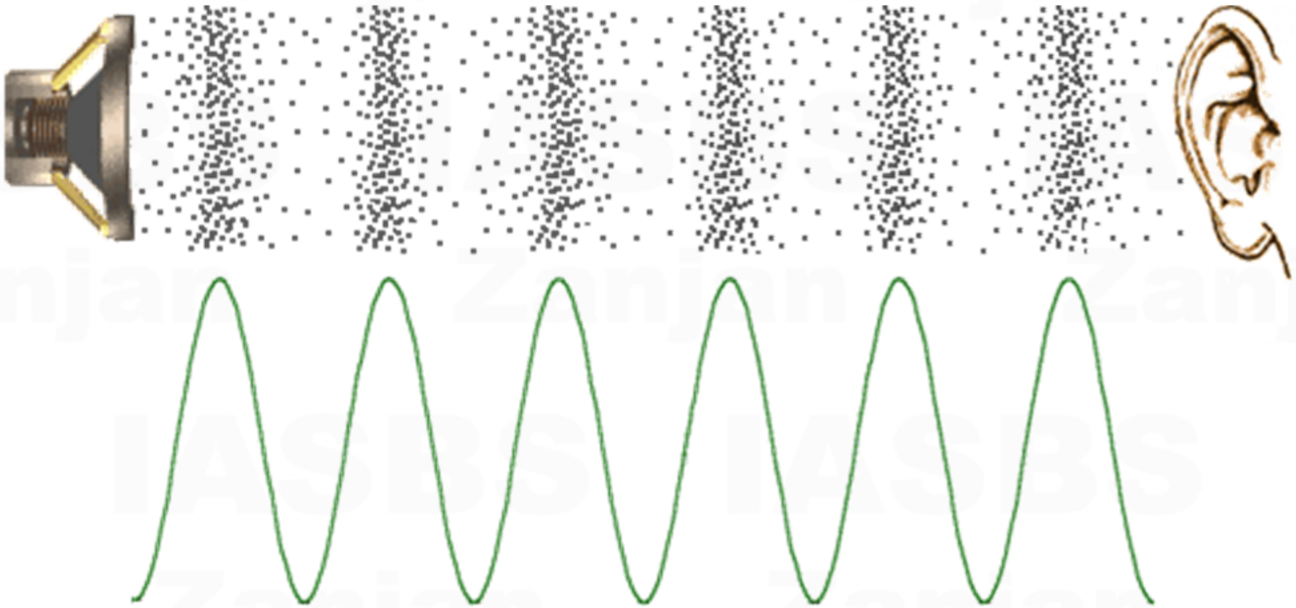
To be discussed:

- Digitization of sound
- PCM
- Lossless predictive coding

What is Sound?

A disturbance that travels (or propagates) through space and time, usually by transference of energy is known as a **Wave**.

Sound is a wave phenomenon, involving molecules of air being compressed and expanded under the action of some physical device.



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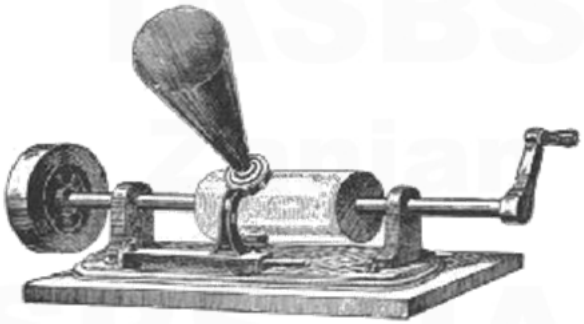
What is Sound?

- A speaker vibrates back and forth and produces a **longitudinal pressure** wave that perceived as sound.
- Since sound is a pressure wave, it takes on **continuous** values, as opposed to digitized ones.
- It has ordinary wave properties:
  - Reflection
  - Refraction
  - Diffraction

What is Sound?

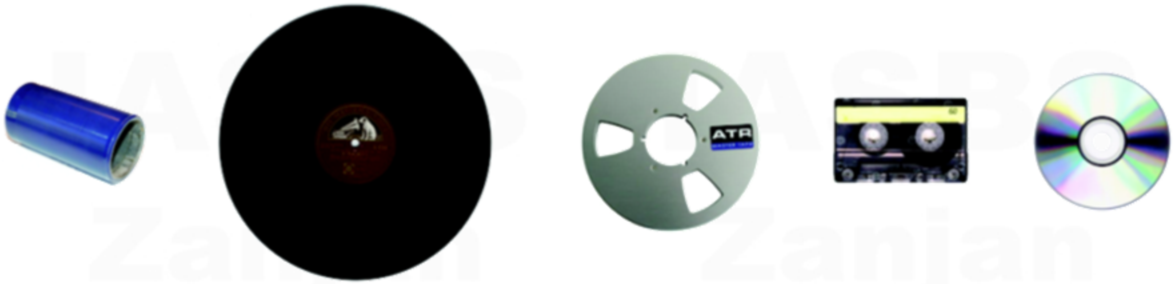
Thomas Edison's Phonograph 1877

- first device to record and reproduce sound
- a tinfoil sheet phonograph cylinder



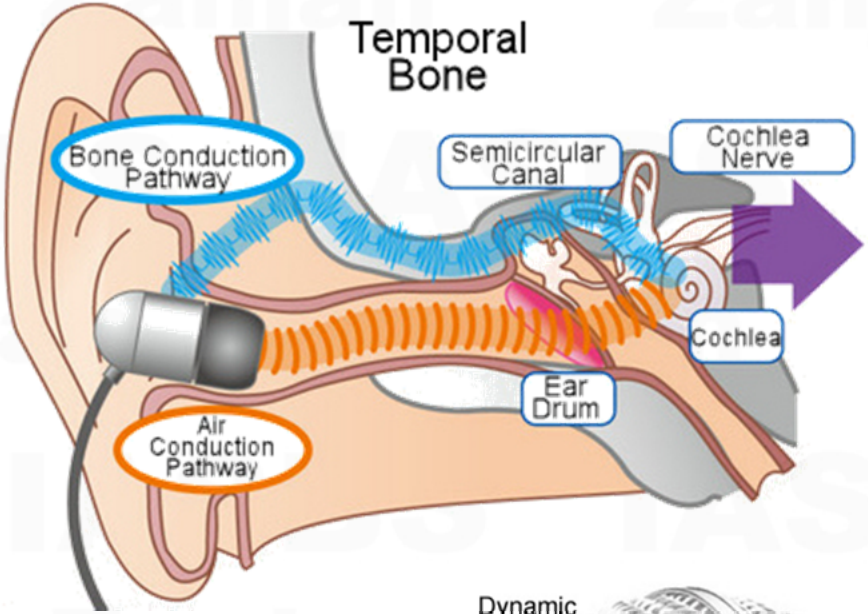
Alexander Graham Bell's improvement in 1880s

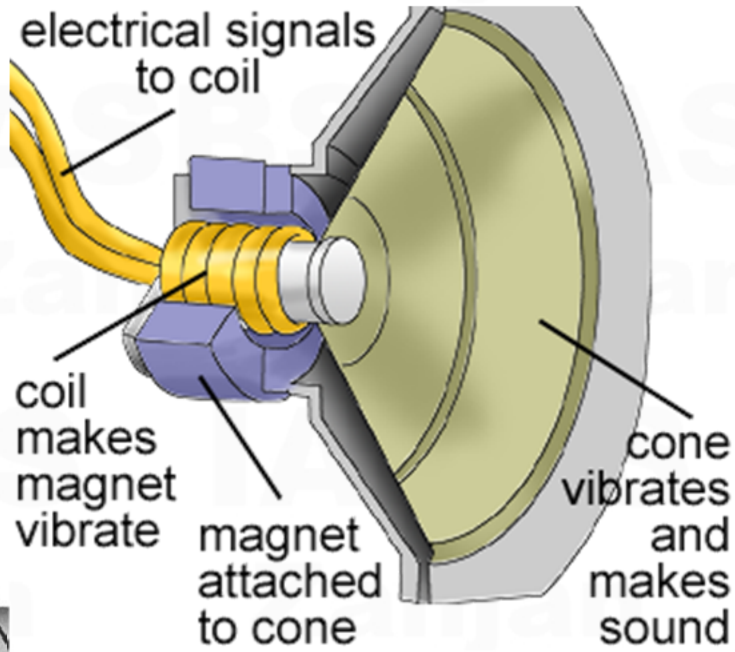
Emile Berliner’s gramophone in 1887 (double-sided discs)

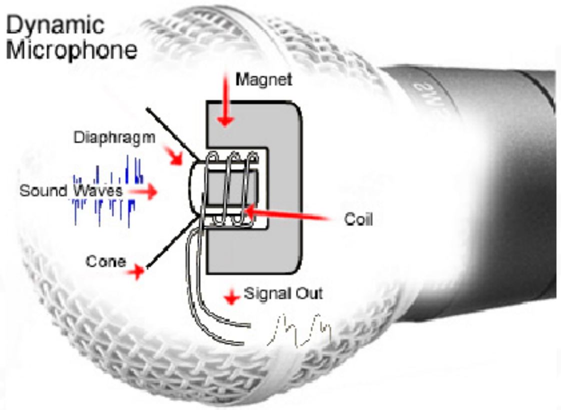


What is Sound?

The physical world is often analog!



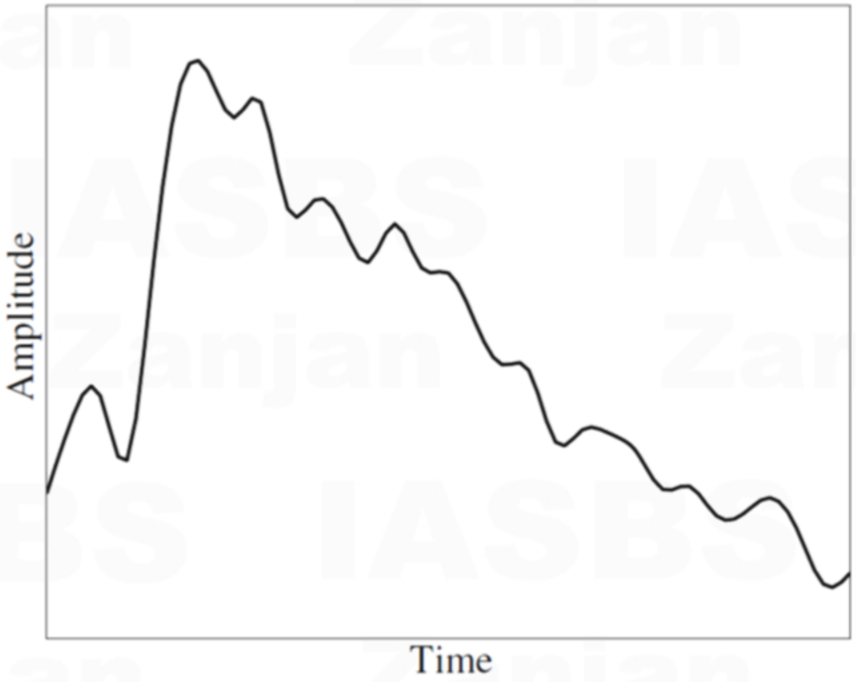


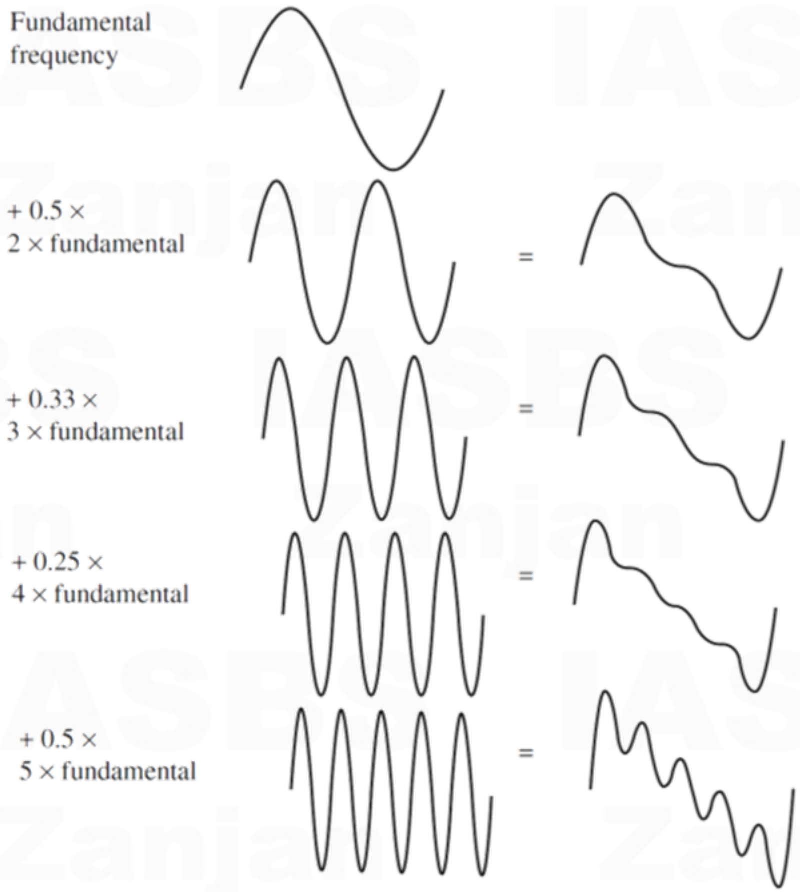


What is Sound?

1-dimensional nature of sound: **amplitude** (sound pressure/level) depend on a 1D variable, the **time**.

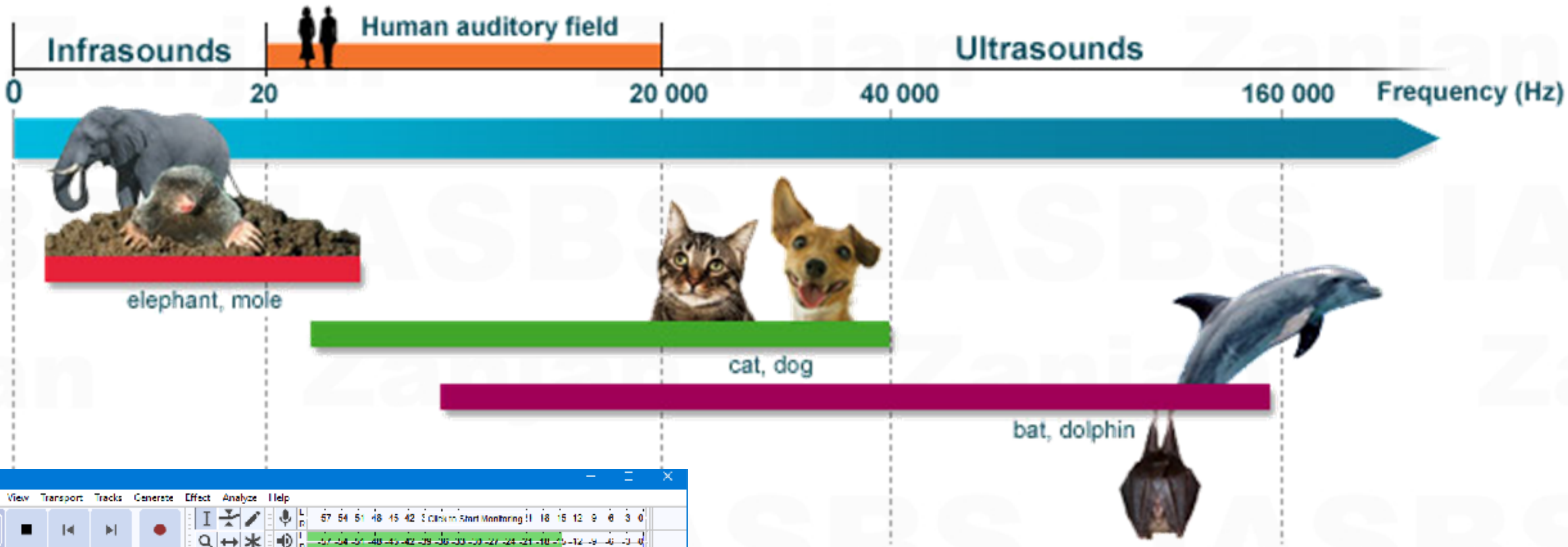
- Input from microphone is analog signal

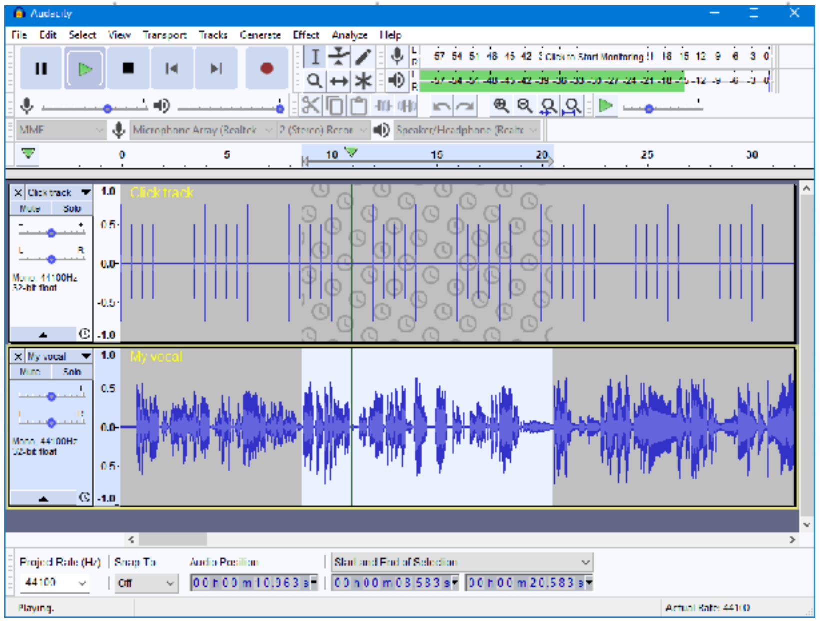




What is Sound?

Approximate hearing rage in frequency (Hz) in Mammals





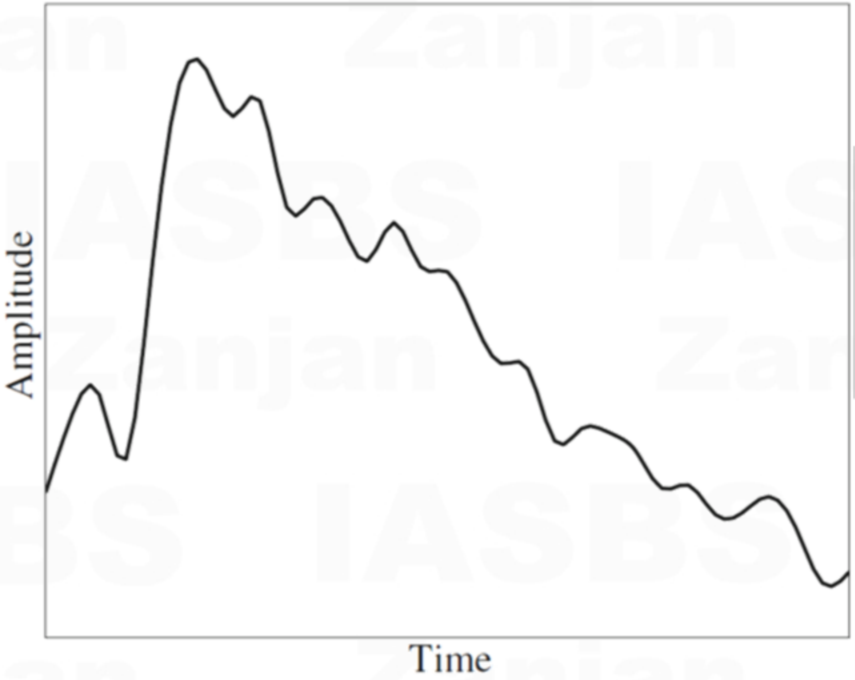
Use the tone generator in "**Audacity**" to test your hearing.

Digitization

The **amplitude** value is a **continuous** quantity.

- continuous-valued voltages

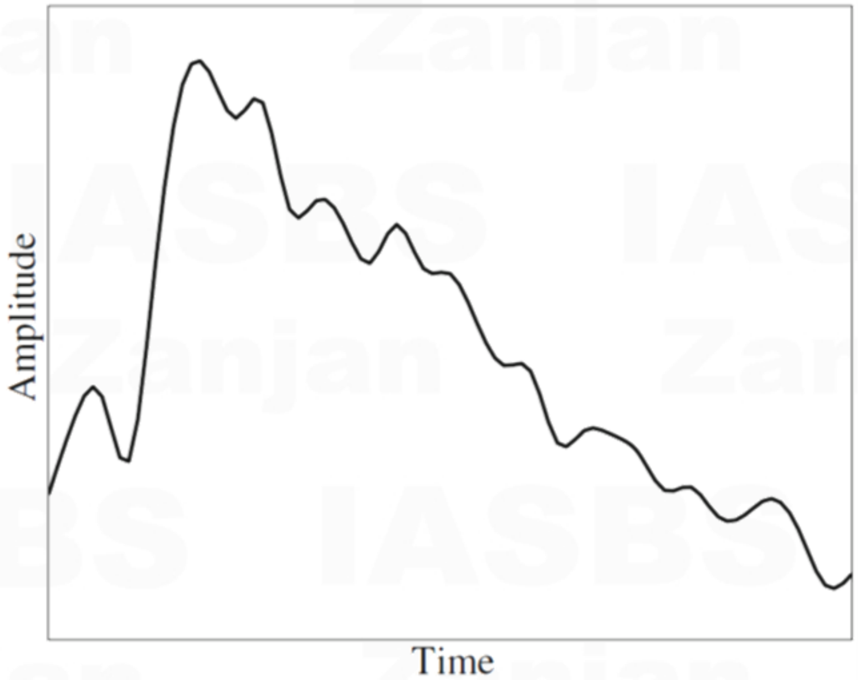
we must digitize the analog signals produced by microphones.



Digitization means conversion to a stream of numbers, preferably *integers* for efficiency.

Digitization

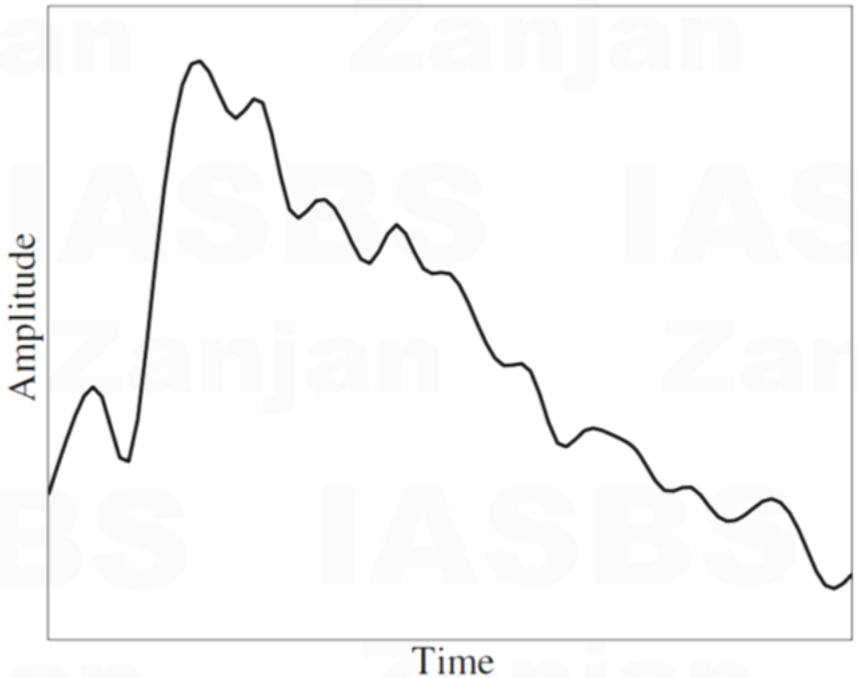
- **Digitization** must be done in both time and amplitude by **sampling**.
- Sampling means measuring the quantity we are interested in, usually at evenly-spaced intervals.

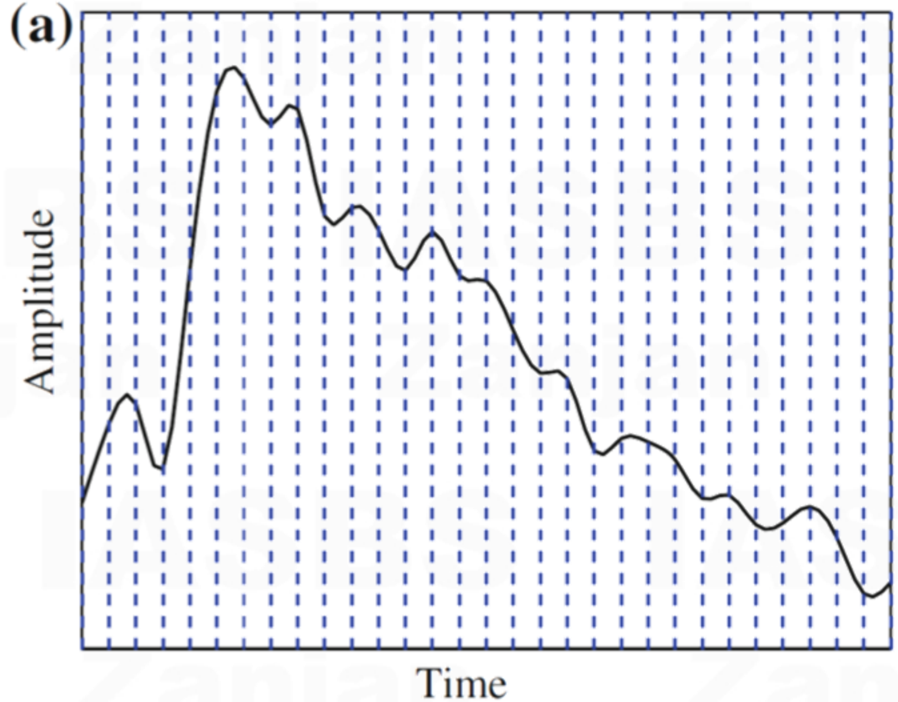


Digitization

First kind of sampling, using measurements only at **evenly spaced time intervals**, is simply called ***sampling***.

- The rate is called the sampling frequency
- For audio, typically from 8 kHz (8,000 samples per second) to 48 kHz.

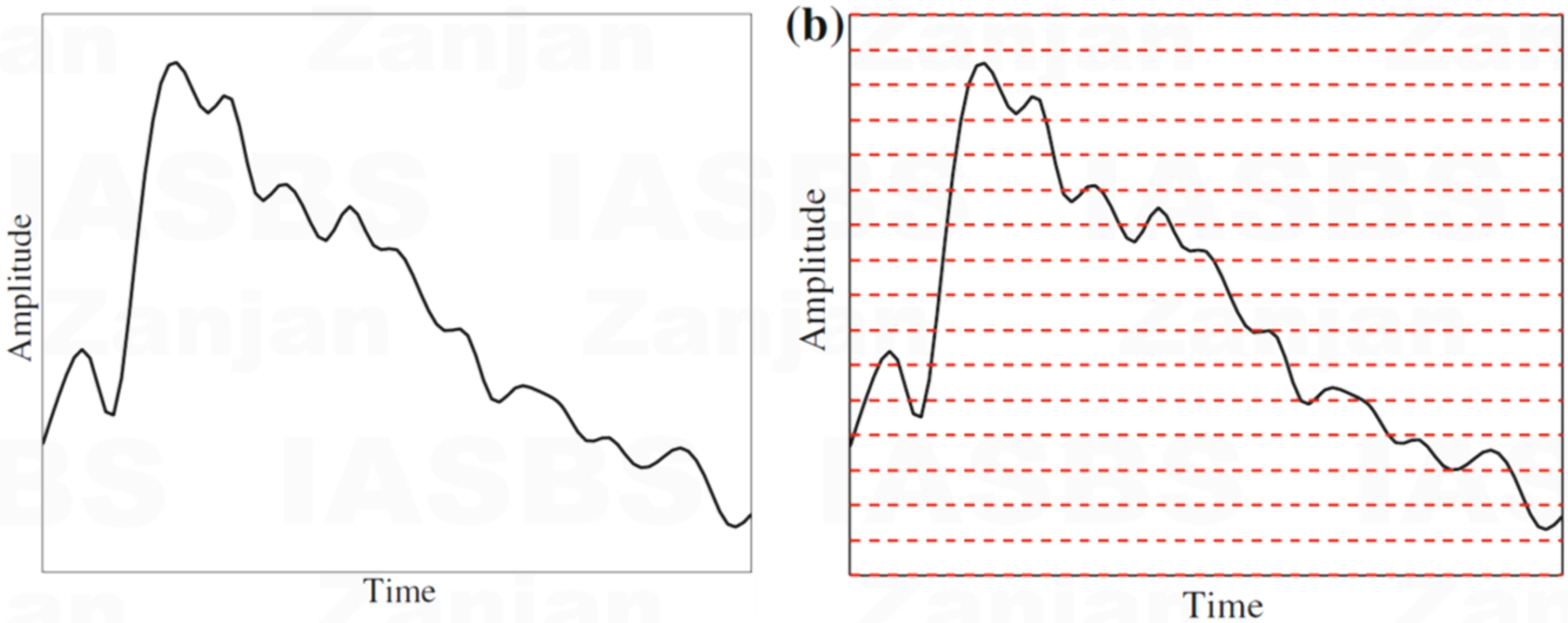




Digitization

Sampling in the amplitude or voltage dimension is called ***quantization***.

- Typical uniform quantization rates are 8-bit and 16-bit

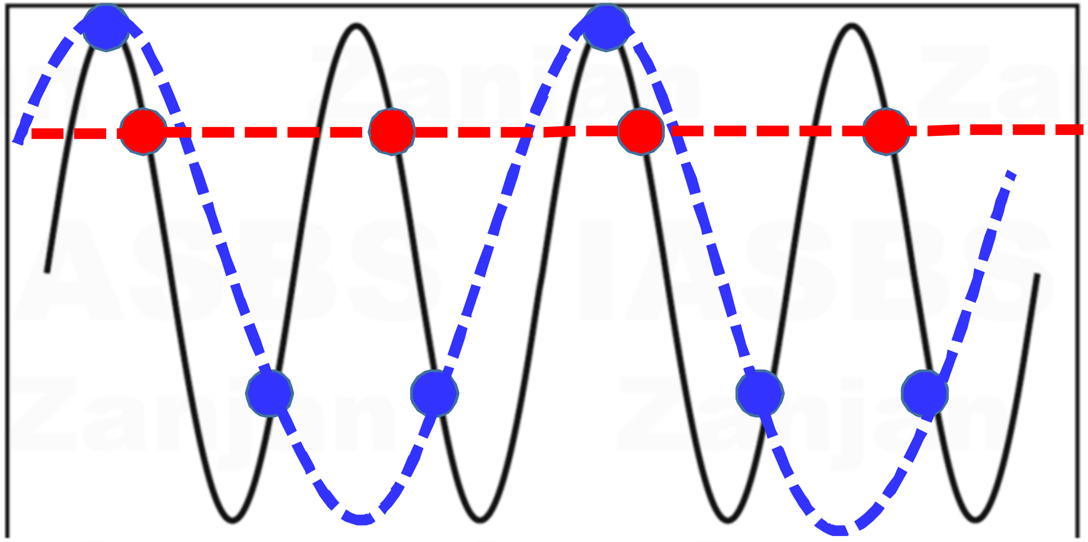


Digitization

To decide how to digitize audio data, we need to answer the following questions:

1. What is the sampling rate?
2. How finely is the data to be quantized?
3. How is audio data formatted?

Nyquist Theorem



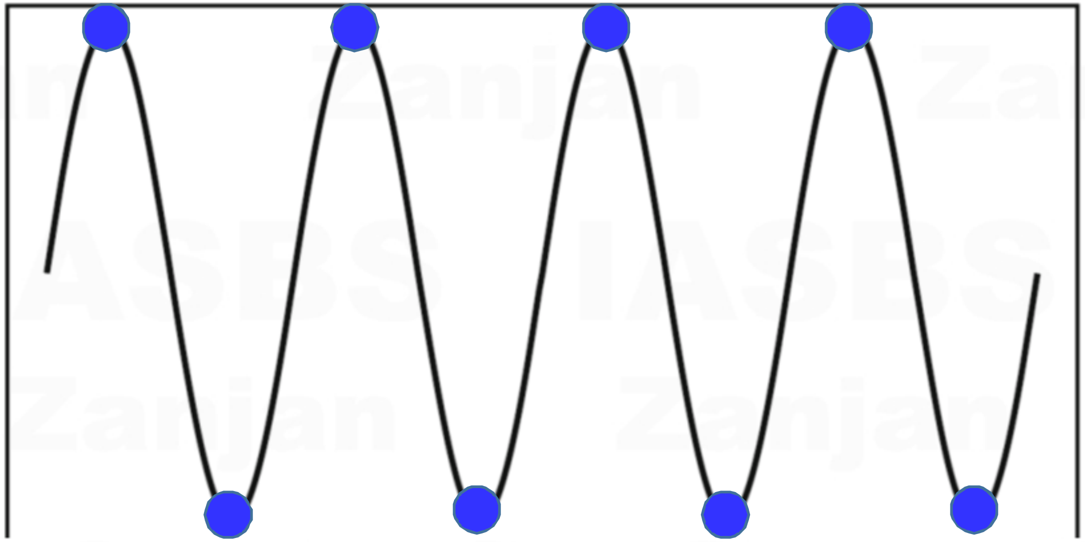
If sampling rate just equals the actual frequency

- a false signal is detected (constant, with zero frequency)

If sample at 1.5 times the actual frequency

- an incorrect (**alias**) frequency that is lower than the correct one (half the correct one)

Nyquist Theorem



For correct sampling we must use a sampling rate equal to at least twice the maximum frequency content in the signal.

sampling at 44,100 samples per second

This is called the ***Nyquist rate***.

Undersampling

Nyquist Theorem

The Nyquist theorem states *how frequently we must sample in time to be able to recover the original sound*.

More generally, if a signal is *band-limited*—that is, if it has a lower limit f_1 and an upper limit f_2 of frequency components in the signal—then we need a sampling rate of at least $2(f_2 - f_1)$.

- The frequency equal to half the Nyquist rate is called the **Nyquist frequency**!

$$f_{\text{alias}} = f_{\text{sampling}} - f_{\text{true}}, \quad \text{for} \quad f_{\text{true}} < f_{\text{sampling}} < 2 \times f_{\text{true}}$$